YAMAHA AW4416, JOHNSON J-STATION, ULTRAFUNK SONITUS:FX, AND 11 MORE REVIEWS

Electronic Musician®

www.emusician.com

March 2001

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The HDR24/96 versus recording on

A fully-equipped LESS than three

With an HDR24/96 it's so easy to record, edit, and manipulate tracks, so easy to be creative…whether you're recording for yourself, your band or for a Fussy Client.

With all due respect, recording onto linear media (a.k.a. tape) has some pretty severe limitations: Access time to cue points is slow. Punch-ins erase stuff you previously recorded. And the tracks just sit there side-by-side on the tape with no chance to easily slip, slide, cut or paste them in new ways.

Hard disk recording and workstation editing for less than the price of linear recording.

It's no secret that non-linear hard disk recording is the way to go. But until now, 24- track/24-bit recording and playback required serious investment in a digital audio workstation. — And a heckuva lot of mousing and clicking.

Only the HDR24/96 combines the intuitive, analog-like convenience of a tape deck with the editing versatility of a computer-based workstation.

As easy to use as an analog recorder.

All basic functions are right there on the HDR24/96 front panel including transport buttons and individual Record Enable buttons for each track. Just hit Record and Play

* based on current U.S. list and pro audio dealer "street" prices at the time of ad production and on the assumption you will buy a CRT-type SVGA monitor and not an ultra-pricey flat panel model like our art director insisted on using for this ad.

** based on average of length of current pop songs using 24 tracks @48Hz/24-bits and a liberal number of extra regions and virtual takes. Does not apply to extended trance remixes.

long thin strips of rusty plastic.

Mackie 24-track HDR24/96 Hard Disk Recorder/Editor costs tape-based, 8-track digital recorders*...and does much more.

without even cracking the manual. But if you plug in an SVGA computer monitor, things get even better.

The graphic interface that tape recorders always should have had.

Even if you immediably don't use the HDR24/96's editing functions, you'll love the graphic interface for recording.

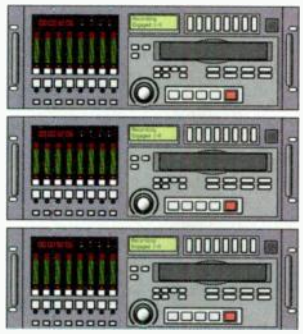
It gives you one-click access to all deck functions without a lot of annoying pull-down/fly-out menus.

Choose from 2x, 4x, 8x, 12x or 24-track views and then watch them scroll smoothly past a centerline.

Mark hundreds of cue points and four locate points for looping and auto-punch-in modes. Cue points are visible on screen and are accessible from a side list.

Use the mouse to "scrub" individual tracks, Cue, Punch and Loop points with continuously variable velocity.

Each track also supports eight "virtual tracks," so you can do multiple takes and comp them together easily.

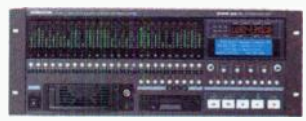


Think of MackieMedia™ as "tape in a brick."

Right out of the box, the HDR24/96's internal drive will record 90 minutes of 24 tracks at 48kHz. Your backup choices are simple—

1. Record directly to a MackieMedia™ M90 external drive. They're considerably less expensive than the SCSI drives some HD recorders require — $10 a song** — and they're in stock at your Mackie dealer.
2. For a quick back-up of just a song or two, we also offer an optional 2.2GB ORB™ drive that uses *really* inexpensive media.
3. For real economy use the HDR24/96's 100BastT Ethernet port to back up to your computer and its media.

Even with three OPT•8 I/O cards, a MackieMedia removable disk, SVGA monitor, keyboard and mouse, the HDR24/96 costs less than three digital tape recorders...which don't offer loads of workstation-style editing features, super-fast access and true 24-bit recording.*



Serious editing tools built in... with 999 levels of un-do.

Once you've experienced non-destructive editing of tracks, you'll never go back to linear recording.

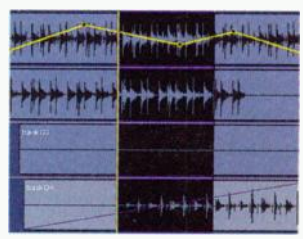
You can mark a segment (or multiple non-adjacent segments) as a *region* and then cut, copy and paste it anywhere — onto a blank track or right in the middle of an existing track without erasing anything (the part of the track after the insert just "slides down").

You can audition regions or modify their start/end points instantly, capture them as "sound elements" for later use or quantize them to user-defined time grids. And all regions are easy accessible from a side menu.

Create fade-ins, fade- outs and crossfades just by dragging and dropping them ...and then set their length by dragging the mouse.

Add volume envelopes for simple level automation of regions or whole tracks.

Then use Track Render to combine all or selected regions of a track just as you hear it — complete with crossfades, volume envelopes, mutes, etc. — into a single region.



Zoom in to the waveform level. Drag, cut, paste, and slip tracks and segments of tracks just like on super-expensive workstations. Adjust track levels. Add editable crossfades. All with 999 levels of undo.

Get a demo at a Mackie dealer.

This ad only scratches the surface of the HDR24/96's features, options and capabilities. Visit our web site...or get your hands on an HDR24/96 and experience (pun intended) *unparalleled* creativity.

*Twenty-four track songs for under ten bucks each!! Divide the cost of a MackieMedia M90 into the 20+ pop tunes you can record on it and you're looking at under a ten-spot for each 24-track master. —Much less if you do a little disk drive housekeeping. Remember, hard drives can choose only to store audio data, not silence. Tape just rolls merrily along...costing money... whether you're using a track or not.***

Hard-shell case. Really ugly pink anti-static foam.

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KARMA ...7 YEARS IN THE

At Korg we're always dreaming up innovative ways to make our keyboards more expressive, more powerful, more musical, and when we first learned about a brand new technology called KARMA™, we knew we'd found one. Seven years after assembling a global team of skilled engineers, programmers and musicians, the Karma Music Workstation brings new music-making possibilities to life.

To start, its synth and effects engines are identical to Triton's, so you know how good it sounds. It's also compatible with Triton sound data and supports the same user-installable PCM and MOSS expansion options so your sound palette can grow. Plus, the Karma Music Workstation provides the same feature-packed 16-track sequencer that has made the Korg Triton the leader in music workstations. So what makes *this* new keyboard so special?

It's KARMA, our revolutionary technology that generates amazing phrases, grooves and other musical effects that can be altered and randomized in real-time. With a bank of knobs and switches, you control elements like rhythmic complexity, harmony, melodic repeat, phrasing, panning...even the synth's sound and effects.

KARMA gives you the power to play impossible, interweaving cascades of notes, techno arpeggios and effects, dense rhythmic and melodic textures, natural-sounding glissandos, intricate fingerpicking and guitar strumming, swooping portamento and pitch bend moves, to name but a few. The only limiting factor is your imagination.

Karma like this only comes around once every seven years. Fortunately for you, the wait is over.

GOOD KARMA. WICKED SOUND.

SPEC SHEET

- 61-key synth action keyboard with velocity and aftertouch sensitivity
- 62-note polyphony
- 32 Mbyte wave ROM, expandable to 64 Mbytes with 2 EXB-PCM expansion boards
- 640 Programs (including GM Level 2)/384 Combis – expandable to 896 Programs/640 Combis
- Support for EXB-MOSS (adds 128 new Programs/64 new Combis)
- 102 Insert/90 Master effects (up to 5 Insert, 2 Master effects plus 3-band EQ)
- 16-track 200,000 note sequencer, 200 Songs, 20 Cue Lists, 100 patterns per Song, 150 preset drum patterns, 72 RPPR patterns per Song, 16 preset/16 user Template Songs
- Joystick, 4 assignable knobs, 2 assignable switches and 1 assignable slider

KARMA features:

- Over 1000 Generated Effects (1 GE per Program, 4 GEs per Combi or Song). A GE contains over 400 parameters to generate notes, control synth and effects parameters, and provide randomization of these events
- 8 knobs, 2 switches and 2 scene memories, plus joystick, slider and pedals for real-time control over GE parameters
- 4 programmable Chord Memory buttons for triggering chord voicings easily
- The ability to load new GEs as they are released

MAKING

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MUSIC WORKSTATION

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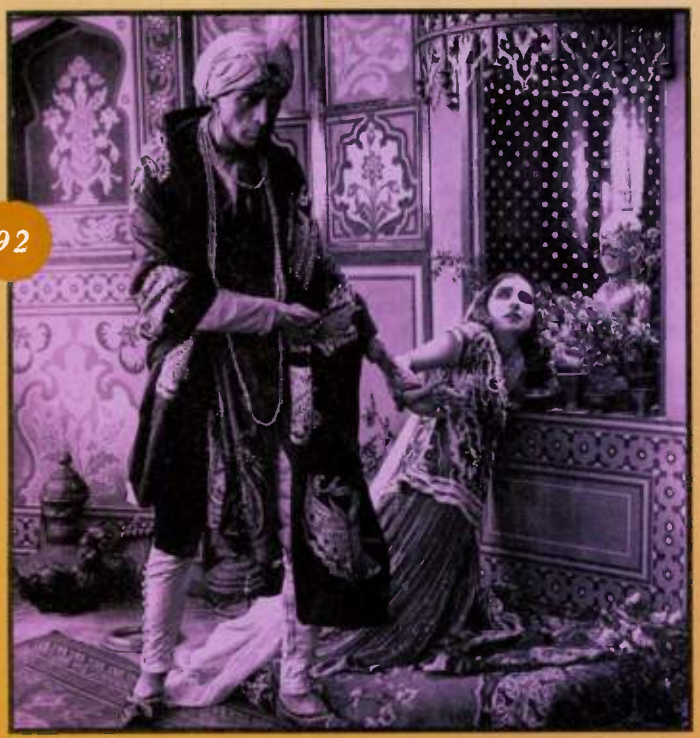
I N S I

FEATURES

Cover illustration by Tivadar Bote



DEPARTMENTS

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MARCH 2001 VOL. 17, NO. 3

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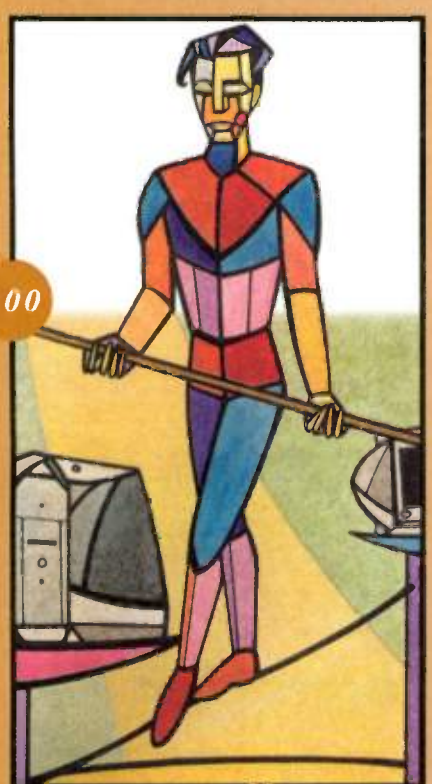
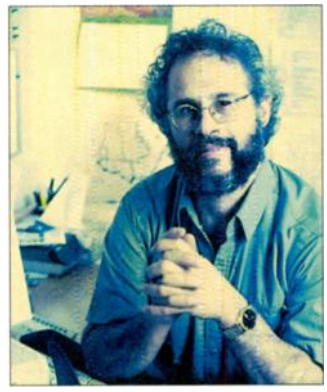
15 YEARS EM 1986–2001

COLUMNS



REVIEWS

Giving Back


ANTHONY PIDGEON

Readers of this magazine are, in general, an exceptional group of people. According to our surveys, most **EM** readers are well educated, and many of you have achieved at least some success in your businesses, music related or not. Given the magazine's subject matter, it's reasonable to conclude that most of you have musical talent and an above-average affinity for technology. And obviously you have great taste in magazines!

Much of what is good in your musical life probably happened partly because of your own hard work and partly because people helped you. Perhaps family, friends, teachers, or local musicians supported your musical aspirations, helped you get started, and showed you the ropes. Someone showed you how to play a simple tune, helped your first band land its maiden gig, and urged you to get started in recording. Friends and mentors explained the basics of stagecraft, showed you how to wire your studio or stage rig, and turned you on to **EM**. Those mentors and supporters may have had a variety of motives, but in many cases they helped simply because they felt it was the right thing to do. Most likely they remembered people who helped them along their way.

The best way to thank your benefactors—besides saying thanks privately and occasionally crediting them publicly—is by helping others. In addition to your experience with music technology, you undoubtedly have other useful skills. Use them to help someone else in your community.

You can contribute money, of course. Sometimes just 10 or 20 bucks can make a big difference to a music program or a nonprofit organization. But donating your time is even cooler. America's greatest strength is arguably its citizens' willingness to volunteer for good causes.

Therefore, I urge you to get involved in something that benefits society, whether music related or not. Think about your interests and use your skills, or learn new ones, for a good cause. You can make a difference by playing occasional free gigs for charity events, which is something all bands should do. But you can go beyond that. Your recording skills or live-sound engineering skills could be useful to a charitable organization. (I found I could contribute significantly in that way.) Perhaps you can find ways to use design skills you acquired when creating your Web site.

Of course, you can work on big social issues like feeding the hungry, but you also can improve your community in more subtle ways. You don't have to join an organization to show a child how to tune a guitar, help a senior citizen or disabled person learn to use a computer, or teach a band that is eager to improve how to polish its act. Just pass the knowledge on.

Yes, you will encounter frustrations, even when helping people who sought your help. But your contributions can be far more important than you realize. And you can't beat the "job" satisfaction; in my experience, the good you do will indeed come back to you tenfold.

Electronic Musician®

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The Soul of Analog, the Power of Digital

Professional Audio Recording Integrated Solution

Analog Sound Quality
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Intuitive and Flexible Software
PARIS 3.0 Software offers you the most intuitive user interface on the market, emulating the logic and flow of an analog console, patchbay and tape multitrack. And PARIS imports and exports OMF files (Avid's Open Media Framework®), giving you compatibility with Pro Tools® projects and studios.

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It's easy to expand PARIS Pro – simply add the I/O and Sync modules you need to your MEC (Modular Expansion Chassis). Need more tracks and effects? Each EDS PCI Expansion Card gives you 16 more tracks (up to 256), effects and 64 EQ's.

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PARIS Pro offers you a dedicated 16 fader control surface, Modular Expansion Chassis, true Mac/PC cross platform software, and EDS PCI Card for $4995. Choose PARIS Pro and then take the money you saved and treat yourself to some nice mics and a rack of high-end preamps – you deserve it.

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*RS-5 also available with 61-note keyboard

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Keys: 88 semi-weighted	100%	64-voice polyphony	100%
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Roland

WHAT GIVES?

I have been a subscriber to your magazine for about five years now, and I owe you a debt of gratitude. It was through **EM** that I gained a new interest in home recording. I own a professional digital 16-track recorder, and I am having a blast. Thank you!

Although I have enjoyed your magazine for years, I have noticed a change in your focus. Are you still primarily concerned with the home recording artist? Has the Internet explosion changed your vision? It seems I have not seen any articles lately on the recording process itself. Where is the practical instruction on mic placement, how to tune a room, or how to EQ a bass? Do you have fewer sound engineers on staff now? Lately I have been finding less articles in your magazine that are helpful to my endeavors.

Am I, Mr./Ms.-home-recording-artist-with-a-dream-of-making-the-ultimate-CD, still the bread and butter of **EM**, or has there been subtle shift of your clientele that I am still clueless about?

Ken Schleimer
Scotts Valley, CA

Ken—I'm surprised at your comments. It's true that we write a lot about music production for the Web because the vast majority of that recording is being done in personal studios. However, we also have published a steady stream of recording stories. You specifically mention mic selection and placement; perhaps you missed our September 2000 cover story "Build a Microphone Cabinet on Any Budget" and our August 2000 "Master Class: The Bottom Line" on recording electric bass, not to mention our "Recording Musician" columns, many of which discuss miking. Recent features cover such personal-studio issues as mixer topology and usage ("Mixed Signals" in the November 2000 issue), compressors ("The Big Squeeze," February 2001), mixing for 5.1 surround ("You're Surrounded," October 2000), and cabling ("Good Connections," January 2001).

Add to that our September 2000 interview with Eddie Kramer ("Mixing With a Master"), an assortment of Master Classes on hardware and software synthesizers (the Proteus 2000 in the November 2000 issue, Csound in October 2000, and Native Instruments Reaktor *in September 2000) and a full slate of reviews, and I think you'll find a heck of a lot for personal studio owners. That's just in the last six issues, and I left out a lot.—Steve O*

SCSI SHOCK

I recently went out to purchase a recorder, and boy, was I in for a shock. My ignorance has brought up a question that the equipment manuals, people at music stores, and online research can't answer.

On a digital hard disk recorder, there is a SCSI output for backing up your work or for connecting a SCSI CD-R to record a master CD. Can I connect it to my computer and record a master CD on my computer's CD-R? I have a SCSI card in the computer.

Keith McCormick
via e-mail

Keith—The answer is it depends, but you probably should not directly connect the two. Here are the issues:

1. The SCSI port must be capable of saving to a fixed hard drive, not just capable of burning a CD. The owner's manual will have that information. Some HDRs have built-in drivers for CD burning, but that does not mean they will also work with a fixed drive. However, most will work that way.

2. The computer software would have to be configured with a driver to talk with your specific HDR, just as with a sampler. That means the HDR is also designed to talk to a computer. For instance, if you have a K2000 sampler and use BIAS Peak *on the Mac, you can connect the K2000 to the SCSI chain on the Mac because* Peak *has a built-in K2000 or SMDI driver that handles the communications with the K2000. Without software support, you can't do the transfer. Also, the SCSI spec only permits one controller at a time on the SCSI bus. Without a driver, the HDR and the PC/Mac would fight for control of the disk. That might not damage anything, but the system would hang.*

Some portable digital studios/HDRs and software are designed to work together. For instance, C-MEXX makes software for the Roland VS-1880. But in that case, the audio never goes to the computer; it is merely controlled by the computer and resides on the VS-1880. That is not the same thing as what you want to do, which is to actually transfer the audio.

There might be a work-around if the HDR supports a standard PC disk format and file format, such as DOS disks and WAV files. If the HDR's SCSI bus is capable of backing up to an external DOS-format hard drive, you can format the drive as a DOS disk (do that on the Mac or PC, not on the HDR), connect it to the HDR, and record the data as WAV files. Then shut down the system, switch the hard drive back to the computer's SCSI bus, and boot up. Now you have the WAV files on a DOS disk mounted on your Mac or PC. (Macs can mount DOS disks and read WAV files.) From there you can open the files

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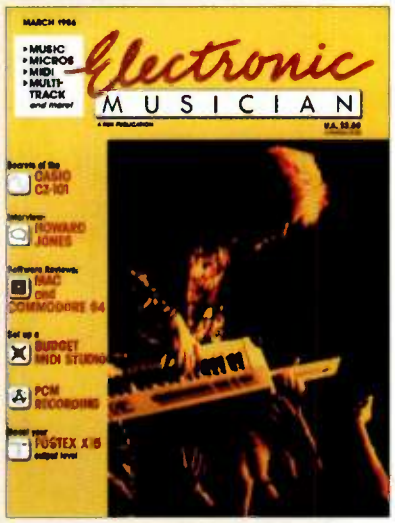
Creativity First

FIFTEEN YEARS AGO IN EM

We dove neck deep into MIDI in March 1986; in fact, we created a whole new MIDI section of the magazine. All three MIDI stories were about live performance: one on creating an all-MIDI band, one on getting started with MIDI, and one about increasing the keyboardist's mobility on stage by using a strap-on MIDI keyboard controller. This emphasis on live performance reminds us that in the early years of MIDI, the protocol was most often used for controlling two or more synths from a single controller. Sequencing was still relatively new.



The cover photo of Howard Jones also reflected our "mobile MIDI" story: it shows Jones playing a Yamaha KX5 strap-on keyboard controller. (I remember my KX5 fondly; it had small keys, but it was lightweight and had a cool pitch-bend ribbon instead of a wheel.) Inside, Jones discussed his cutting-edge gear, including a Yamaha DX7 synth, E-mu Emulator II sampler, Sequential Prophet T8 synth, Roland Juno 60 synth, Yamaha CP80 electric grand piano, and eight Yamaha TX7 synth modules. That was a rig to die for.

The March issue included two DIY projects. Thomas Henry explained how to build a high-frequency voltage-controlled oscillator (VCO) that could be modulated over a 7-octave range. Henry included several cool features: his VCO could do linear or exponential frequency modulation, had coarse- and fine-tuning controls, and provided temperature compensation, which helped stabilize the frequency.

Our DIY other project was simpler. Author Craig O'Donnell noticed that the four individual track outputs on his Fostex X15 4-track cassette recorder delivered hotter audio levels than did the stereo and monitor outs. To fix this problem, he came up with a DIY modification that increased the audio output level significantly, thus improving the X15's signal-to-noise ratio.

Our top applications story was part 2 of Larry the O's series on digital reverbs, "Sing a Song of Reverb." The O got down to products in March, rounding up the nine digital reverbs then available for less than $1,500: the Alesis XT:c; ART DR1, DR2a, and 01a; DOD RDS-6400; Lexicon PCM60; Roland SRV-200; Ursa Major StarGate 323; and Yamaha REV7. We offered specs, features, and a hands-on evaluation of each unit except for the RDS-6400, which shipped too late to be tested.

Casio synth lovers responded enthusiastically to Alan Gary Campbell's CZ-101 programming story. Incidentally, after an extended hiatus, Campbell is back in this March 2001 issue (see "Square One: What's a Web Page" on p. 110).

The review section was all but dominated by two Geary Yelton reviews of Macintosh programs: Mark of the Unicorn's *Professional Composer* music-notation software and Opcode's *MIDIMAC* patch librarian, a predecessor of *Galaxy*. The other big review was by Commodore 64/128 expert James Chandler, who gave us the lowdown on a package from Moog Music called the Moog Song Producer. The bundle included a MIDI interface (one In, four Out, one Thru); *Songstepper* and *MIDI Drum Songstepper*, a pair of nearly identical drum sequencers; and a version of Dr. T's *Keyboard Controlled Sequencer* for the Moog MIDI interface. Finally, assistant editor Vanessa Else gave us some relief from high tech with her review of Invisible keyboard stands.

—Steve Oppenheimer

● LETTERS

in a CD-burning program and burn your CD from your computer.

I have done this with my K2000RS because it is located a bit too far from my Mac for its SCSI-1 bus to work properly. So I use a pair of old SyQuest drives, one on the Mac and one on the K2000. I format a cartridge in DOS format on the Mac, move the cartridge to the K2000, save my samples, and move the cartridge back to the Mac. But I could do the same thing with a hard drive if I had to.—Steve O

GREAT SCOTT

It's great to see **EM** explain some of the important aspects of electronic music's history. The Joel Chadabe series and the Raymond Scott article ("Circle Machines and Sequencers," December 2000) really thrilled me. A lot is happening in contemporary music, but it's at times a bit too self-important. When your magazine chooses to show how 30 years ago a guy took 100 electromagnets to build a sequencer, it puts today's practices in perspective.

At the same time, your Web articles do a great job of sifting the wheat from the chaff. Much the same way the *New York Times* is used to teach in college classes, I see **EM** taking the same role in music schools. Keep up the good work.

Peter DuCharme
via e-mail

"O" FOR OMISSION?

I love your magazine. Your articles have made my life a lot easier when it comes to my home studio—and a lot more difficult when it comes to my gear fetish.

I appreciated your article in the December 2000 issue of **EM** entitled "The Route Less Traveled." I have recently been lying awake at night wondering how I am going to get all my digital gear talking. One thing I did note, however: you did not mention Mark of the Unicorn's 2408. At $999, its primary purpose is still as an audio interface for the PC, but it also happens to provide a much more extensive digital routing system than anything else at

VST - The Sampler

HALion is Steinberg's VST sampler!

Perfect integration with Cubase VST.
Perfect sound: 32-bit and 24-dB filter.
Perfect grooves: sample-accurate timing.
Perfect interface: revolutionary drag and drop navigation system.

The big picture. The program window allows you to organize and recall your programs to work more efficiently.

HALion includes all the functions you've come to expect from a sampler like pitch, modulation, and reverse while the Modulation / Tune window gives you the flexibility to experiment in realtime.

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The Loop Editor finds the exact loop point while the crossfade function creates smooth transitions.

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Steinberg
Creativity First

that price. I am currently using it to route between two TDIFs and an S/PDIF.

I think this omission is also important because the writer mentions that he only found one box that does TDIF.

Jonathan Baker
Irvine, CA

Jonathan—I considered including the 2408 and other I/O interfaces, but it would have opened the door to so many devices with some degree of ancillary routing that the article would lose its focus. The other reason I didn't include I/O interfaces is because they are not standalone devices, as the rest of the units mentioned are.

Having said that, I agree that the 2408 (which I, like you, use in my studio) offers more versatility in routing than other similar devices, allowing stereo pairs of channels to be freely routed between analog, S/PDIF, ADAT Lightpipe, and TDIF.—Larry the O

MIC MYTH

Your January 2001 evaluation of the AKG C 2000 B microphone by Karen Stackpole perpetuates a myth about the frequency response of large-diaphragm condenser microphones.

Extended low-frequency response is not a large-diaphragm trait. The microphones that have most extended and smooth low-frequency responses are small diaphragm microphones such as those built by Bruel & Kjaer and marketed by DPA (among others).

Large-diaphragm microphones excel in vocal use because they are less sensitive to "popping" and have inherently less self noise and higher output compared to their small-diaphragm counterparts.

A small-diaphragm microphone also has faster transient response that may translate into a more "open" high end, but that is not at the expense of low frequencies.

Richard H. Kuschel
via e-mail

Richard—You are correct about small-diaphragm condenser mics having, in general, a more extended low-frequency response than large-diaphragm condensers—especially if you're referring, as I assume you are, to true omnidirectional microphones such as the DPA (formerly B&K) 4006 and Earthworks QTC1. Of course, whether the sound of that low end is "smoother" is largely subjective. True omnis (as opposed to dual-diaphragm mics that achieve an omnidirectional pattern by summing the signals from the two diaphragms) are known for having a very accurate low-end response, regardless of distance from the source. But accuracy and a "big" low end are often two different things.

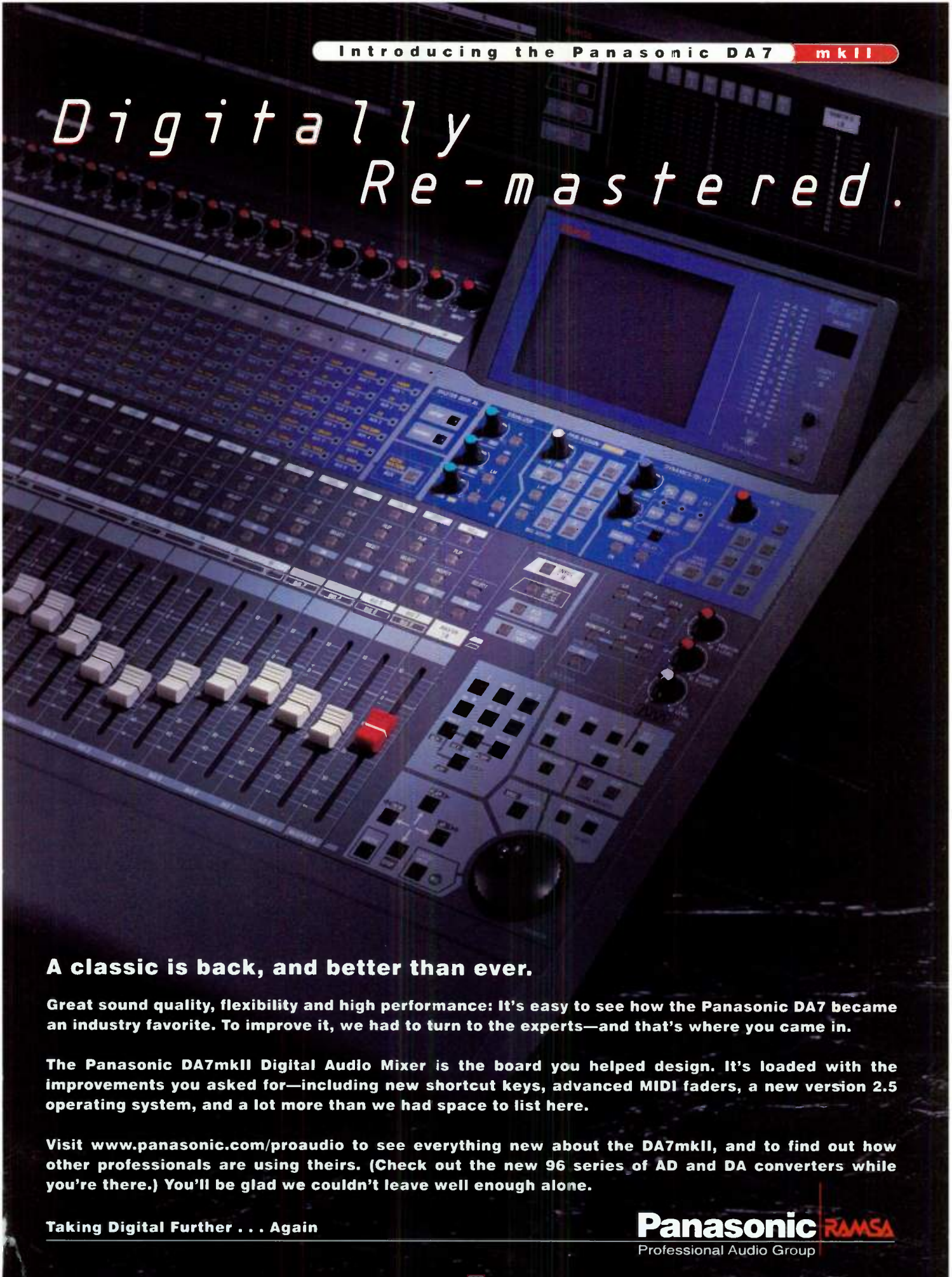
The "myth" that large-diaphragm mics have a more extended low-frequency response probably issues from the fact that large-diaphragm cardioid condensers are often used to record vocals at very close range: say, one to three inches from the grille. At this distance, bass boosting from the proximity effect is typically quite pronounced, which could lead to the perception that the mic has a "big" low end. Furthermore, coloration resulting from other design features (tube circuitry, for example) may make certain large-diaphragm condenser mics more desirable for recording vocals than others.

In fact, small-diaphragm cardioid condensers tend to exhibit more bass boosting from the proximity effect than large-diaphragm cardioid condensers. Because they are more susceptible to popping—as you noted—they are less commonly used for miking vocals at close range.

As for whether Stackpole's review "perpetuates a myth," note that she correctly attributes the claim to the manufacturer, then merely goes on to examine that claim.—Brian Knave

WE WELCOME YOUR FEEDBACK.

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



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WHAT'S NEW

By Marty Cutler

▲ MEDIAFORM REFLECTION

The Reflection from MediaForm ($1,299) is a standalone, one-to-one CD-duplication system targeted at personal studios with modest duplication needs. The desktop unit weighs less than 15 pounds. You can do one-off CD copying of any audio and data format at 12× speed. Depending on file size, the Reflection can burn up to 70 CDs in an hour.

Although it is a standalone unit, the Reflection's audio track-extraction feature lets you compile CD tracks from multiple sources. It can also resume writing to a CD after buffer underruns; the unit will continue to write the CD from the point where the error occurred, saving your CDs from becoming coasters. MediaForm; tel. (800) 220-1215; e-mail info@mediaform.com; Web www.mediaform.com.

▶ CARVIN CM98ST

Carvin's CM98ST dual-diaphragm, multipattern tube microphone ($1,299) is geared toward recording instruments and vocals. The mic features dual, 1-inch, gold-sputtered diaphragms. The capsule and tube preamp are internally shock mounted and enclosed in a machined housing.

A 10 dB pad switch lets you tailor the mic for higher SPLs. The mic also has a low-cut switch for reducing unnecessary low frequencies. The mic comes with a 48V phantom-power supply and a gold-plated 7-pin cable.

The CM98ST boasts a frequency response of 20 Hz to 20 kHz (±2 dB), a dynamic range of 133 dB, and an output impedance of 300Ω. The microphone and power supply come with an external shock-mount suspension, a soft carrying pouch, an XLR cable, and a foam-padded aluminum flight case.

Carvin; tel. (800) 854-2235 or (858) 487-1600; Web www.carvin.com.

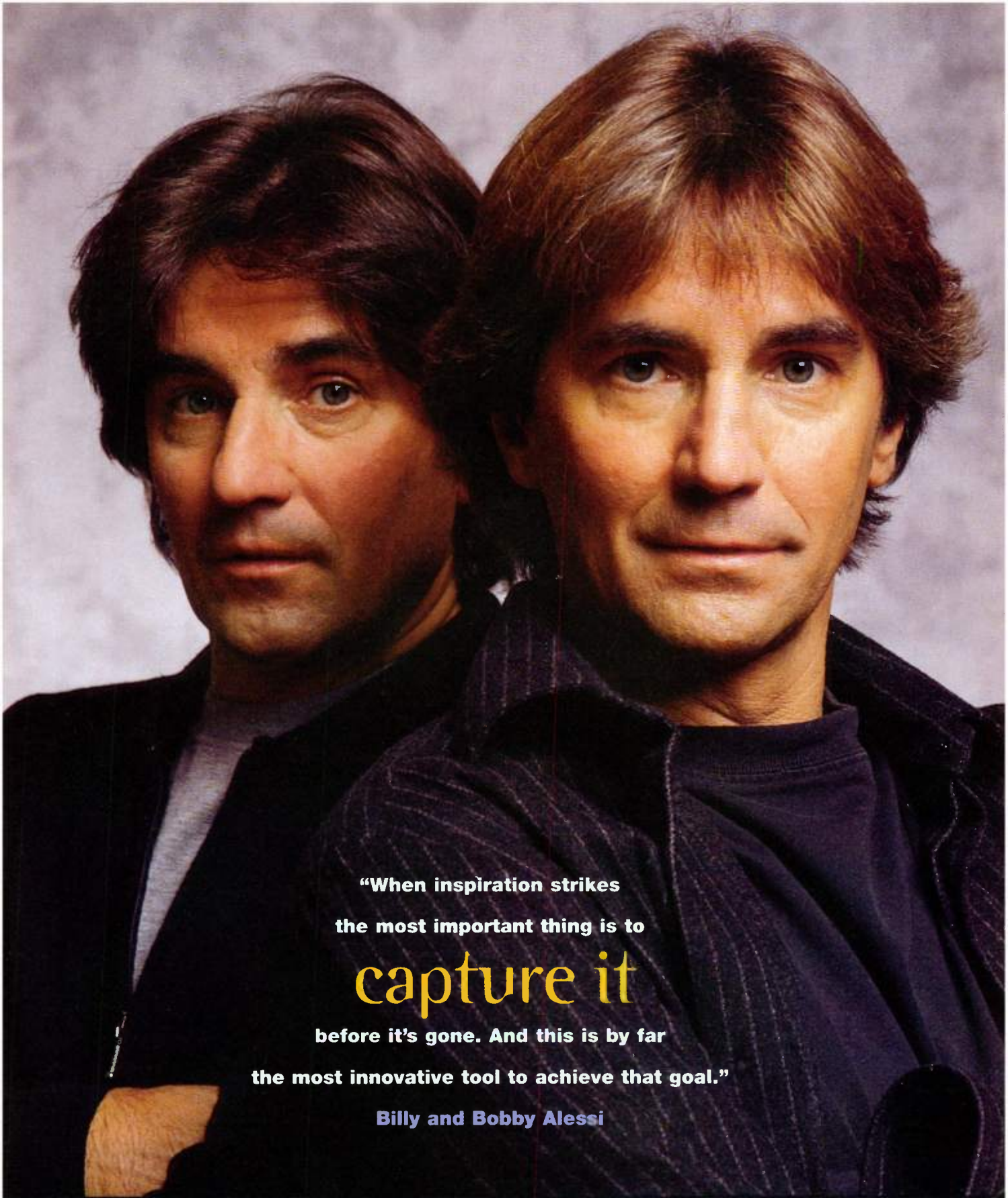
▼ LINPLUG RM 2

LinPlug's *RM 2* rhythm sampler ($45) is a software drum machine for Windows-based, VST 2.0–compatible hosts. Like many hardware drum machines, *RM 2* offers six individual outputs as well as a separate stereo pair. The virtual front panel has mute and solo switches, provides access to other programming parameters, and offers 18 pads for triggering samples. You can even load samples from the front panel without accessing the Files menu.

You can load WAV or AIFF files of up to 32-bit resolution with sample rates up to 96 kHz. *RM 2* can also load kits from LinPlug's earlier *Rupsta* virtual drum machines and from Steinberg's *LM-4* and *LM-9*. You can then export kits as *RM 2*–native files.

The volume, panning, and pitch range (±2 octaves) can be adjusted for each sample. You get Attack, Hold, and Decay envelopes for the volume and pitch of samples assigned to each pad. You can layer as many as 32 samples per pad, and the samples can be mapped to any of 32 Velocity ranges so you can cross-switch between samples.

You can modulate parameters with external MIDI controllers and modulate the parameters using MIDI Control Change messages; the software will automatically "learn" and assign the controller. As a VST Instrument, *RM 2* enjoys the post-processing and automation capabilities of the host software. Timing is sample-accurate. Along with a VST-compatible host program, you will need at least a Pentium II/266 MHz with Windows 95 or 98. RAM requirements depend on kit size and sample rate. Linplug; e-mail support@linplug.de; Web www.linplug.de.



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The PS-02 is a multitrack digital recorder with on-board multi-effects and 200 bass and drum patterns. It even has a built-in mic for recording vocals and acoustic instruments. And it uses the latest SmartMedia technology to capture every nuance of your recording with digital accuracy.

So if you want to start turning your ideas into reality, do what Billy and Bobby do. Get the new Palmtop Studio from Zoom and take your inspiration with you.

To find out more about the Alessi Brothers visit www.alessibros.com. And be sure to listen to the Alessi Brothers' PS-02 demo at www.samsontech.com.

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JAMBIENT.COM JAMBIENT

Turn your computer's joystick into a real-time, improvisational remixing tool with Jambient .com's *Jambient* ($50) software. The program is a software mixer that can load and play up to 16 WAV-format samples.

Each sample has independent controls for volume, pitch, and pan, and you can control those parameters simultaneously with a joystick or other game controller. Apart from panning, you also have 3-D control for sound localization.

You can save *Jambient* mixes in a native file format for exchange with other users and can program scripts to control the program's behavior. *Jambient* requires at least a Pentium/133 MHz, 32 MB of RAM, Windows 95 or 98, and *DirectX* 7.0a. Jambient.com; tel. (416) 603-3309; e-mail omnivore@sympatico.ca; Web www.jambient.com.

FUTURE RETRO MOBIUS

Although the manufacturer touts the Future Retro Mobius ($325) as a standalone hardware sequencer for TB-303–type synths, the unit's ability to record MIDI and control voltages makes it useful for sequencing MIDI gear along with analog synthesizers. You can sequence in real time, use Mobius as a step-sequencer, or simply use Mobius as a MIDI-to-CV converter.

The 3U Mobius offers 256 patterns on a single track. Each pattern can record notes, duration, glide, Velocity, and more; each can also have a different time signature and loop point. You can edit patterns on the fly; the Mobius will automatically save your changes. Editing features include copy and paste, transpose for each step, pattern shifting, and multiple pattern cueing.

Patterns are arranged into any or all of 16 songs, and each song can hold up to 3,580 measures of music. Patterns linked to a song retain their original time signatures and loop points; however, you can transpose patterns from the original key. The Mobius supports System Exclusive dumps of patterns and songs for offline storage.

The rear panel includes MIDI In, Out, and Thru, and a DIN Sync output. The front panel has ¼-inch unbalanced outputs for volts/octave CV, Hz/Octave CV, a 12V Gate out with switchable polarity, a 12V trigger out, 5V Accent, and 5V clock out. You also get a 5V Clock Reset Out for resetting analog sequencers or frequency dividers. The Mobius can sync to external MIDI Clock or serve as the master device, providing MIDI Clock, DIN Sync, and analog clock signals simultaneously. Future Retro Synthesizers; tel. (785) 827-9278; e-mail info@future-retro.com; Web www.future-retro.com.

ZARG MUSIC ORION

The Zarg Music *Orion* is a software-based, 2-oscillator, analog-modeling synthesizer ($169) offered by noted sound designer John Bowen for Korg's OASYS sound-design workstation.

The synth provides amplitude modulation, hard sync, and filter overdrive. Oscillator 1 can be pulse-width-modulated from any of three sources: LFO, oscillator 2, or the filter envelope. You can choose which of the two oscillators is carrier or modulator for amplitude modulation, and AM can be controlled by the AM Mixer or shaped by the filter envelope.

Orion has a virtual switch for selecting one of three audio modulation algorithms: Clip shaves off any values above the maximum setting; Absolute gives you the exact value of the multiplied output; and Shift offsets the output by 1.25 percent to avoid DC artifacts. You can also use oscillators for modulating filters at audio rates. The 24 dB lowpass filter has a dedicated envelope generator with an inverter.

Thanks to the modular design of OASYS, you can apply virtually any number and combination of effects to *Orion*'s output. Zarg Music; tel. (408) 238-4355; e-mail johnbowen@bigplanet.com; Web www.zargmusic.com.

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▼ YAMAHA MLAN

Visitors to the Yamaha booth at AES, NAMM, and Macworld Expo conventions during the past few years are probably aware of mLAN as a developing new standard for communication between computers, electronic musical instruments, and digital-audio devices using FireWire.

Yamaha claims that a single mLAN cable can transmit roughly 100 mono channels of 24-bit, 44.1 kHz digital audio and about 256 times the amount of MIDI data transmitted over a single MIDI cable. Because mLAN uses a single type of connector, integration of MIDI and digital-audio gear is greatly simplified. As with most new standards, its success is dependent on the availability of software, hardware, implementation, and support. The first Yamaha mLAN products should be shipping by the time you read this.

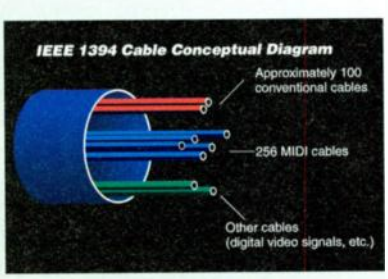
The mLAN8P ($1,495) is Yamaha's first mLAN product. It lets you connect conventional MIDI and stereo analog- or digital-audio devices to an mLAN network through FireWire. You can connect digital-audio gear through eight channels of S/PDIF I/O; MIDI devices hook up through one MIDI In and two independent MIDI Out ports. On the FireWire side, input and output are combined in a single connector. The mLAN8P has three such FireWire ports, letting it carry 8 audio channels and 48 MIDI channels. The audio is fed to a 12-channel, 4-bus digital mixer that is roughly equivalent to Yamaha's 01V digital mixer.

In addition to the FireWire ports, you get two ¼-inch, unbalanced analog line in and out jacks, a pair each of coaxial and optical S/PDIF I/O, one MIDI In and two MIDI Out ports, a serial To Host computer port, and a stereo headphone jack. A headphone volume control, A/D input gain control, and rotary encoder are provided.

You can install the mLAN8E ($699.95) in Yamaha's S80, CS6x and CS6R synthesizers, and A4000 and A5000 samplers. The card provides up to 16-input internal mixing capabilities for supported devices, and connectors include three 1394 ports for 8 channels of digital-audio I/O and 16 MIDI channels.

Yamaha has not left digital mixers out of the picture; you can install the CD8-mLAN interface card ($599) in 02R and 03D mixers for eight channels of digital-audio I/O. The interface provides 16 channels of MIDI I/O for the 03D but does not offer MIDI interfacing for the 02R.

The hardware is bundled with patch-bay and mixing software along with drivers for ASIO and Open Music System. Yamaha is negotiating with other manufacturers to provide components for mLAN-compatible gear. The Korg Triton Rack (Korg USA, Inc.; tel. (516) 333-9100; Web www.korg.com.) has a slot for an mLAN connector, and you will be able to add Korg's EXB-mLAN expansion board, which is expected to ship in March. The expansion board will offer mLAN output through the unit's FireWire connector. Yamaha Corporation of America; tel. (714) 522-9011; e-mail info@yamaha.com; Web www.yamaha.com or www.yamahasynth.com.



COURTESY YAMAHA

▲ CREAMWARE SCOPE/SP

Scope/SP ($4,595) is a software synthesis, sampling, effects processing, and hard disk recording system comprising a PCI card and software. You can use Scope/SP in conjunction with practically any sequencer or digital-audio program or as a standalone system.

The standard system features two ADAT Lightpipe interfaces, stereo analog and S/PDIF digital I/O, MIDI In and Out, word clock, and ADAT 9-pin sync. You can expand the I/O and even add DSPs if the PCI card's stock 15 SHARC chips aren't enough.

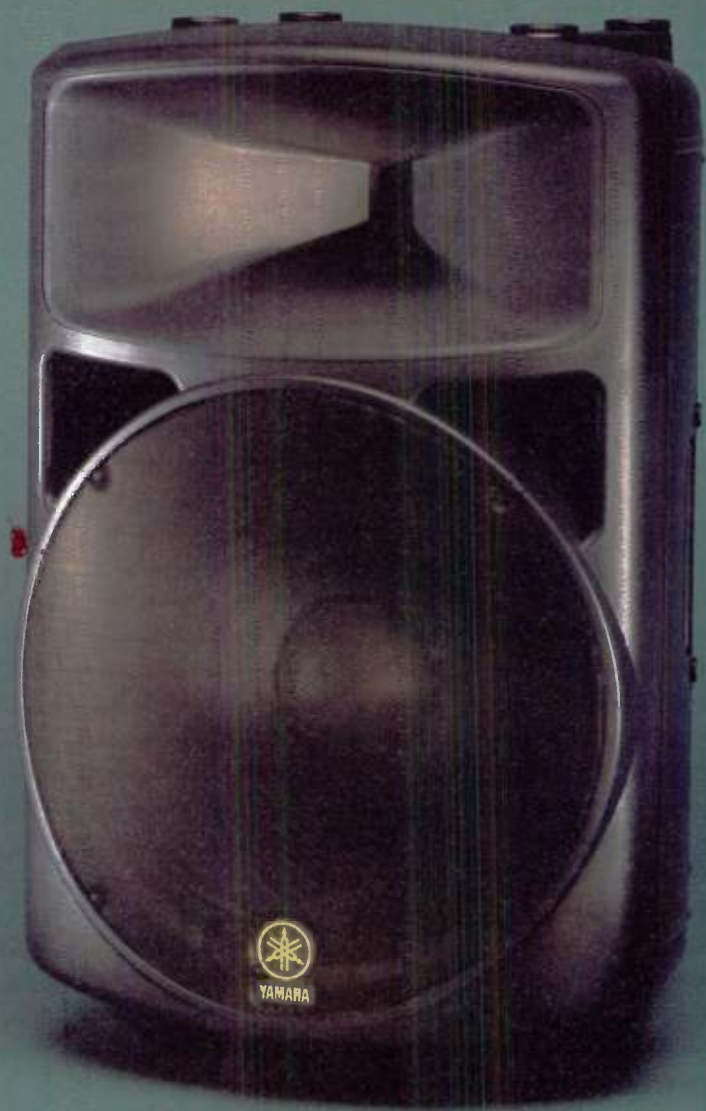
Software includes *SP-Mix,* a virtual mixer application offering 48 channels (with four effects inserts each), 8 buses, and 8 aux sends and returns. The mixer has a freely configurable 4-band EQ and a compressor with a sidechain.

STS-5000 is a software sampler offering real time time-stretching, pitch-shifting, and formant correction. You get 25 different filter types and up to 64-voice polyphony. The sampler can deliver 32-bit resolution samples at a sampling rate of up to 96 kHz.

VDAT and *VRC-128* are a virtual ADAT (recording program) and virtual BRC (control program), respectively. They use the card's ADAT I/O and 9-pin connections to integrate virtual ADAT tracks with external ADAT units. You can record up to 128 16-bit tracks (or 96 24- or 32-bit tracks) and lock the tracks to your hardware ADATs using *VRC-128.*

You get more than 12 different software synthesizers, including virtual analog, analog modular (with more than 140 modules), and FM; additional mixers; a 128-channel meter bridge; and more than 40 DSP effects. With Scope/SP's SHARC DSPs, multiple software devices can be configured and used without latency or heavy CPU drain. CreamWare US; tel. (800) 899-1939 or (604) 435-0540; e-mail info@creamware.com; Web www.creamware.com.

MORE THAN JUST A PRETTY FACE.



Introducing the Yamaha MS400, a complete PA system in a speaker. Its two amplifiers pump out 400 watts of power – 100 to the 2" HF driver and 300 for the 15" neodymium woofer. An on-board mixer – with independent mic and line inputs and EQ – means you get instant music once you plug in a guitar and microphone. Then figure in the multiple mounting options, the ability to daisy-chain speakers and a great-looking and rugged enclosure and what do you get? The Yamaha MS400: A pretty face – and so much more.

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Guitars include vintage Gibsons, Fenders, Martin, Lap and Upright Slides. Great playing, recorded to ILIO spec! Put the sauce of real guitar in your remixes and tracks!

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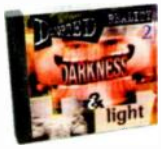
SPECTRASONICS

Distorted Reality 2
"Darkness & Light"



The completely new adventure in sonic dementia from producer **Eric Persing** is here at last! DR2 was created with the very latest in sonic sculpturing technology. Spinning metallic textures, singing oceans, vast amounts of unique loops (50-480 BPM!), impacts, pads, subsonic low-end and the many joys of feedback only begin to describe the wonderful noises in this collection. If you are a fan of volume 1, your sonic arsenal is simply incomplete without "Darkness & Light."

"Spectrasonics had a lot to live up to with DR2, and they didn't disappoint!"
–KEYBOARD ***Awarded Key Buy!***

★★★★★ – *SOUND ON SOUND*

CD-Audio $99

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Roland, Akai/Emu, SampleCell, Kurzweil

Hans Zimmer Guitars
Volume 2

This brilliant follow-up to Volume 1 raises the bar in recording, sampling and programming techniques, perfected by veteran sound designer, Bob Daspit. These instruments feature velocity switching for both up and down slides, and an intuitive new method for controlling vibrato with dynamics. It's beautifully programmed and a thrill to play. Volume 2 features Jazz Hollowbody Guitar, Lap Steel Slide Guitar, Electric Baritone, Flamenco Nylon, and a killer new Steel String Acoustic. Plus, check out the rare Middle Eastern, South American and other world instruments like the Charango, Mandolin, Cavaquinho, Cuatro, Oud, Saz, Electric Sitar, Cumbus, and Yayli Tambur. All are amazingly captured in this stunning new release!

★★★★★ *"... a must-have disc for anyone needing first-class plucked instrument sounds... this collection deserves all five stars."*
– SOUND ON SOUND

CD-Audio $ 99
CD-ROM $299
Roland, Akai/Emu, Kurzweil, SampleCell

Bizarre Guitar

Sound designer **Eric Persing** and guitarist **Peter Maunu** blaze a new trail of sound design via guitar. It features the extreme contrasts of breathtakingly beautiful **ambience beds and searing harmonic slashes**, bowed **guitar, strange funk and unsettling washes.** Created with tape loop techniques, e-bow, the Roland VG-8, and cutting edge signal processing tricks, you won't believe your ears. The CD-ROM version includes selected Groove Control™ loops.

"...The range of sonic ideas is vast...brings to mind the dreamlike soundscapes of Fripp and Eno...an aural addict's dream come true."
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"...meticulous attention to detail... perfectly suited to ambient and film music...Bizarre Guitar will appeal to those who like Distorted Reality, providing a wealth of strange and beautiful sounds ."
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CD-ROM $199
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Roland, Akai/Emu, Kurzweil, SampleCell

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NEW

Spectrasonics takes Retro into the future! Producer **Eric Persing** has created a collection of Classic live drumming with serious feel, vintage sonics and a huge variety of original drum grooves and fills from four incredible drummers! These authentic sounding loops are ultra-flexible, bringing that magic "vibe" to any modern track. And with Groove Control, the grooves are at your command, effortlessly blending their timing and human feel to match your music.

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★★★★★ *"... amazingly useful... infectiously musical... a fantastically wide range of drum sounds... amazingly tight and powerful...'must sample' sound quality... one of the best!"* – *SOUND ON SOUND*

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Burning Grooves

Finally, a drum library that delivers maximum impact. Killer kits & blazing live grooves laid down by L.A. flame-thrower **Abe Laboriel Jr.** (k.d. Lang, Seal, Jeff Beck), with creative re-mixes by producer Eric Persing. If you're looking for attitude, search no further!

★★★★★ *"A slammin' combination of drum kits & loops... Sizzling cymbals & punchy drums... Expressive & realistic kits... killer beats... rock-solid playing... Excellent."*
–KEYBOARD
"The quality of both the playing & recording is first class... Impressive!"
–SOUND ON SOUND
★★★★★ *–ELECTRONIC MUSICIAN*

CD-Audio $99 (grooves only)

CD-ROM $199
(grooves and kits)
Roland, Akai/Emu, Kurzweil, SampleCell

Liquid Grooves

Now, more "liquid" than ever! The unique and inspiring grooves from the timeless library **"Liquid Grooves"** are now available in a Groove Control™ version. This revolutionary new method allows you to mix and match grooves in any tempo, and will follow tempo changes from your sequencer! Groove Control also allows you to change the pattern, feel and pitch of the rhythms to your music. Imagine being able to tune the Wavedrum™ grooves to the key of your song in real-time, without changing the rhythm! Program your own original patterns and fills with the same sounds and mixes from this library!

★★★★★ *"The standard of the sounds and their rhythmic content can't be praised enough, and making all this available in the Groove Control format has extended their usefulness enormously."*
– SOUND ON SOUND

CD-Audio (*without* Groove Control™) $99

CD-ROM (*with* Groove Control) $199
Includes sampler CD-ROM and Data CD
Roland, Akai/Emu, Kurzweil

Call about our special Upgrade price!

Bass Legends

Marcus Miller, John Patitucci & Abraham Laboriel present their coveted electric and acoustic basses & grooves on this critically-acclaimed 650 meg collection.

TRIPLE ★★★★★ *review*
"...a truckload of great multisampled basses ... Definitely put this one on your list of must-haves!"
–KEYBOARD

"...superb... well organized, exquisitely recorded, and brilliantly performed."
–ELECTRONIC MUSICIAN

CD-Audio $99 (grooves only)

CD-ROM $299 (basses and grooves)
Roland, Akai/Emu, SampleCell, Kurzweil

Symphony of Voices

A massive FOUR CD-ROM SET, which includes a variety of multisamples, phrases & fx from some of the world's most prestigious singers.

- **The London Chorale.** 80 voice choir with multisamples and fx
- **English Boys' Choir.** solo and ensemble multisamples and phrases
- **Classical Soloists** Operatic tenor and soprano phrases and expressive multisamples of each soloist
- **Gregorian Men's Choir** Multisamples and ancient Chants
- **Lush Multitracked Pop Stacks** "Enya-esque" pads

"... simply stunning... without a doubt the ultimate library of choral samples." –KEYBOARD 10 of 10 rating. ***Awarded Key Buy.***

★★★★★ *"... the definitive vocal sampling work."*
–SOUND ON SOUND

CD-ROM $499 4-disc set
Roland, Akai/Emu, Kurzweil, SampleCell

Vocal Planet

NEW

Spectrasonics is proud to announce that the eagerly anticipated follow-up to Symphony of Voices is now available! Vocal Planet is an epic Five Disc CD-ROM library produced by Eric Persing, that features both multisamples and phrases, an awesome variety of over 12,000 samples of authentic male and female singers. These are incredible one-of-a-kind sounds. Best of all, the samples have all been pre-cleared for your musical use without any additional licensing fees!

The main categories include **Gospel** with choir multisamples, shouts, hums, praises and phrases plus soloists - **Jazz/Blues** with Take-6 style vocal ensembles, scat singers and Delta Blues men – **World** with Tuvan throat singers, Colombian, Peruvian, Brazilian, Native American, Rasta, Indian, Serbian, Mid-East, Nordic, Balkan, Celtic, much more – **R&B/Dance** with Chromazone background licks, soulful phrases, dance hooks and shouts, "street corner" acapella, human synths, thousands of vocal percussion hits and **Groove Control** loops!

More than EIGHT HOURS of wall to wall vocals. Vocal Planet is the ultimate way to "humanize" your tracks!

CD-ROM $399 5-disc set
Akai/Emu, Roland, Kurzweil
Complete library includes Data CD with Soundfinder system

CD-Audio $149 3-disc set
Selected phrases

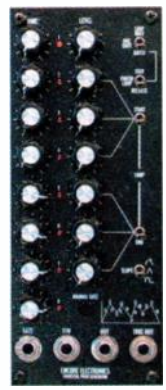
ENCORE ELECTRONICS UNIVERSAL EVENT GENERATOR

Encore Electronics' Universal Event Generator ($289) can serve as an eight-stage envelope generator, a step sequencer, or an LFO. Pots on the front panel shape the envelope. You get eight pots to control rates and seven for levels. Each rate stage can be as long as eight seconds.

In Loop Only mode, the Unit can function as an LFO. One-Shot mode moves through the eight stages once. In Gated mode, the unit operates as a step sequencer and can cycle stages 2 through 7 while the gate is on.

With the Finish Loop feature, the envelope will cycle to the end of the loop regardless of when the gate stops. Release mode jumps from stage 2 directly to stage 8.

You also get a button for manually gating the unit, and you can use the ¼-inch, unbalanced Gate input to apply control voltages from external units. The Universal Event Generator sports a time-control voltage input, a 0 to 5V output, and a 5V trigger out—all on ¼-inch, unbalanced jacks. Encore Electronics; tel. (925) 229-8875; e-mail sales@encoreelectronics.com; Web www.encoreelectronics.com.



TL AUDIO VP-1

TL Audio's VP-1 single-channel voice processor ($2,499) combines a tube preamplifier, a selectable tube or optical compressor, and a parametric tube EQ in a single unit. Additional features include an expander/gate, a de-esser, a peak-limiter, and an option for adding word-clock input and AES/EBU and S/PDIF digital outputs.

The processor has seven tube stages. A Siemens EF86/6267 pentode handles the first stage and is followed by six Sovtek ECC83/12AX7A tubes: one in the secondary preamp stage, one for the compressor, and four for the EQ section.

The preamp accepts mic, instrument, or line levels through a mic input and a +4 dBu line input with balanced XLR connectors. The unit also includes two ¼-inch, unbalanced inputs: one on the front panel, and one on the rear. You get a transformer-balanced mic input with a choice of tube or class-A signal paths. The unit also offers defeatable phantom power, a 30 dB pad, phase reversal, and a variable high-pass filter.

The expander/gate section offers control of threshold, attack, and release. With the de-esser, you can control depth, frequency, and bandwidth. You have a choice of optical or tube-compression stages, and controls are available for threshold, ratio, attack, release, and gain makeup. You can choose hard- or soft-knee compression. A switchable Hold circuit reduces low-frequency distortion. There is also a peak-output limiter.

A four-band tube EQ section is switchable to pre- or post-compressor and features low- and high-frequency shelving with fully parametric mids.

The master section has output fader control and VU metering of input, output, and gain-reduction levels. Two VP-1 units can be stereo linked. HHB Communications USA (distributor); tel. (310) 319-1111; e-mail sales@hhbusa.com; Web www.hhbusa.com.

ROLAND V-PRODUCER

Roland's *V-Producer* ($995; Mac/Win) provides visual feedback for arranging, processing, and mixing AIFF or WAV audio files destined for a VP-9000 Phrase Processor.

V-Producer's drag-and-drop interface lets you sequence audio files into a complete song; a Preview feature lets you audition files from the Main window. The Phrase Scope window provides a piano-roll screen for accurate visual editing of pitch, time, and formants. You can also edit VP-9000 loop points.

Mixer and Effects windows provide control over volume, internal effects, and signal routing. When you finish looping, mixing, and processing, the V-Trainer feature lets you perform batch-process encoding of audio files for use in the VP-9000. You then save encoded audio to a Zip disk for loading into the hardware.

V-Producer can be locked to a sequencer using MIDI Clock or MTC. You can save complete *V-Producer* arrangements, which includes saving Phrase Scope parameters as Standard MIDI Files. The software offers control of up to six VP-9000 units from your computer.

Mac users need at least a Power Mac 604e/233 MHz with Mac OS 8.6 or later, 60 MB of RAM, and an Iomega Zip drive for shunting files to your VP-9000. Windows users need at least a Pentium II MMX/266 MHz processor, Windows 98, 64 MB of RAM, and a Zip drive. Of course, you will need a VP-9000 for playback of encoded audio. Roland Corporation U.S.; tel. (323) 890-3700; Web www.rolandus.com.

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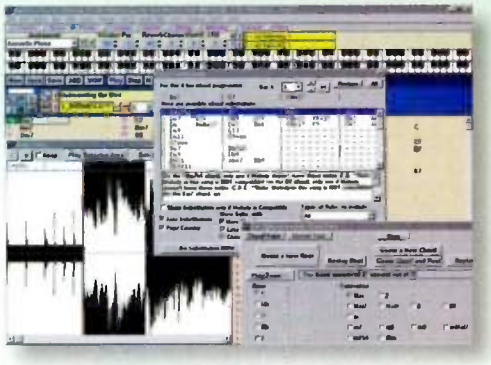
REV UP ▲▲▲▲

▶ PG MUSIC



Version 10 of PG Music's Band-in-a-Box for Windows ($88; upgrade from Version 9 $49; upgrade from earlier versions $59) offers 50 new enhancements and features. Enhancements include changes to both MIDI and digital-audio features of the program.

New MIDI features include guitar-oriented styles that offer a choice of pop, jazz, country, or folk guitar playing with realistic picking and strumming. A new Guitar Tutor feature illustrates authentic guitar-chord voicings for any of the available Styles. You also get a Chord Substitution Wizard that provides alternate chords for any progression.

The Live Harmonies feature was enhanced; even when the "transport" is stopped, the program follows and harmonizes with your performance in real time. A new Ear Training Window provides a host of chord and interval recognition exercises.

New digital-audio recording upgrades include *DirectX* plug-in support and an Audio Edit window where you can overdub and perform general edits such as copy, cut, and paste. You can burn digital-audio tracks to CD directly from the program or save tracks in Windows Media Format for uploading to your Web site. Printing enhancements include the addition of a Print Preview and a Print to JPG feature. PG Music; tel. (250) 475-2874; e-mail info@pgmusic.com; Web www.pgmusic.com.

MACKIE

Version 3.0 software for Mackie's D8B digital 8-Bus mixer is available as a free download from the Mackie Web site. The new software adds support for third-party plug-ins, new surround-mixing features, expansive networking options, and a number of user-requested features.

The D8B's support for third-party plug-ins lets the board run any plug-in designed for the Mackie Realtime OS. Available plug-ins include Antares' *AutoTune,* Massenburg's *MDW2x2* EQ, TC Electronic's *TCFXII Reverb,* IVL's *Vocal Studio,* Acuma's *Filter Machine,* and plug-ins from Drawmer. You will need to spring for a Mackie UFX card ($499) to run the plug-ins, but Mackie bundles TC Electronic's *TCFXII Reverb* and Mackie's *Mono* delay with the card.

Responding to user requests, Mackie added a Cross Patching feature that lets you customize each fader bank. For example, you can move controls for a channel's effects return so that the controls are adjacent to the track's dry channel. A new Dual View function lets you view two fader banks simultaneously; Return channels are displayed at the top of the screen, and the selected fader bank is at the bottom. Other requested upgrades include a retooling of the channel dynamics processors for a smoother, more natural sound.

New networking features include the ability to define as many as eight different outboard devices that you can control from the D8B, such as additional D8B units and Mackie's HDR24/96 hard disk recorder.

Version 3.0 software adds three major additions to D8B surround capabilities. The Low Frequency Enhancement control lets you adjust gain from individual channel's output to the subwoofer. A Depth of Center control adjusts each channel's gain in the center speaker, and a global Pan View graphically displays an overview of surround panning for channels 1 through 72. Mackie Designs; tel. (800) 898-3211 or (425) 487-4333; e-mail sales@mackie.com; Web www.mackie.com.

▶ EARTHWORKS M30BX

The Earthworks M30BX ($600; $1,100 for a matched pair) was originally intended to be a high-quality measurement microphone, but it is useful for recording, too. The condenser mic does not need phantom power, offers enough gain to plug in directly to most sound cards, and comes with a special adapter to connect to a sound card's RCA input jack.

The omnidirectional M30BX is powered by a single 1.5V AA battery, which can last more than 1,500 hours. An internal switch automatically disconnects the battery when the unit is not connected to an XLR cable. To eliminate noise due to possible intermittent battery connections, the mic features a 15-second power buffer. You can boost the mic's gain to 14 dB with an easily accessible internal switch.

The M30BX's frequency response is rated at 9 Hz to 27 kHz, ±1 dB, and it can handle 132 dB SPL—and up to 118 dB SPL with the 14 dB gain switch engaged. The microphone comes packaged in a molded plastic first-aid-kit–type case and includes an XLR-to-RCA female adapter and a 1.5V AA battery. Earthworks; tel. (603) 654-6427; e-mail earthwks@jlc.net; Web www.earthwks.com.

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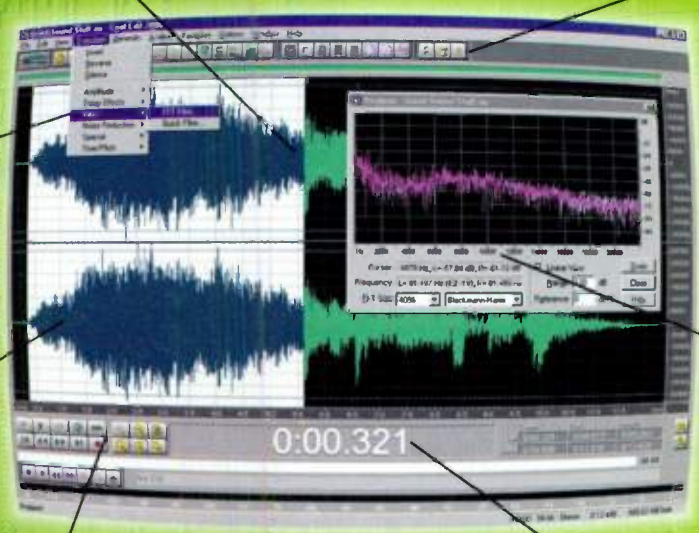
Optional *Tweakin' Toys Plug-In* gives you four fun effects: Hard Limiter, Pan/Expander, Pitch Bender, and Convolution!



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WRH

▲ RPM OPTOELECTRONICS RACKLIGHT

One of the least enjoyable tasks in personal studios or live performance is hunting around equipment racks with inadequate lighting. Conventional lighting systems also introduce heat, and bulbs burn out and need to be replaced. RPM Optoelectronics offers the Racklight ($90), which uses LEDs, thereby eliminating hot lamps and uneven light distribution.

The Racklight's solid-state design eliminates bulb replacement and more easily withstands the rigors of transportation than other systems. You can set the unit up to shine in any of six colors. RPM Optoelectronics; tel. (707) 573-0801; Web www.rpmllc.com.

▼ BENCHMARK MEDIA AD2K+

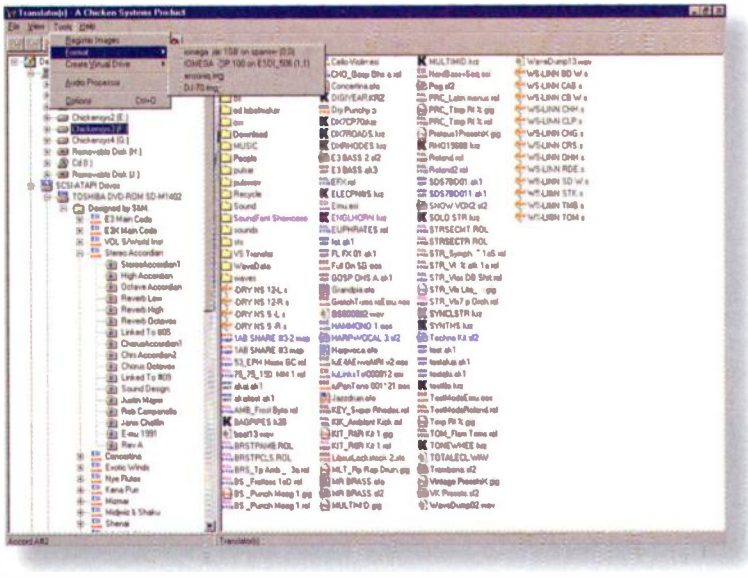
The AD2K+ ($1,795) is a 2-channel, 24-bit, 96 kHz A/D converter designed for use in location recording as well as post-production and mastering. It can run for eight hours on a single 7.2 A-Hr battery. The converter uses special DSP noise-masking algorithms that were developed from psychoacoustic research.

Front-panel controls include input sensitivity, sample rate, and seven word-length reduction settings to ensure compatibility with 16-bit systems. You get a pair of independent postconversion meters for monitoring your signal. A switch scales the meters in 1 dB increments so you can identify individual channel peaks and ascertain your headroom.

The converter has a dynamic range of 120 dB, A-weighted, and offers selectable sampling rates of 44.1, 48, 88.2, and 96 kHz. The output section of the AD2K+ supports four simultaneous signals with the same word length or two signals at 24 bits and two at 16- or 20-bit resolution. Sonic Sense, Inc. (distributor); tel. (877) 324-4463 or (303) 753-0201; e-mail nutter@sonicsense.com; Web www.sonicsense.com.

▼ CHICKEN SYSTEMS TRANSLATOR

Converting file formats between samplers is no picnic; in addition to raw samples, most samplers' native files contain information about keymaps, modulation routings, envelopes, and more, depending on the unit's architecture. Enter Chicken Systems' *Translator* ($149.95), which supports a slew of sampler formats and can read from and write to any sampler's storage device, be it CD, Zip, or hard drive. You can even batch-convert multiple sampler formats to a single format. Where possible, all available modulation routings are converted to the target sampler's architecture. If you only have one sampler, you can buy a limited version of the program ($74.95) that will convert the format of any sampler to your sampler's format only.



The program supports samplers from NemeSys, Roland, E-mu, Kurzweil, Ensoniq, Akai, Emagic, and Korg. Roughly 50 file formats are supported in all. *Translator* presents a directory-tree interface for fast and easy browsing of your samples. You can audition any file directly from disk.

Translator can also create a virtual drive on your computer that you can format for any of the supported samplers. Drag and drop files from one sampler's virtual drive into another, and it will be translated into the new format. You can then burn virtual drives to a CD in the sampler's native format. Additional features include the ability to read files from Mac format disks and the ability to work as a shell program within your choice of sample-editing software. A utility lets you defragment proprietary sampler hard drives.

At the time of this writing, the program was only available for Windows, but a Mac version should be available by the time you read this. *Translator* works with Windows 95 or NT and can even run on a 386/33 MHz PC. You can run *Translator* with the minimal 8 MB RAM requirement; the program's built-in Memory Manager will stream audio to disk when it runs out of available RAM. Chicken Systems, Inc.; tel. (800) 877-6377 or (320) 235-9798; e-mail support@chickensys.com; Web www.chickensys.com. ●

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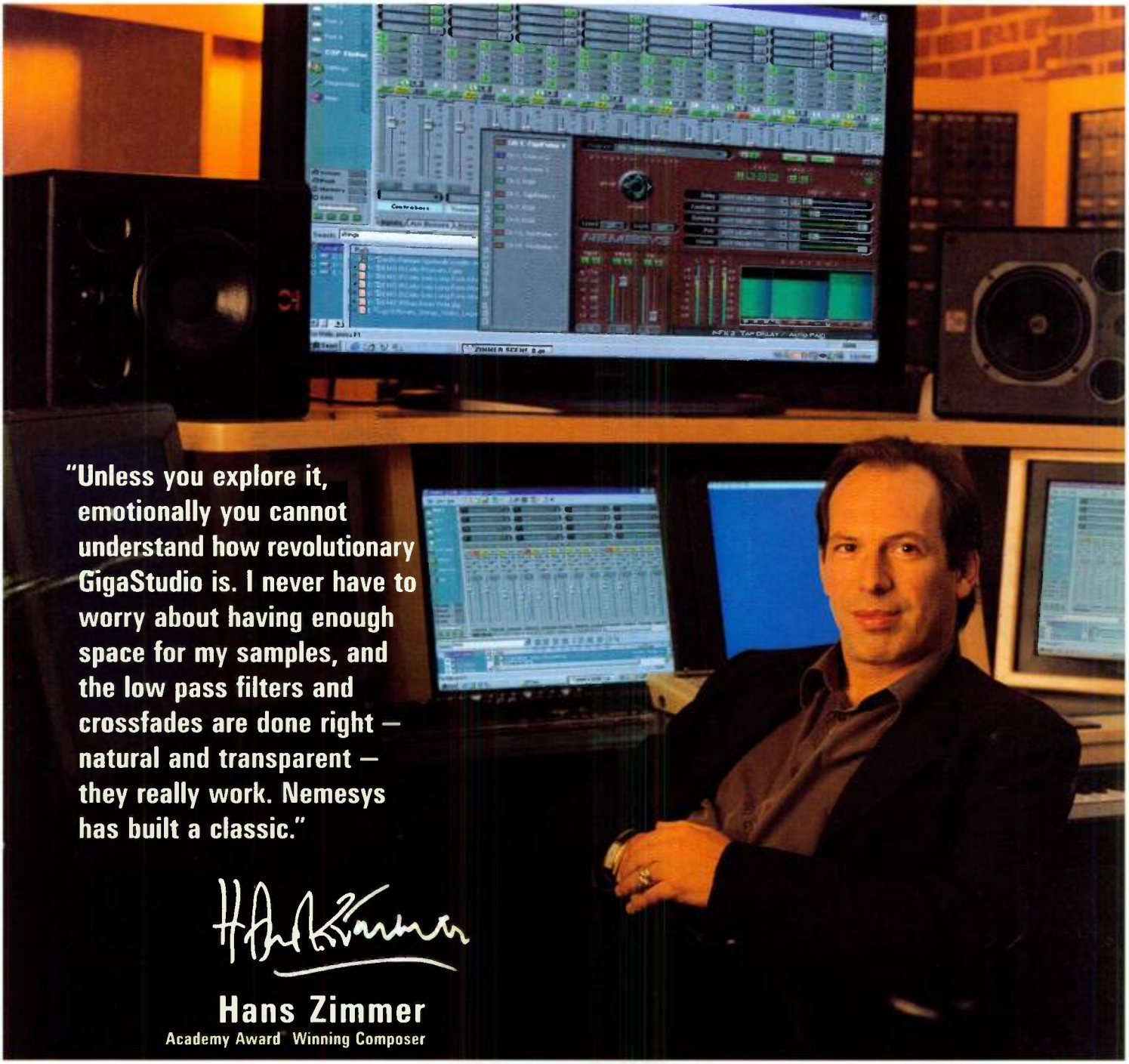

"Unless you explore it, emotionally you cannot understand how revolutionary GigaStudio is. I never have to worry about having enough space for my samples, and the low pass filters and crossfades are done right – natural and transparent – they really work. Nemesys has built a classic."

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WEB PAGE

By Peter Drescher

WEB SITE OF THE MONTH

Mickey Tachibana's quirky and fun Drum Machine Museum (www.drummachine.com) contains a wealth of information on practically every drum machine ever made. Launched in 1998 from his San Francisco–based studio, Tachibana's site features more than 60 classic drum machines. His intention, his site states, is to create a permanent exhibit "dedicated not only to the machines but [to] preserving and archiving items representative of the sociocultural impact that electronic music has had across genres, as well as documenting the artists and music that have profoundly changed the way we all listen and live."

Drum Machine Museum contains photos and specifications for a wide range of instruments, from the venerable Linn 9000 and Roland TR-808 to more exotic instruments such as the Acetone FR-1 and Hillwood HR-2. Even more impressive is the large collection of sounds from the machines themselves. Tachibana posted hundreds of RealAudio streams, MP3s, and WAV files of cymbal crashes, snare hits, and factory-preset patterns. The sounds provide a historical document detailing the evolution of percussion synthesis.

The site also includes forums on a variety of instruments, links to drum machine manufacturers, and articles for drum machine connoisseurs. As a service to owners of older machines, Tachibana maintains a library of manuals. He will send you a photocopy of any manual he has for a nominal fee, and he'll trade you an official Drum Machine Museum T-shirt for any manual not in his collection. If you're in the mood for a little shopping, the Drum Machine Museum store carries new drum machines, effects processors, sample libraries, software synthesizers, and other related gear.

DOTDOTDOT.COM

Until now, if you were a struggling songwriter, you had to deliver your promotional material and demo tapes to music industry executives by hand or through the mail. Enter **SongPitch.com** (www.songpitch.com), a Nashville-based company that uses the Internet to connect record producers to songwriters. Acting as both a database and a secure Internet Service Provider, the site lets producers log on and listen to streaming RealAudio renditions of members' songs, which are cataloged by style and author. The site also provides an Online Interactive Pitch System, so songwriters can sell their work directly to producers looking for a specific type of music. . . . Atlanta's **SongScope.com** (www.songscope.com) provides a similar service. As part of the Vision Music Group, a collection of Internet music marketing companies, SongScope gives songwriters direct access to music publishers, record labels, and A&R people. It also produces compilation CDs, maintains a searchable database of songs and authors, and is affiliated with **The Muse's Muse** (www.musesmuse.com), a resource site for songwriters, and RadioWired (www.radiowired.com), an Internet radio station. . . . **The International Computer Music Association**'s Web site (www.computermusic.org) is a valuable resource for composers, performers, researchers, and developers interested in the integration of music and technology. The site offers a searchable database of musical works performed at the International Computer Music Conference (ICMC) since it began in 1974, portions of the ICMA's online newsletter, *Array*, and details about the 2001 ICMC to be held in Havana, Cuba, in September. Of particular note are the links on the Resources page, which include Academic Institutions, Research Institutions, Composers, Education, and Electronic Musical Instruments and Alternative Controllers.

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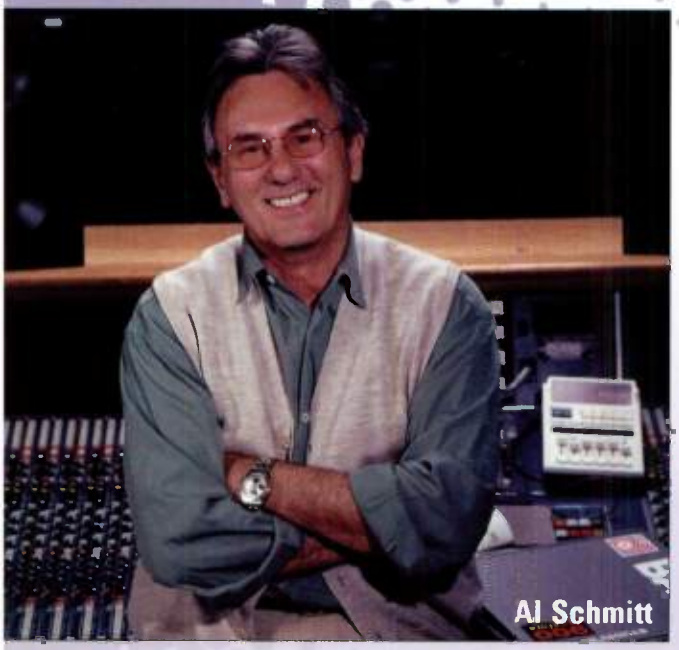

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WEBCAST

The iMusiCast site (www.imusicast.com) is the brainchild of Bryan Matheson, owner of Skyline Studios, a recording and post-production facility in Oakland, California. Matheson leased a large warehouse space next door to his studio and built a venue specifically for broadcasting live performances on the Web. The room contains everything a band needs to produce a video of its show: a large stage, P.A., lights, mics, monitors, video cameras, shooting platforms, and plenty of room for an audience. "People want compelling live content to fill up their broadband pipes," says Matheson, "and we intend to give them that live-music experience."

The Internet magic happens upstairs in the control room. The audio and video feeds from the floor go into a Trinity video production system running Globecaster software from the Play Streaming Media Group. The video director gives instructions to the camera operators, switches views, and posts title graphics like those in a regular television broadcast. The A/V output then goes into three computers (a Mac G4 and two PCs) that send out high- and low-resolution streams in Windows Media or RealMedia formats to a server farm in San Jose, California. From there, the Webcast can be accessed live by any computer in the world. IMusiCast also provides an Internet Relay Chat room so audience members can interact with the performers and each other.

Once a show has been Webcast, it is archived on the site and can be viewed later on demand. So far, the company has Webcast a capella vocal groups, electronica, heavy metal bands, and the ongoing "Acousticast" series, which features local acoustic acts.

DOWNLOAD OF THE MONTH

Allegroassai (www.allegroassai.com), formerly known as Sincrosoft, is an excellent resource for composers, arrangers, educators, and instrumentalists seeking to study and perform pieces from the classical repertoire. The term *allegro assai* means "a very fast tempo," which is an appropriate description for a Web site that offers instant access to sheet music. Allegroassai also maintains a directory of links and news stories of interest to performers and music teachers.

The Italian company produces four cross-platform applications: *Opus,* an inexpensive but powerful music-notation application; *Amadeus,* a lite version of *Opus* that accepts MIDI keyboard input; *Opus Editor,* which lets users edit scores but doesn't accept MIDI input; and *Opus Viewer,* a freeware application for viewing, printing, and playing scores. Scores produced using these programs are saved in the *Opus* format and can be edited or played back like MIDI files. Site visitors can even add fingerings to the downloaded scores and print out high-resolution sheet music for use away from the computer.

The sheer number of pieces available in Allegroassai's online catalog is impressive. Once you subscribe to the service, you can download hundreds of scores—including 385 pieces by J. S. Bach—a number that's growing every day. Single movements from larger works, such as the opening allegro of Beethoven's Piano Sonata no. 15, op. 28, or individual parts from ensemble pieces are also available for purchase. The entire Allegroassai catalog can be searched by composer, title, instrument, or ensemble.

BAND ON THE WEB

"We didn't name the band Banga Tribe after the tribe based in Nigeria," says Paul Bidault, vocalist and guitarist for Banga Tribe. "The word banga appears in many different dialects and has many meanings, ranging from sword to man to water. Our music is similar in that it has many different meanings and styles. We used the word tribe in our name out of respect and admiration for all the ancient cultures of the world."

Based in Mexico City, Banga Tribe (www.bangatribe.com) combines the efforts of Bidault and Pablo Guessi (bass and programming). The two met in school seven years ago and played in a variety of bands before deciding to try their hands at creating more innovative music.

With influences as diverse as Miles Davis, Jimi Hendrix, Peter Gabriel, and Massive Attack, Banga Tribe blends bubbling synthesizers, processed vocals, ethnic percussion, mandolins, and electric guitars. The result is a smooth, somber, mystical sound, like a soundtrack for a thriller set in a Buddhist monastery.

Working out of their PC-based home studio, Guessi and Bidault create their tracks using a Roland MC-505 Groovebox, Boss DR-5, and a variety of effects and acoustic instruments. "These days anyone can make electronic music," says Guessi. "Just hit a couple of keys and the work is done. We believe that when using electronic equipment, the artist's input must be strong, and that the gear is merely a way to convey the music."

Three of the group's tracks—"Seven Stars," "Wind Star," and "Liquid Constellation"—have found success on MP3.com, and Banga Tribe is currently working on a CD that will be distributed through its Web site. "This is a very exciting time to be a musician," says Bidault. "Without the help of big companies or labels, we have managed to get our music heard all over the planet. The Web lets us interact with fans and get their thoughts, complaints, and opinions. It also gives us the chance to meet and collaborate with artists from other countries."

JUAN EDUARDO TRIGUEROS

WEB APP

An attempt to make the Internet experience richer, easier, and more sophisticated, the Extensible Markup Language (XML) is an exciting new Web development.

"XML specifies the syntax, or format, of a document without specifying what kind of information the document contains," says Phil Burk, a developer of interactive music software. Among other things, Burk uses XML to store synth patches for *JSyn,* an interactive computer-music Application Programming Interface (API) he developed using Java. (For more details on *JSyn,* see "Web Page" in the June 2000 issue of **EM**.)

"For instance, XML lets you to define your own tags, so you can mix and match information from different applications and different sources within the same document," Burk says. "This is useful in situations where you want to organize a variety of data types that can be easily parsed. XML gives you the tools to do this whether the data is musical notation or a database."

A markup language is used to design ways of describing information for transmission to, storage in, or processing by an external program. The most familiar markup language to the average Internet user is HyperText Markup Language (HTML). Although HTML allows users to define the format of documents and includes limited hypertext and multimedia capabilities, the language describes only a single class of documents.

By comparison, XML is a metalanguage that allows you to create customized markup specification languages. Like HTML, XML is written in the internationally standardized markup language know as Standard Generalized Markup Language (SGML). "With XML, you don't have to invent your own syntax because it's already defined," Burk says. "By simply conforming to the conventions of the language, you can take advantage of a number of existing tools, like XML parsers."

In fact, the conventions in XML are considerably more rigorous than in HTML. For example, in an HTML file an opening tag, such as , can often have its effect without using a corresponding closing tag, such as . However, in XML, every opening tag must be followed by a closing tag. And unlike SGML and HTML, XML is case sensitive. This means, for example, that the beginning and end tags, <body> and </body>, must match each other exactly in terms of upper and lowercase letters.

Although they may seem more cumbersome, tougher rules such as that give XML its power. They make XML more flexible than HTML while avoiding the programming difficulty of SGML. In fact, XML is merely an abbreviated version of SGML.

Ultimately, XML isn't meant to replace HTML but rather to serve as an alternative for programmers who want to go beyond the limits of HTML. To successfully view XML documents, users need an XML-specific browser. Although SGML and HTML browsers accept XML documents, they do not fully recognize the contents. One option is to use Java to create browser plug-ins that handle XML. To learn more about XML and examine related markup languages such as XHTML, visit the useful World Wide Web Consortium (W3C) site at www.w3.org. The University College Cork Web site at www.ucc.ie/xml also has a helpful XML FAQ.—*Gino Robair*

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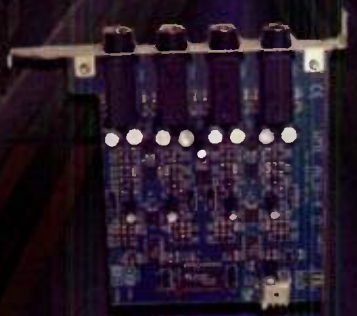

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VINTAGE PAGE

By Julian Colbeck

Hohner Clavinet

Produced: 1971–85
Made in: Germany
Designed by: Ernst Zacharias
Number produced: 100,000
Sound-production system: hammer action, strings
Price new: $700

Today's prices:		
	Like new	*$2,000*
	Like, it's okay for its age	*$1,500*
	Like hell	*$900*

Of all the classic keyboards and synths, none has had greater highs and lows than the Hohner Clavinet. All the rage in the 1970s when Stevie Wonder's "Superstition" catapulted its uniquely spiky, funky sound into everyone's consciousness, the Clavi went into a deeper funk—in the negative sense—in the 1980s; you could scarcely give one away. Even in 1996, the typical secondhand price was less than $1,000. Now a top-quality Clavi can fetch twice that.

The Clavinet began as another curious invention from Hohner staff designer Ernst Zacharias. A man with a quest to modernize and electrify a whole battery of baroque keyboard instruments, Zacharias also designed the Hohner Cembalet and Pianet. The first Clavinet—with built-in amplification—appeared in the early 1960s. Clavinet Model 1, the prototype, recently made its debut at Austria's Keyboard Museum. The first Clavinet was followed by the strange, triangular, three-legged period piece (complete with '60s-style reverse-color keys) called the Clavinet L.

But the D6—with its distinctive light-wood casing, flip-top lid, and screw-in legs—was the big hit. The D6 is a curious contraption: it requires an annoying 9-volt battery, nominally has only one tone, and needs tuning. The pickups, single coil and prone to lots of hum, are embedded in epoxy resin, easily broken, and a bummer to repair. And the D6 weighs a ton (actually 68.35 pounds). The 1979-vintage E7 retained the same innards but sported new Tolex livery and more modern-looking tubular steel legs.

To appreciate what all the fuss is about, you have to play a Clavinet—a real one, not a sample or emulation. Only then will the full weight of this percussive, catchy, edgy, eminently playable instrument become apparent.

You'll quickly notice that the 60-note, firm-but-fast keyboard is not only velocity sensitive but also aftertouch sensitive. (I'm talking about mechanical sensitivity here; this is a pre-MIDI instrument.) Simply leaning hard into a note presses the hammer hard up against a string and (pitch) bends it, as in an 18th-century clavichord. A serious player can even produce vibrato by further waggling the key. But the main advantage of owning an actual Clavinet—as opposed to owning a set of samples or a digital synthesizer with a Clavi tone—is that a whole range of tones is, in fact, at your fingertips.

Magnetic pickups lie at the other end of the string, and six rocker switches on the top panel control the instrument's pickup arrangement and polarity. Those switches—Brilliant, Treble, Medium, Soft, A/B, and C/D—let you concoct a vast array of tones. A slider-operated string damper can let notes ring or progressively muffle them.

Aside from knowing which pickup configuration produces what tones, you don't need a lot of technical skill to play a Clavinet. But sound is produced by hammers hitting strings, so two more skills come into play: tuning and replacing broken strings. You have to check the tuning on a regular basis; a tuning key was originally supplied with the instrument. Although you can replace strings, doing so is somewhat of a drag. Luckily, strings don't break often unless your technique is fairly brutal.

When inspecting a prospective purchase, check the strikers, in case the previous owner was Arnold Schwarzenegger. Under heavy hands, pads are prone to splitting, and split pads require replacement. You can check for splitting easily by gently holding

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The D6 is Hohner Clavinet's most well-known model. The Clavi's roller-coaster ride through keyboard history has reached high peaks as prices have soared to $2,000—more than double the new price 30 years ago.

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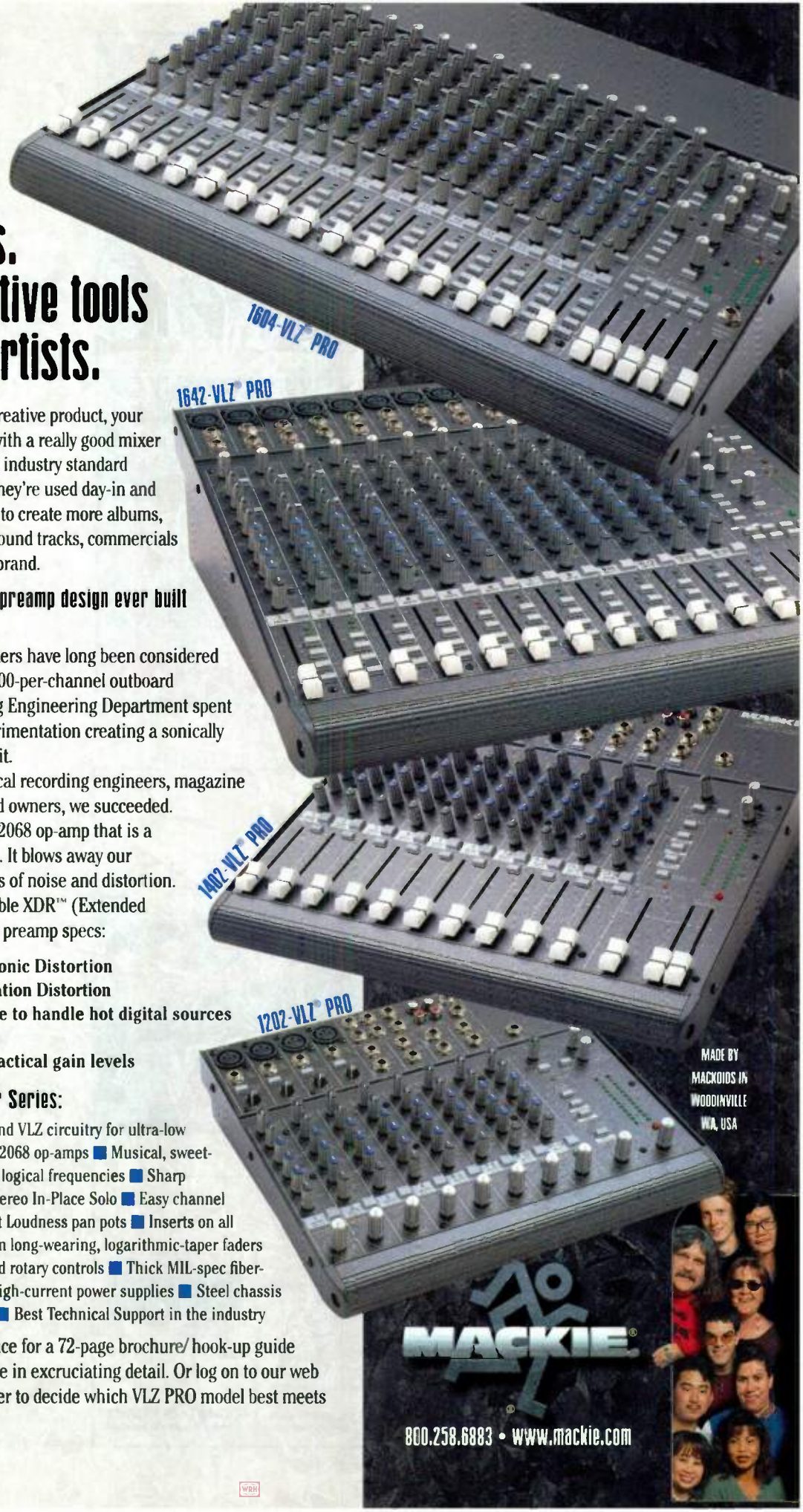

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down each key and making sure it doesn't temporarily stick to the string. Check the Clavi's interior for rusty strings and make sure that the bed in which the keyboard sits is not warped. Cracked pickups are a common problem; fortunately, you can purchase new ones from the Hohner Clavinet Resource Homepage (www.clavinet.com). The new pickups even improve on the originals thanks to their double-coil, humbucking design. An Upper and Lower set sells for $349.99.

Remarkably, the D6 Clavinet design did not change substantially during its long production run. Hohner no longer produces the original rubber hammer tips, but you can buy new tips from other sources, including the aforementioned clavinet.com ($74.99). Hohner resourcefully recommends using O-rings, cut to size, from a Chevy engine as an adequate substitute. You can ease changing a broken string by weaving a plastic drinking straw through the yarn and then passing the string through the straw.

The Clavinet's steely, percussive sound made it the Fender Stratocaster of keyboards. In fact, many guitarists—from Jeff Beck *(Wired)* to John Paul Jones *(Physical Graffiti)* to Frank Dunnery (who actually fretted a Clavinet bed) to Lenny Kravitz—have found it a natural extension of their normal plank-spanking endeavors.

The Clavinet does inspire, if not require, a particular playing style that is heavy on syncopation and feel and light on individual note articulation. Rick Wakeman on the Clavi was never a pretty sight.

Clavinets also benefit enormously from effects, notably wah-wah (if it isn't funky enough already) and phase shifter or chorus. Unaltered, the Clavinet is pretty dry and stark. It doesn't have a sustain pedal, of course. The lone audio out is mono, and though most people generally run a Clavi through a DI box for recording, a Fender bass or guitar amp (such as a Dual Showman or Twin Reverb) or even a Leslie cabinet can add considerable weight to the sound.

Hohner ceased dealing in Clavinets years ago, but you can obtain spare parts and service from companies that specialize in vintage-instrument repairs. On the Web, the top source is www.clavinet.com.

Julian Colbeck *has toured everywhere from Tokyo to São Paulo with artists as varied as ABWH/Yes, Steve Hackett, John Miles, and Charlie.*

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By Scott Wilkinson

Out of My Head

Dolby elicits multichannel sound from stereo headphones.

Multichannel surround sound is becoming increasingly important, not only for movie soundtracks but for music recordings. Of course, you need a playback system with at least six channels of amplification and corresponding speakers to hear these multichannel music disks and movie soundtracks in their full glory. But what if you want to listen to such a disc while your spouse sleeps in the next room—or even next to you in bed? With a stereo source, headphones are the obvious answer. But how can stereo headphones deliver multichannel sound?

Several companies have tried to address this problem without much success. I've listened to some of those companies' systems and been quite underwhelmed—until now. Lake Technology (www.laketechnology.com), an Australian R&D firm, has developed a sophisticated DSP algorithm that simulates the surround-sound experience astonishingly well, using any pair of stereo headphones. Lake has licensed the technology to Dolby Labs (www.dolby.com), which in turn licenses it to manufacturers under the name Dolby Headphone.

I first heard a Dolby Headphone demonstration at the 2000 Audio Engineering Society convention in Los Angeles. A small home-theater system was set up with five speakers and a subwoofer, in addition to a set of headphones for each seat. A short video presentation explained the system's basic principles and played various audio clips in stereo and multichannel mode for comparison. During the first multichannel clip, I took off the headphones to see if the external speakers were on; at that very moment, the pre-recorded announcer said, "No, the speakers aren't on!" The rich, open sound seemed to come from all around outside my head. In contrast, the stereo version seemed quite harsh and strident inside my head.

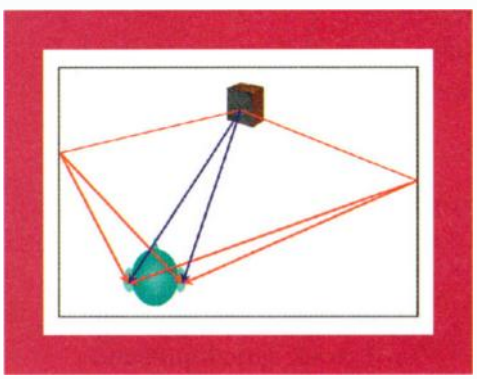
How can the multichannel experience be simulated with conventional stereo headphones? The Dolby Headphone algorithm models the acoustic behavior of five speakers and a listener in a room. It includes several primary elements: inter-ear arrival time (the difference between the moments at which a sound arrives at each ear), head-related transfer functions or HRTFs (which describe how a sound's spectrum changes as it diffracts around your head), and room response, including early reflections and reverb tails (see **Fig. 1**).

The acoustic path from each virtual speaker to each ear is simulated by a 7,000-tap finite impulse response (FIR) filter, which is said to be 350 times more precise than those used in previous headphone-surround systems. Ten such filters are required to reproduce the effect of five speakers placed around the listener. That many large filters normally require excessive processing power and introduce pronounced latency; Lake's new convolution techniques eliminate these problems.

Lake has coded the algorithm for several general-purpose DSP chips, including the Zoran ZR38650, Analog Devices ADSST-MELODY-1000, and Motorola DSP56300 family. In addition, the company has developed core code for application-specific integrated circuits (ASICs) dedicated to Dolby Headphone, which Sanyo Semiconductor recently announced it will manufacture. Finally, the algorithm can run as software on a PC with an Intel Pentium II or III or an AMD K6-II CPU running at a clock speed of at least 400 MHz.

The obvious application is to include a Dolby Headphone switch on consumer A/V receivers and surround processors; hit the switch, and the five main channels are routed through the algorithm to the headphone output. (Because bass frequencies are not directional, the ".1" LFE channel is passed unaltered to both sides of the headphones.) Other potential applications include game consoles and computers.

Dolby Headphone is set to become an important part of the emerging multichannel-audio market. I look forward to using it while my wife slumbers beside me. ❦



COURTESY DOLBY LABS

FIG. 1: Dolby Headphone simulates inter-ear arrival times, head-related transfer functions, and early reflections within a room for each virtual speaker.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PROFILE

By Mark Smith

Metal Machine Music

Chas Smith coaxes haunting music from all things metallic.

If there's one thing composer and instrument builder Chas Smith knows, it's that nothing captures the songs in his head like cold metal.

Welder, machinist, and musician, Smith transforms his affinity for metal into strange creations with names such as "PezEater," "Mantis," and "DADO," alien monikers that suit the otherworldly sounds they emit. Like famed composer and instrument builder Harry Partch, whom Smith readily acknowledges as an influence, Smith creates unsettling music that is both beautiful and eerie. His new CD, *Nikko Wolverine*—which was recorded in his home studio, Tijuana Donkey Show Studios, in Encino, California—features his creations and pedal steel guitar. Smith draws ghostly, shimmering sounds from his unique instruments with a bow or by thumping them with mallets.

"It's all metallic sounds I work with," says Smith, who lent his talents to 1999's *American Beauty* soundtrack. "I don't have the touch for wood. I'm very comfortable with metal. Metallic sounds are more complicated—there's more stuff to work with."

Recording *Nikko Wolverine* took six years, time Smith spent grappling with the challenges of recording instruments that come with tuning systems and tones all their own. Despite his admiration for Partch's music, Smith does not adhere to the avant-garde composer's microtonal tuning system. Smith's works develop largely through experimentation and chance, legacies of his unique creations.

"I'll record all night, let it sit there and percolate, then come back a few weeks later," Smith says. "A lot of the time I'm looking for happy accidents."

Smith deliberately keeps the recording process simple. For tracks that feature his creations, Smith used Audio-Technica AT4050 and Earthworks QTC1 mics to capture the instruments' complex tones. Placed in a room built for a nine-foot Steinway piano, the instruments were either close-miked for rich details or miked with ample space to capture sounds with plenty of air. He opts for the AT4050s when he wants a tone bereft of treatment.

"[The AT4050s] aren't warm and friendly," Smith says. "They're very unforgiving. You hear everything. Because I'm dealing with complex sounds, I want to hear everything. The music is in the details; all of the sounds are basically the same color. If you smear the details with reverb or friendly mics, there's nothing to work with."

When tracking the album, Smith alternated between a Tascam DA-88 digital multitrack recorder linked with a Prism MR-2024 interface and a Fostex D-5 DAT recorder. The tracks made their way into Sonic Solutions' SonicStudio digital audio workstation, and Smith occasionally turned to BIAS *Peak* as well as U&I Software's *MetaSynth* for its granular effects.

Smith took a different approach to recording his pedal steel guitar tracks, using a Mackie 1604 VLZ-Pro 16-channel mixer and an assortment of preamps and outboard effects processors, including Metasonix's TS-21 Hellfire Modulator.

Smith gradually learned how to use the equipment he bought for his home studio but refused to become fully immersed in gear. "It's a tool," says Smith, who holds a master's degree in music composition from CalArts. "It's the age-old thing of motorheads and gearheads—you can get caught up in what it is or what it does. I have a cool car, but it's also for transportation.

"I like to see myself as a composer, but I see myself as a welder some days and a machinist others," Smith says. When not building complex instruments, he plays pedal and nonpedal guitars in a Western swing band and guitar for a heavy metal outfit. He's working on another CD due out in the fall.

"I think the magic is in the music, whether it's the sounds I work with or any of the other music forms," Smith says. "I want to be where the magic is."

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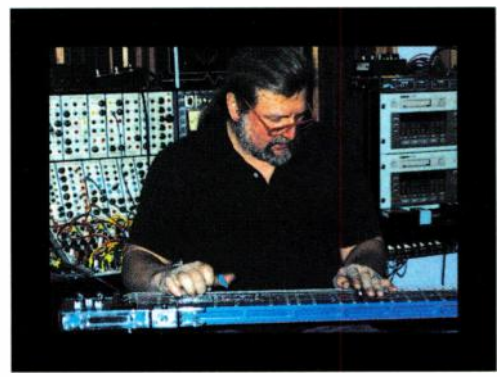

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By Nick Peck

Sound by Design

Make pictures come to life with sound.

If you've been to the movies lately, you've probably seen the name of a sound designer in the credits along with the names of the composer and cinematographer. You're also likely to find a sound designer as part of the production team for every computer and video game you come across, not to mention theatrical productions, Web sites, and even radio dramas. But exactly what is a sound designer, and when did that position reach its current status?

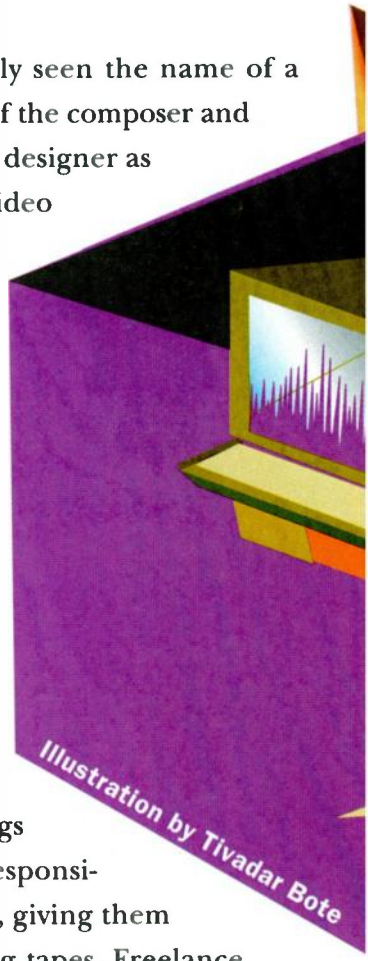


Illustration by Tivadar Bote

Sound design is as old as talking movies, but it really became a serious full-time endeavor when the use of sound was brought to new heights in films such as *Apocalypse Now* and *Star Wars*. Since the mid-1970s, sound design has become an essential part of most major films as well as the majority of computer and video games.

Today, sound-design work is available for virtually every form of visual entertainment. Large game companies often have junior sound-assistant openings that can lead to more senior positions with greater responsibility. Film post-production houses often hire interns, giving them the chance to learn while making coffee and labeling tapes. Freelance studios occasionally have roles for junior and senior sound designers to help with Web and game projects.

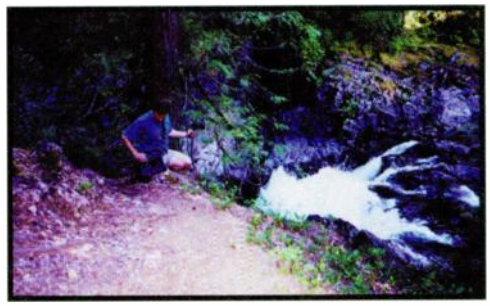
The art of sound design has provided fine careers for many musicians who are not interested in traditional job opportunities in music but love audio production and the creativity of the sonic arts. It's a booming profession that requires a combination of skill, patience, and hard work; professional sound designers are highly sought after and can command good fees.

BY ANY NAME

So what is sound design? How is it used in different media, and what is the process by which sonic tableaux are brought to the big and small screens? In the broadest sense, the purpose of sound design is to augment or enhance the telling of a story. In most cases, that involves the creation, manipulation, and organization of nonmusical sonic elements. Those elements can include door slams, cricket chirps, or computer beeps.

Sound design is the process that turns James Earl Jones's deep voice into Darth Vader's evil growl, and it adds the swishes and smacks that pepper the combat scenes in Hong Kong action flicks. Sound design is the sound of lasers firing and ships exploding in science-fiction games or waves lapping gently against creaky docks in a pirate adventure. What do these disparate sonic examples have in common? They reflect the imagination and taste of the sound designer as he or she tries to enhance a story with sound.

The sound-design process is no real mystery. In fact, you can break most jobs down into seven key steps: determine what sounds are needed, collect the raw sonic materials, manipulate and edit the sounds, integrate them into the project, revise until satisfied or time runs out, mix the sounds, and deliver the finished product to the client. This article will look at each of these steps and define a number of common terms. (See www.filmsound.org for a great collection of articles about sound design, including a glossary.) By the end, you should have a solid understanding of the technical and artistic elements that go into successful sound design.



DAWN LORRAINE

FIG. 2: Field recordings can be a valuable source of authentic material. For a pirate-adventure game, Nick Peck recorded numerous watery locations.

SONIC STRATA

The sonic elements in a project can normally be broken down into several layers that serve different functions. Often, different people work on different layers simultaneously. The sound layers are combined with dialog and music during the mix, which creates the finished presentation. (In the case of interactive media, the mix consists of programming the volume and pan levels of the various elements in code rather than on a mixing console.)

Most projects begin with a *spotting session*, which is attended by the sound designer and film director or game producer. *Spotting* is the process of watching a scene, making a list of the sonic elements that are needed, and dividing them into their constituent layers. I'll define the layers of sound by spotting the following scene.

It is a foggy midnight near the docks. Lapping against the pier, the waves are restless, and a light breeze is kicking up. In the distance, a foghorn blows. The hero stumbles into the frame, his old leather shoes scuffing and scraping the sidewalk as he struggles to keep his balance. He hears tires squealing behind him and whirls around. A half-empty vodka bottle falls out of his coat pocket and explodes like a grenade on the sidewalk. He turns and runs forward, right into a fruit crate that was home to an alley cat, which yowls in protest and runs off into the night.

Foley. *Foley* is the process of recording the sounds of human action in a studio to mimic actors' onscreen movements (see Fig. 1). Footsteps, the rustling of clothing, the handling of objects, and other sounds are recorded by the Foley team while watching the picture. Foley is a deep and dedicated art form, and good Foley artists have a rare combination of skills that include amazing reflexes, great physical control and stamina, a rich imagination, and a knack for coaxing the best sounds out of inanimate objects.

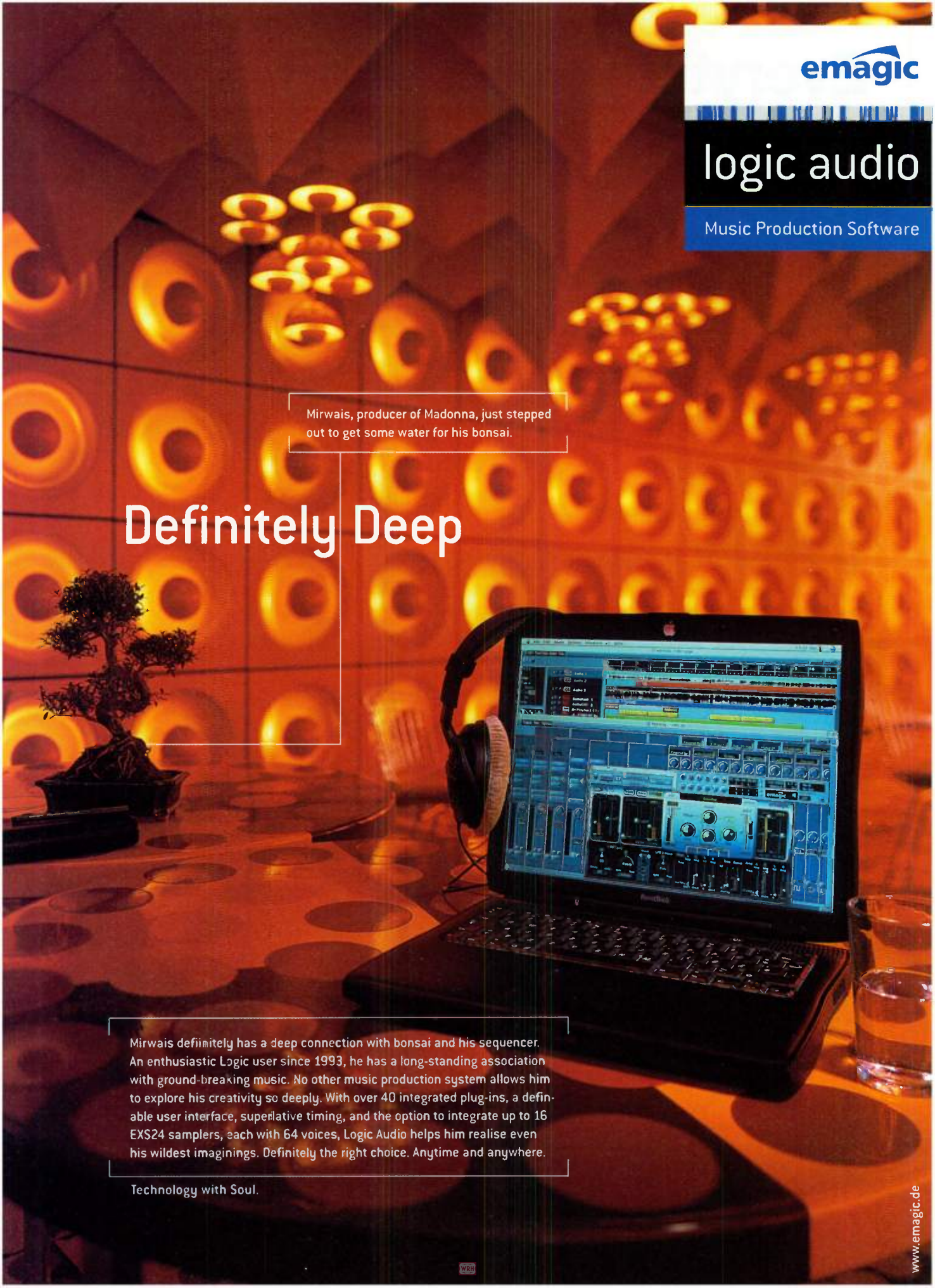
For the scene described above, the Foley artists would find a pair of worn leather shoes to imitate the character's and then record all the scuffles and foot scrapes in a Foley pit, a concrete box with sand in the bottom. They would also record clothing rustles to mimic the character's movements, and the sound of the bottle slipping out of his coat pocket. The final recording would be the clattering of the fruit crate.

Hard SFX. *Hard* or *principal SFX* (sound effects) are the primary up-front sounds that sync to important events on the screen, highlight the drama, and help tell the story. The lead sound designer typically works on those sounds. In the sample scene, the hard effects are the tires squealing, the bottle smashing, the cat yowling, and possibly the clatter as the hero runs into the crate. The final mix might include the Foley artists' crate clatter, the sound designer's clatter, or a combination of both.

In large-budget film projects, the hard SFX category is further subdivided into *editorial* and *principal effects*. Editorial effects are routine, everyday sounds such as doors closing and cars starting. Principal effects are the big, production-specific laser zaps, explosions, and dinosaur footsteps.

NICK PECK

FIG. 1: Foley is the process of adding the sounds of human activities, such as footsteps and the rustling of clothing, to a scene. Here, a Foley artist works in a sand-filled concrete box, called a Foley pit.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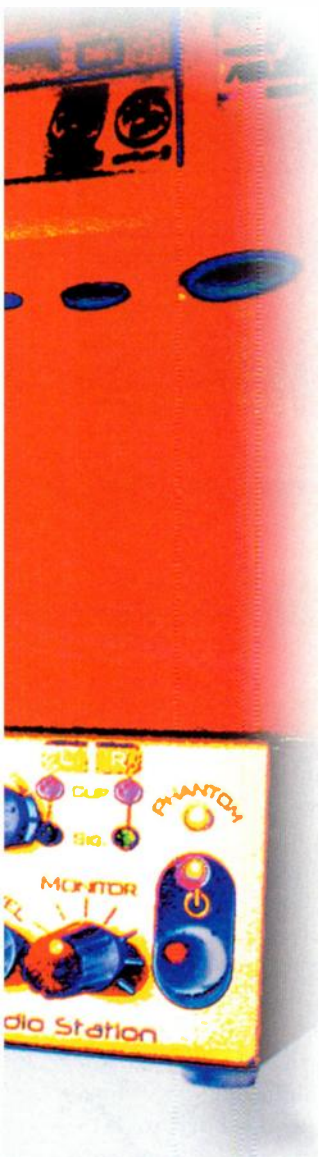

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For example, in the film *Being John Malkovich*, different people created each type of effect. Ren Klyce, Malcolm Fife, and I were contracted to provide all the weird sounds, including the tunnel sequences, everything that takes place inside Malkovich's head, and the bizarre restaurant scene. Another team provided the everyday sounds. In the film *Fight Club*, however, Klyce created a unified aesthetic mood throughout the film by creating all the sounds with his small team. (Klyce was nominated for an Academy Award for that film.)

Production sound. On most live-action projects, the sound at the scene of the shoot, called the *production sound*, is captured by a production recordist. That person's main job is to record the dialog, but he or she might also capture some of the editorial sound effects as well as backgrounds and room tones. That task is made difficult by the sound of cameras whirring, generators buzzing, and directors screaming, but the production recordings are a gold mine of material. Of course, animated projects such as games have no production recordings.

Background/ambience. *Backgrounds* (also called *atmospheres*) are sounds that are not synched to events on screen. Those sounds set the mood and define where something is taking place. Distant city traffic, the ever-present rumble of a starship, or a chorus of birds in the jungle can serve to reinforce the visual image and enhance the story's believability. Backgrounds also come in two types. *Ambiences* are long, continuous recordings that set a mood with something that doesn't call attention to the track. *Stingers* or *specifics* are short elements added to the ambience tracks at certain times to spice things up.

In the scene example, the ambience track is a continuous recording of waves lapping against the piers, with a bit of wood creaking and some distant traffic. That tells the audience that the scene takes place in a city, near the water. The stingers include the distant foghorn and a breath of spooky wind placed at just the right moment.

Room tone is a special type of ambience, typically a recording of the atmosphere of an interior space with no specific sound. It's not very dramatic, but room tone is important for creating a subtle undercurrent that ties together the other elements in a scene. It is critical in matching ADR (automated dialog replacement, which is dialog recorded after the fact) with dialog recorded during the shoot. Room tone is mixed with the replacement dialog, which helps conceal the fact that it was recorded elsewhere.

COURTESY NIHILISTIC SOFTWARE

FIG. 4: Human sounds can be used to give nonhuman characters lifelike qualities. The sounds used for the Szlachta character in the game *Vampire: The Masquerade* started as baby sounds.

COURTESY RYCOTE MICROPHONE WINDSHIELDS LTD.

FIG. 3: Wind socks, such as the Rycote Windjammer, are used in outdoor field recordings to tame wind rumble.

FROM THE TOP

There are various steps in the sound-design process, including ways to find material, process sound, and combine sources. Keep in mind that sound design is a post-production process; that is, it takes place after some or all of the primary production work (filming and animation, for example) is complete.

In determining how to design sound for a project, you need to gather everything related to the project that you can find. That might include scripts, documents, footage dubbed to VHS tape, QuickTime movies of game animations and alpha versions of the game, or rough versions of a Web applet in progress. Whatever content your collaborators can provide is of utmost importance; you need to see what is going on before you can design sound for the project. (See the sidebar "Preliminary Procedures" for additional suggestions about preparing yourself for the task.) Doing it any other way guarantees a mismatch between the sound and visual elements and a mess at the end of the project.

Also, because sound is added at the

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end of the process, all the delays and late deliveries throughout the production, as well as all budget overruns, will happen before you get to ply your trade. It isn't fair, but that's how it goes.

ACQUIRING RAW SOUND

Once you have decided what sounds you will need, the next step is to acquire the raw sound recordings. Those recordings are the clay from which your sonic sculpture will come, and it is essential to pick the right material from the start. Look for sounds that you think are interesting, rich, and full of life; it's difficult to breathe life into dull or listless raw materials. The right well-recorded sound needs virtually no processing to get the message across.

Field recording. *Field recording* is the process of taking a portable tape recorder and microphone into the unruly world outside your studio. Field recording is terrific for capturing ambiences, animals, airplanes, or nearly anything else that you can't bring into your studio. There is no part of sound design I love more than field recording. It's supremely satisfying to strap on a bunch of gear and go out into the world to get a sound you need. The results are authentic, unique, and your own—frozen moments in time and space that you have committed to tape.

When traveling, I always take my field recorder (a portable DAT with a stereo mic preamp and twin mics) so I'm ready to capture whatever interesting sounds I come across. I've recorded airport ambience in Beijing, hippopotamus calls in Kenya, traffic on a rainy night in Amsterdam, and the rhythmic chugging of a train cutting through the Italian countryside. For my work on Lucasarts Entertainment's pirate-adventure game *Escape from Monkey Island*, I recorded as many watery locations as I could (see **Fig. 2**). Field recordings can fill a game with original ambiences that you won't get from a commercial sound-effects library.

Field recording does have one distinctive drawback: the frequent intrusion of unwanted noise, such as wind and human sounds. Putting a microphone in a windy spot doesn't just give the whistling, rustling sound you hear as wind in the movies. When the air blows directly on the diaphragm of the mic, it creates unbelievably loud low-frequency rumbles that ruin your recording. Try blowing directly on a mic and you'll hear what it sounds like.

PRELIMINARY PROCEDURES

Because the sound designer is part of a collaborative effort, you'll be working with other creative people who have their own perspectives on the project as well as technical folks who will be integrating your work. Developing good relationships with these people is crucial, and the key to making that happen is communication.

You must have several skills that are just as important as any artistic abilities you bring to the project. The first is listening to the needs of the other team members. You must really hear the creative desires of the project leader. At the outset, discuss his or her ideas and conceptions, starting with the broad viewpoint—"I want a film noir, 1930s New York City–style approach"—and working down to specifics—"It would be great to hear that taxicab with a 1934 Ford engine sound."

Take as much of their time as you can for the initial meetings, and bat ideas around until you all feel that you're on the same page. Take notes. Do more listening than talking but don't be afraid to chime in with your opinions. Be friendly and warm—that is critical to establishing their trust in you. Most importantly, remember that they are the bosses, so listen to them and satisfy their desires to the best of your ability. If you have an opinion that is different from theirs, don't be afraid to voice it—that is what they are paying you for—but if it is overruled, smile and move on. Whining, struggling, or playing the *artiste* guarantees that you will not work with them again.

Next, be prepared to clearly communicate what you will do, when you will do it, and what you need from others to do your job. The collaborative process fosters personal and artistic growth, as well as great finished material. The tricks to making it work are having a confident, nonarrogant demeanor and projecting the vibe that you really care about the project and respect the people you are working with.

Develop as much technical understanding of the underlying delivery platform as possible. Beyond having the know-how to operate your equipment and create beautiful sound, it is essential to understand the needs of the people to whom you will be handing your work. Every film sound designer needs to know what a "pull-up" is and be able to speak coherently in feet and frames. Every Web sound designer understands the need for incredibly small files and what can and cannot be done with technologies such as Flash and Javascript. Every game sound designer understands the limitations of the particular gaming platform for which they are designing in terms of RAM availability, sampling rates, looping capabilities, file formats, and so forth. And it doesn't hurt to have a basic conceptual understanding of computer programming.

Finally, make friends with the programmers; they are the people who have ultimate control over whether your hard work will sound right in the game or on the Web. Having clear technical conversations up front will save time, rework, and frustration near the end of the project.

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I recommend three approaches to dealing with wind on field recordings. First, try to avoid it by recording on non-windy days or during times that the wind is not kicking up. Second, use wind socks on your microphones. These are large furry coverings, such as the Rycote Windjammer (see **Fig. 3**), that block and distribute the wind air pressure before it hits the mic. Third, be careful with mic placement. If you can place the mic out of direct contact with the wind, perhaps behind a rock or tree, you can record the whoosh of the wind interacting with everything else around it. If you get the occasional buffets of wind across the mic, you can edit them out later and still have a nice, windy recording.

The bigger problem is human noise pollution. Walk out the front door of your home, close your eyes, and listen. Unless you live on a windswept plain in Wyoming, chances are that you can hear traffic, low-flying airplanes, or other human noises. Unfortunately, the microphone is impartial to noise and picks up all the unwanted stuff in addition to what you are looking for.

That problem has two primary solutions. The first is to be where (and when) people aren't. Try to do your exterior field recording after midnight, when there's much less traffic and fewer people around to hassle you (unless you want the effect of an urban background). Bioacoustician and sound pioneer Bernie Krause travels the world trying to record natural habitats without the intrusion of human noise, and he ends up with an extremely low ratio of usable sounds to recorded raw material.

If you are trying to get specific sounds that emanate from one small source, you can fight the problem of excessive noise by using shotgun microphones. Those mics have a hypercardioid polar pattern that rejects any sound not directly in front of the diaphragm. They are also commonly used in production-dialog recording to pick up the actors' voices on the set while minimizing the sound of cameras and other noise. If recording in a busy environment, have patience. Eventually, there will be a lull in the action, and you will get what you want.

COURTESY THE HOLLYWOOD EDGE

FIG. 5: Commercial CDs are a good source of high-quality sound effects. The Hollywood Edge's *The Edge Edition* is one of several libraries that include a wide range of traditional effects.

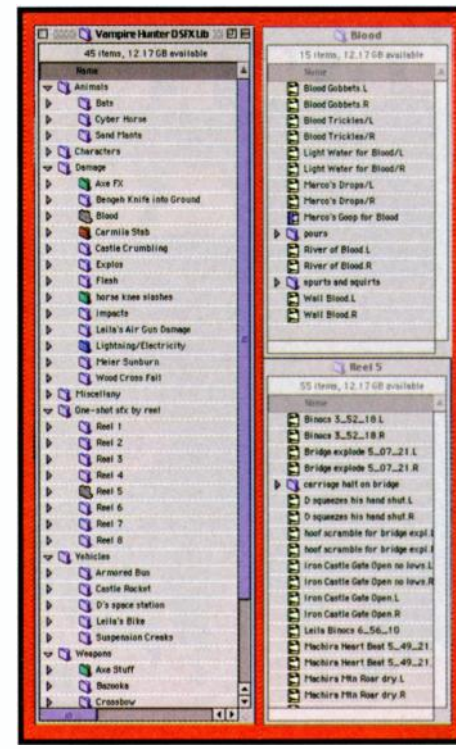

FIG. 6: Organizing your material for a project can help you locate the sounds you need quickly. Sound designers commonly create folders that reflect the type of medium for which they are creating effects.

IN THE STUDIO

Recording in the field is terrific for getting sounds in the context of their environments, but sometimes you want materials that are as clean, clear, noise-free, and devoid of ambience as possible. For those situations, studio recording is the way to go.

I record sound effects in two different areas of my studio. When I want a dry, clean sound with as little coloration as possible, I close-mic the source in my vocal booth with a large-diaphragm mic between six inches and two feet from the sound source. If I want a more open sound, I record in my larger room, with the mic placed one to six feet from the source. I also experiment with shotgun mics, lavaliere mics, and anything else I have available.

If you are recording in a bedroom-size space, deaden the surfaces around the recording area with acoustic absorbers and turn off any unnecessary computer fans. Use the nicest, most neutral mic and best mic preamp you have. If you're recording voice, use a pop filter.

Voice. The most flexible sound generation tool is the human voice. The voice has been used as the basis for a great deal of sound design, and it's capable of mimicking all sorts of animals and birds, among other strange and surprising sounds. The emotive character of the voice expresses movement and interest no matter how much manipulation it undergoes. Voice sounds great slowed down and reversed, and it can impart a sense of the familiar within a context of nonhuman sounds.

One of my favorite uses of voice was in the creation of a creature known as a Szlachta for the game *Vampire: The Masquerade* (see **Fig. 4**). I recorded a baby cooing, grumbling, crying, and whining. That material, slowed down and dropped in pitch, became the disturbing gibbers and moans of a misshapen monster. Our natural empathy toward babies evoked an extra shade of pathos for the hideous creature.

Props. When I'm working on sounds in the studio, I tend to record a wide variety of props. I am always on the

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lookout for items with interesting sounds to add to my collection. Bits of metal, chunks of wood, tools that slide or slip or ratchet, rough pieces of cloth and Velcro, balloons, nails, and sections of chain all can be recorded and manipulated to great effect.

I often think about the physical components that make up an object when I'm imagining how to create its sound. What's it made of? Does it have bits that rattle? Is it squeaky or smooth? Once I have imagined what the sound should be, I look for props I have that would work as an element. Remember: the prop doesn't need to look like what it represents; it just has to sound like it. If I don't have a suitable prop, I head out to find the right thing. Thrift stores are terrific places to find great props at low prices.

GET THEE TO A LIBRARY

Regardless of how much field recording you do and how big your personal library is, it is always useful to have commercial sound-effects libraries on hand. You'll find general and specific sound libraries that are filled with excellent, well-organized material, both basic and exotic. Those libraries are convenient and useful, but always remember their chief limitation: other people own them, too, and the same effects will be used repeatedly in audio projects of all kinds. Try to use commercial effects sparingly, and give the sounds your unique stamp by editing, layering, and manipulating them in every way you can.

There are two major distributors of royalty-free, professional sound-effects libraries on CD: Sound Ideas and Hollywood Edge (see **Fig. 5**). Both companies sell high-quality libraries that cover a broad range of commonly used effects. Both also offer libraries that focus on specific areas such as vehicles, explosions, and footsteps, as well as inexpensive starter sets. In particular, Sound Ideas' *The Library* set and Hollywood Edge's *The Edge Edition* are both good basic libraries.

Good CD libraries can be pricey, averaging $25 to $50 per disc. The reason is simple: the market is quite small, and each disc represents a lot of labor by some of the best sound designers in the world. I try to offset the cost by purchasing one or two new libraries at the beginning of each major project.

One of the problems with purchasing large sound-effects libraries is that you end up paying for many sounds that you will never use. An alternative to buying complete libraries is to utilize Internet-based services that let you audition specific sound effects, then download and pay for just the sounds you need. The two largest online distributors of individual sound effects are Sound Dogs (www.sounddogs.com) and SFX Gallery (www.sfx-gallery.co.uk).

For example, at the Sound Dogs site, you can search for sounds by category (such as "doors") or keyword (such as

"bulldozer") and then go to a page listing all their entries. You can audition low-resolution versions of the sounds and select the ones you want to purchase. The price depends on the length of the sound and format you choose; higher quality means higher prices. When you've made your selection, download the sounds from an FTP site or have a CD-R burned and mailed for a nominal extra fee. Other sites, such as www.ultimatesoundarchive.com, offer a monthly subscription that includes access to whatever sound effects are available on its site.

In addition to online commercial libraries, you can also download repositories of free sound effects and use them legally in your projects. Those free effects are wonderful for starting your collection; however, their resolution and quality vary considerably. Web sites with free sound effects include www.partnersinrhyme.com, www.alcljudprod.se, www.stonewashed.net/sfx.html, and www.wavplace.com. Keep in mind that you get what you pay for.

Synthesizers. Synths aren't generally used for creating real-world sounds, but they can be a great source when you need abstract, otherworldly material. They're well suited for making all sorts of sci-fi, computer, robot, and starship sounds, but avoid clichés. I used a Nord Modular synth to good effect on a recent pair of *Star Wars* games by modeling my sounds on the ARP 2600-based, ring-modulated style that Ben Burtt used for the droids and machines in the films.

Synths are great for creating filtered white-noise sweeps, which make terrific whooshes and swishes. You can use those sounds to create a sense of motion or action, particularly when they're augmented with panning and Doppler processing. Synths are also useful for creating earthquake and rocket rumbles. One of my favorite techniques is to start with white noise, lower the cutoff frequency of a lowpass filter until there is nothing left but the extreme low end, then use a sample-and-hold LFO to modulate the resonance or filter cutoff. I further enhance the low end through EQ or subharmonic synthesis; the result is instant beef.

GET ORGANIZED

After you have recorded the raw sound materials, transfer your sounds to a computer and organize them into clear categories. I work on a Mac with Pro Tools, but I know sound designers who use many varieties of systems and platforms.

Once you develop a good-size library, accessing and organizing your data becomes decidedly important. I have my entire sound library, some 200 GB as of this writing, online at all times. When I start a new project, I create a new folder that becomes the master library for the project. Within that folder, I create subfolders organized by whatever system makes sense for the project (see **Fig. 6**). For a film, I might create subfolders by reels ("reel 1," "reel 2," and so on), whereas for a game, I typically

use general categories that refer to the way the sounds function (for example, "backgrounds," "footsteps," or "objects"). In both cases, I might also organize by specific categories of sound, such as doors, vehicles, and weapons.

Next, I pull my DAT field recordings into Pro Tools, edit the materials, move them into my library folder, and add descriptive names such as "night cricket river amb 01." Any recordings or synth elements I do in the studio are recorded directly into Pro Tools, then moved into the library. I like to keep ambience recordings stereo and about two minutes long. Point-source sounds, such as door slams, are usually mono. I keep all my master material in Sound Designer II format, the Macintosh/Pro Tools industry standard, at 16-bit/44.1 kHz. If possible, always transfer your source material into your computer digitally. I also recommend having an extra hard drive around to make quick backups.

PROCESS THIS!

Once the raw materials are organized in your system, polish them to suit the project. That manipulation phase can be as simple as a bit of editing or EQ or as radical as mangling them into totally new sounds, utterly unrecognizable from the original. What and how much to do depends on the context of the work and the taste of the sound designer.

If the project is a film involving realistic characters in the present time, the majority of manipulation will focus on careful editing, EQ, and dynamics processing. If the project is an ultra-hip techno-cyber sci-fi game, manipulations will involve every weird plug-in and effects processor you can get your hands on. My personal taste leans toward finding a terrifically beautiful sound, recording it really well, and using as little processing as possible. I do rely heavily on pitch-shifting, EQ, and reverb when needed, however.

Now I'll discuss the traditional techniques of sound manipulation, which are appropriate for all sorts of projects, as well as a few tools for producing more radical results. Check out the sidebar "Who Wants to Be a Sound Designer?" for some other projects you can try.

WHO WANTS TO BE A SOUND DESIGNER?

Reading about sound design is fine, but to get into it, try it yourself. I've put together four sound-design missions that you can try at home.

Record an ordinary object. The object can be a glass clinking, a ruler vibrating on the edge of a table, or anything you have lying around. Transfer it to your computer and manipulate it by filtering, reversing, speeding up or slowing down, chopping off the attack, fading it in or out, or splitting it into numerous parts and rearranging them at will.

Make an ambience tape. Grab your portable recorder along with a stereo microphone (or two identical mics in a stereo configuration). Capture the sound of a place and time by recording five minutes from a still position: a busy city street, a restaurant, a cricket-filled twilight, a gentle rainstorm. Breathe quietly and be aware of clothing rustles and hand movements on the microphone or mic cable.

Transfer the recording to your computer, edit the best elements together, and make a two-minute loop. Make sure the loop point doesn't call attention to itself; the loop should be able to run smoothly forever. Remove any events that pop out (for example, a bus's hydraulic door opening, or hissing sounds in your city scene) and save them separately as ambient stingers that can be added in later if desired. If you are recording in nature, notice how often your recording is spoiled by humanity—airplanes flying overhead, distant traffic, and so on.

Collect a lot of something. Record multiple versions of the same thing, such as a door opening and closing, a car starting and stopping, or footsteps. Try recording them in different spaces and at different distances from the mic. Notice how much more of the room the mic seems to pick up than your ears do. Also notice how the high-frequency content of the sound tends to roll off as a function of the distance from the microphone. By adding reverb and judiciously filtering the high end of the sound, you can simulate distance in a close-miked recording. You are now on the road to creating your own sound library.

Make it from other things. Replicate the sound of something familiar by using other objects' sounds and editing and layering them together to create the final product. For example, try to create the sound of a 35 mm autowind camera by recording similar sounds that can be put together to make the target sound. Start by analyzing the components of the camera sound: the click of the button, the rapid open and close of the shutter, the whine of the autowind motor whisking the film along, and a final clunk as the motor stops.

Now create your own version without using a camera. The button press might come from a light switch or a button on a blender. The shutter could be a pair of scissors opening and closing quickly. The autowind could be the servomotor on a car sideview-mirror adjuster or an electric razor. The final thunk might come from snapping the lid onto an aspirin bottle. Load it all into a multitrack digital audio program. Then edit, pitch shift, and slip and slide the elements against each other until you have something that mimics the rhythm and general feeling of an autowind camera. Presto!

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Editing. Editing is simply the process of choosing the part of the sound you like and discarding the rest. I typically record a dozen performances of a given door creak or screw turn and save them one after another in a single file. When examining the file, I listen carefully for the version that best fits the timing and performance of the action. Sometimes the creak of one take fits best with the slam of another take; don't be afraid to mix and match. I've also taken small segments of different sounds and put them together to create entirely new sounds. For example, I once used small snippets of a gun cocking and edited them together with a bicycle gear change to create a metallic tool ratcheting into place. Cutting, pasting, and crossfading are essential tools in your arsenal.

EQ. EQ can be used both correctively and creatively. Rolling off the low end of a wind recording at 85 Hz can eradicate unwanted rumble. But if you move the filter cutoff to 4 kHz, you can eliminate the parts of a sound that provide its basic identity. Try that with the sound of wind and you're left with a wispy, airy sound that could be used for a ghostly ambience. Creating strong resonant peaks in the middle of a sound's frequency spectrum can be great fun as well. Better yet, try moving the peaks over time as the sound plays back.

Dynamics processing. Level compression can be useful in adding punch or body to a sound, but be careful. It is a common mistake to overcompress sound elements before they are layered and mixed because doing so doesn't leave enough headroom once everything is put together. You have to turn down the sound in the mix to make it fit without peaking, negating your original intent. Hang on to your dynamic range—it is a precious commodity in short supply.

Pitch and time shifting. Pitch and time shifting are two methods of sonic manipulation that have been around as long as sound recording. By speeding up or slowing down a tape, you raise or lower the pitch and decrease or increase the duration of the sound concurrently. That technique can change a sound in powerful ways—voices and everyday sounds take on a murky, mysterious quality when dropped in pitch and time, and they assume a cartoonlike quality when raised.

These days, DSP techniques can change the pitch and temporal components of a sound independently. Those functions are useful, but they often impart artifacts to the sounds that disrupt their clarity, beauty, or impact. Try turning off the preserve duration option in the program you use to see if you prefer the results.

Chorusing, flanging, delay. Standard pitch- and delay-based processing have an important place in sound design. Chorusing and very short delays can be useful for converting a mono file to stereo, just as in music. I like using flangers with noise-based sounds to create jets and starship sounds. I've also used flanging on the tail portion of gun and weapon sounds to create more motion and edge.

Delays are great for making sounds bounce around, particularly if the delay taps are panned around the field, with a bit of pitch shifting thrown in for good measure. Short single-tap delays can simulate the early-reflection portion of a reverb, which can evoke the feeling of a tight space without adding a reverb tail.

Reverb. Reverb has two specific uses in sound design: placing a sound in a space and adding depth and size to a sound. If you have a dry sound and want to give it the illusion of being in a particular type of room, you can use fairly short room reverbs with a high wet/dry mix. I've never found this to be as convincing as recording the sound in an appropriate space, but it works well enough. You can also use longer reverb programs, typically plates and halls, to add weight and drama to sounds such as gigantic dinosaur footsteps and cannon shots. For this type of approach, I use the original sound completely dry and add reverb as a separate layer. I then raise the volume of the reverb at the tail portion of the sound.

Worldizing. Digital reverb units are stocked with enough horsepower and brilliant programming to sound terrific, but to my ear, the digital version never sounds quite like the real thing. The most convincing way to make something sound like it was recorded in a room is to record it in one—but sometimes that's not possible. In addition, you may have several sounds recorded under different circumstances that you want to sound as though they belong together.

The solution to both problems is to *worldize* the sounds, that is, to play them back in an appropriate space and record the playback. That means lugging around a high-quality sound-playback system along with your recording rig. Place a speaker in a room or location with the desired aural fingerprint and

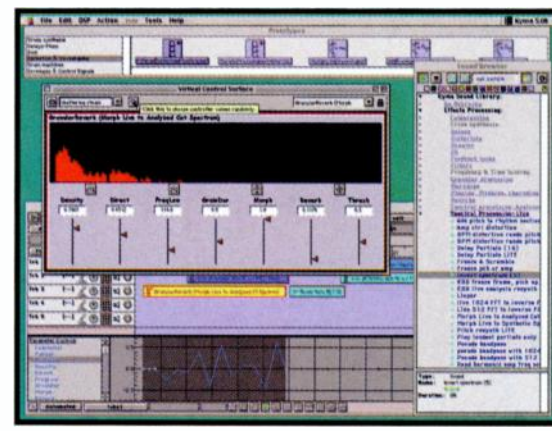

FIG. 7: Symbolic Sound's Kyma System is one of the most powerful sound-design workstations around. It's especially easy to load a "default" source sound into the system and try dozens of different manipulations.

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position a microphone some distance from the speaker. Next, play back your original sounds through the speaker and rerecord them on another tape recorder, capturing the sound with all the reverberant characteristics of the space. That requires much time and effort, but when only the most authentic reproduction will do, worldizing can get you there.

Other options. You can find useful processing functions in special-purpose sound-design tools, such as U & I Software's *MetaSynth* (www.uisoftware.com) and the Kyma System from Symbolic Sound (www.symbolicsound.com; see Fig. 7). Also available is a seemingly endless supply of plug-ins available for transforming audio in unusual ways. But don't worry about owning one of everything; just master the tools you have and try to get the most out of them.

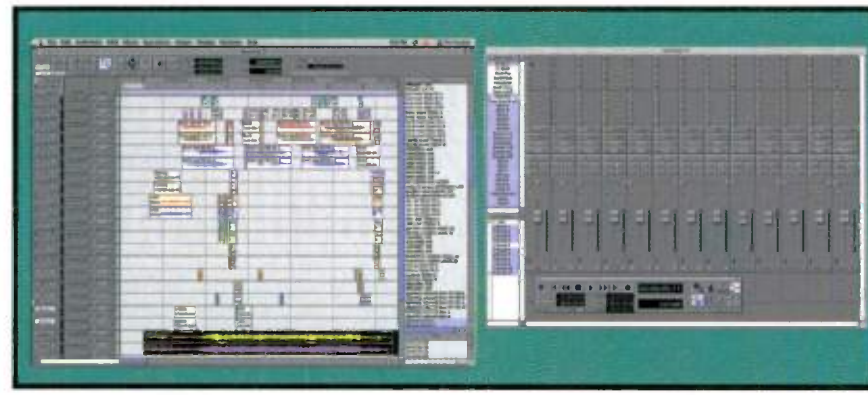

FIG. 8: A typical sound-design session can include numerous individual elements. Here you can see the large number of tracks used to hold the effects in the game *Star Wars Super Bombad Racing*.

PUTTING IT ALL TOGETHER

Once you've collected and processed the sounds you like, import them into a multitrack editor and align them with the visual image. If the visual material has been rendered in a computer, it is delivered to the sound designer in QuickTime format; otherwise, it typically comes on a VHS videotape. In the latter case, I import the picture into the computer as a QuickTime file, which *Pro Tools* plays along with the audio. If possible, it's nice to have a dedicated video card, such as the Aurora Fuse, which takes much of the burden of video playback from the computer CPU.

Sometimes I do the processing and manipulation at this point by letting the visual imagery tell me what the sound needs. I import the sound into the Region bin of *Pro Tools* and drag it into the edit window, roughly where the sound should be. Then I tweak the timing against the visual and adjust frame by frame until it is perfect. Synchronization is as much an art as a mechanical skill—you can't just assume that the closing-door sound should land exactly on the frame in which the door closes. You have to watch each motion repeatedly and adjust the timing of the sound until it feels right.

SOUND AND VISION

When I'm ready to put sound elements to a visual, I create a new *Pro Tools* session and roughly organize the tracks I'll need. Keeping the materials organized by track is critical because a high-density two-minute sound cue can include hundreds of individual elements (see **Fig. 8**). A typical session consists of two stereo pairs of tracks for ambience, a track for each character's dialog, a stereo pair for music, six mono tracks for Foley (two footstep tracks, two clothing tracks, two prop tracks), and four stereo pairs and four additional mono tracks for principal sound effects. At the top of the session I also include one stereo pair and one mono track into which I can load sounds. That lets me quickly pull sounds into the load tracks, adjust their timing, and drag them into an appropriate track.

Once each sound is loaded, it must be evaluated. That may seem obvious, but you must really listen to each sound element. Listen to how it sounds by itself against the picture and how it sounds with the other sonic elements. Listen deeply to the detail of the sound; then put your "big ears" on and listen objectively to the element as part of the whole. Does the general character of the sound work with the visual? What does it need? Will editing or processing improve it? Does it fill out the visual, or could it benefit from adding another layer?

Conversely, does the sound have too many details? Is the sound so busy that it distracts from the intended object of the audience's focus? Those decisions are made almost instantly and lead you to the options of keeping the sound as is, modifying it, or starting again. The real trick here is to balance the endless options and your desire for perfection with the need to make progress.

Now that you have the process down, repeat it dozens, hundreds, or thousands of times until all your sound elements are lined up and sounding good. You are prepared for the mix!

SPECIAL DELIVERY

Once you've completed your work, you must deliver it to the next person in the chain of project personnel. Formats and delivery standards must be carefully specified and spelled out in advance to minimize confusion and rework. Ask the project leader exactly what format and medium the files need

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to be delivered in—the all-nighter you prevent may be your own. (See the sidebar "Plan of Attack" for a discussion of additional steps to consider for delivering different types of media.)

Film and video delivery is fairly standardized. The audio standard for broadcast video is 16-bit/48 kHz, but films can be either 16/44.1 or 16/48, depending on the project. If you are responsible for a finished stereo mix of all audio elements, including dialog and music, you can deliver an AIFF interleaved stereo or Sound Designer II split stereo file burned to CD-R or recorded on DAT. If you mix to DAT, add a *2-pop* to the beginning of the file. That is a short beep that occurs two seconds before the audio content begins, and it's used by the editor to align the audio with picture.

Tascam DA-88 tape is the broadcast industry standard. I like it because the time code is stable, and eight tracks provide enough room for a full 5.1 surround mix on the first six tracks and a stereo mix on the last two tracks. Currently, though, it seems that Digidesign's Pro Tools format dominates the film industry; you can typically deliver a hard drive containing your *Pro Tools* sessions to the mixer. When I do this, I deliver a carefully prepared session that has all the audio elements organized and roughly balanced the way I envision them sounding in the final mix. The idea is that the sound mixer, who is not familiar with the material, should be able to bring all faders up to unity gain and have it sound reasonably good. From that point he or she can tweak levels,

PLAN OF ATTACK

In planning the sound-design process, the nature of the delivery medium must be taken into account. Creative, appropriate sound is important in all media, but each medium has overriding principles that define your approach.

Film and video. Film and video are linear media. The audio is always synchronized to the video and plays back the same way every time. The sound quality is high, and the processes used to marry the audio to the picture are time tested and well understood. Technological advances and headaches occur in various parts of the process, but in general, certain standards allow for fairly smooth production. As a result, the primary sound focus for these media is to create the highest quality, most detailed work possible within the time and budget.

Games. Games are about interactivity. Some sections of games are expository, linear presentations that tend to follow a film approach. But the main areas of the game unfold based on user input, changing direction from moment to moment. As a result, the elements are developed in small pieces that are carefully named and organized, and put back together in real time like an ever-morphing jigsaw puzzle. The programmer plays a huge part in carrying out the vision of the sound designer by making sure that the right sounds trigger at the correct times. When programmers have enough time and sensitivity to audio issues, niceties such as pan and volume changes are included, enhancing the experience.

Game technology changes from game to game and platform to platform. Detailed knowledge of the platform's capabilities and programming system is critical for the sound designer's planning, but there is one guarantee: you will not have the resources you want. Graphics always take priority over sound, both in terms of RAM budgets and labor resources. As a result, every sound must be as short as possible while still conveying excitement and interest.

Sounds are also made smaller by decreasing the sampling rate and applying data compression. Lowered sampling rates mean decreased high-frequency response, so qualities such as airiness or sheen in metallic sounds and cymbals disappear. Reverb tails at the end of sounds always get cut, though many gaming platforms now possess built-in DSP effects that can bring some processing back in real time. Fortunately, hardware improvements and greater understanding that sound is a critical part of an immersive, satisfying game experience mean that the situation becomes less daunting each year.

The Web. If motion picture technology is a mature adult and game technology resembles a prepubescent teenager, then Web-audio technology hasn't even begun to crawl yet. MP3 is great for delivering music to download, but interactive presentations on the Web are a different matter entirely. Slow connections and unstable delivery methods mean that most or all of the content for a typical Flash, Shockwave, or Javascript interactive presentation is downloaded before the user starts working with it. Of course, people are impatient and will click away if a presentation takes too long to download, so the size of the files becomes critical. The overall number of sound files is generally very low, and each file must be very short and typically highly compressed.

Interactive audio design for the Web is making the most of a few short, sparse audio events rather than creating an immersive audio experience. That will change over time, of course, with streaming technologies such as QuickTime, Streaming MP3, and Windows Media Format. When enough consumers have a steady supply of continuous, reliable data streams, opportunities for creating rich, deep sound experiences on the Web will abound.

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pans, and mutes without wasting time.

Game elements are very small and obviously used on computers, so I always deliver them on CD-R. With the advent of broadband connections, I occasionally transfer files over the Internet directly to the people who need them. Usually, though, it's less hassle to simply burn a disc and meet in person to go over the material. In any event, delivering the material in the format and sample rate required by the programmer is of paramount importance. I have found 16-bit/22 kHz,

AIFF format to be a safe bet for PC-based games, but the situation depends on the platform and RAM resources for sound. For example, the Sony PlayStation 2 console runs at 16/48. At the project's end, I deliver all the materials in high-resolution format; if the game gets ported to another platform, the audio elements can be converted from the masters.

Audio for the Web is still in a state of flux, with many different competing formats. RealAudio, MP3, QuickTime, Windows Media, and LiquidAudio are the dominant formats for Web playback, but by the time you read this, there could be more. If you are delivering audio to a developer, you might not need to worry about this. For example, if the audio is part of a Flash or Shockwave presentation, you can deliver stereo WAV files at full resolution, and the application will compress the data for you. However, listen to the results before the audio gets posted.

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* Translation: Goodbye

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Alternatively, you can get the appropriate software tool to convert your high-res files into a compressed format. Don't be afraid to tweak settings and compression rates until you get the smallest file that still sounds good.

THE MIX

When all the hard work of the sound designers, dialog editors, and composers is done, one task remains. This is the moment that strikes fear into the hearts of all involved, a process that can turn perfectly nice, sane audio professionals into bloodthirsty maniacs. This is The Mix. Actually, the mix can be the greatest source of satisfaction on a project. In the mix, you can hear, under the best listening circumstances, all the elements—music and sound, foreground and background—coalesce into a beautiful partner to the picture.

The mix can also be a session fraught with frustration, difficulty, and disappointment. Problems can crop up in a mix when there is too much to do within the time allotted, when sound elements are inadequate, and when people disagree about the decisions being made. Nearly all this can be chalked up to miscommunication or lack of preparation. How can you make sure that you have done everything possible to contribute to a smooth mixing process?

You never know ahead of time exactly how a mix will sound. But you can rely on experience and guesswork to imagine which elements will be where and use this information to create your tracks accordingly. Scanning the scene for likely dialog and music moments helps you to leave more space in your work for a cooperative mix.

For example, say you're working on a close-up of two soldiers conversing on the front lines while grenades and bombs are going off in the background. Without hearing the dialog, you know it will be the most important sound in the scene, and mixing explosions behind the conversation will greatly impair the intelligibility of the words. That type of foresight lets you select explosion sounds that fit the context; you want low-end sounds without frequencies in the range of speech that are somewhat dull and that are generally low-key.

Marco D'Ambrosio, a film composer in the San Francisco Bay Area, makes sure never to reach a high point in his music during an explosion. Instead, he leaves room for the sound designer to put in a big boom, and he puts the musical climax afterward. Planning and imagination make the difference between sound components that complement, rather than compete with, one another.

KEEP IT LOGICAL

Make sure your individual elements are logically grouped on as few tracks as necessary to keep the mix from becoming a

massive confusing mess. On the other hand, use as many tracks as necessary to give the mix engineer the flexibility he or she needs to remove or change individual elements. To some degree, that depends on the engineer's style, the amount of time for the mix, and the mixing system's capabilities. I've delivered as little as a stereo pair of hard sound effects and a stereo pair of ambience, and as many as 32 tracks of hard sound effects.

If you handle dialog and music as well as sound effects, never put them on the same tracks—the panning and EQ tend to be totally different for these elements, and combining them creates a conceptual headache. Ask the mix engineer what he or she wants and make it happen. Make sure your sound elements are clean, are well edited, and sync to picture nicely. Time is a precious commodity during a mix, and fixing problems comes at the expense of making the best mix possible.

Always bear in mind that dialog comes first. If a sound effect gets buried or a clarinet part can't be heard, it is unfortunate but not drastic. However, if a spoken line is unintelligible, the audience is robbed of the story. People are tuned to listen to human speech, and they get annoyed when they can't understand the words. Thus, music and effects are always subservient to dialog in a mix.

Dialog is so critical that the standard mixing convention in film is to use the center channel primarily for dialog. Separated from the music and effects that fill the rest of the space, the words are always heard in the middle of the screen. Dialog is sometimes compressed and limited to help it ride above the rest of the mix. You can also use EQ to make sure that too much information in other elements doesn't compete with the same frequency range as the words. If a line is still not reading well in the mix, turn everything else down a bit.

How do you make music and effects coexist peacefully? As mentioned earlier, one answer is to ensure that important events don't happen simultaneously. Another approach is to have the textures augment each other through contrast. For example, music that is tight and nicely percussive matches languid, airy, continuous ambiences. Similarly, ambient, Brian Eno–esque soundtracks contrast well with a tighter, sparser, more event-driven approach to sound effects.

Music and effects can always inhabit different parts of the frequency spectrum. There will be times when the music feels buried or the effects' subtlety is lost. That is all part of the deal—it is most important to tell the story effectively and artfully. If that means whole swaths of sound or music get lowered or removed in the mix, so be it. The creative process of shaping and changing the sound culminates in the mix itself.

Limiters are useful for increasing the overall perceived level of an element

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or mix. I try to avoid using limiters when creating premixes or *stems* (partial mixes); I prefer to preserve as much dynamic range as possible. In the final mix, though, limiters are often used to shoehorn everything in. Kent Sparling, a rerecording mixer at Skywalker Sound, believes that limiting in the mix is a necessary evil. "It's a shame to have to make everything louder to compensate for a problem in the storytelling," he says.

MEDIA MATRIX

In terms of overall approach, the mix should always be considered from the listener's perspective. There is no point in creating a gigantic, wide-ranging mix for a game if it doesn't sound good on computer speakers. Game sound designers typically mix using low-end computer speakers and switch back to high-quality speakers only for reference. That ensures the sound designers have the same sonic experience as the end-users. Here are some other issues to consider when creating sound for different types of media.

Film. No doubt about it, mixing for film is a beautiful thing. There are no limitations on frequency response due to low sampling rates, and having a subwoofer ensures that dramatic rumbles and explosions will be felt as well as heard. In mixing for 5.1 surround, the center channel enhances the clarity of the dialog, which leaves more room in the left and right for music and effects. Finally, the surround channels can be used liberally to enhance the mood and apparent environment size by sending ambient elements, reverb washes, and occasional musical elements to the rear (not to mention flyover effects). The mix's dynamic range can be quite wide, keeping intimate scenes quiet and saving the volume for big, dramatic moments.

Game. Audio for games has improved substantially during the past few years. With increased budgets, enhanced audio hardware, and more recognition of the need for high-quality audio as a significant part of the game experience, things are getting better all the time. Yet problems remain. The necessities of data compression lead to compromises in both frequency response and dynamic range. A typical sampling rate of 22 kHz means no frequencies above 11 kHz, resulting in a lack of sheen, air, and crispness. The audio is also compressed, using some perceptual coding and variable bit-depth scheme. That step adds artifacts and reduces clarity, and it's particularly harsh on quiet sounds.

What's more, computers have loud fans, which increase the ambient noise floor and thus decrease the listener's dynamic range at the computer. Finally, games tend to be loud and punchy. All those factors lead to mixes and individual sound files that make heavy use of normalization and limiting.

Web. The same computer-based limitations for games are inherent in Web-based presentations. An additional factor is download speed, which affects durations and increases data compression beyond that required for games. One general suggestion: turn technical limitations into design considerations by using very short sound files and loops.

END GAME

As you can see, the audio arts outside of music have incredible depth and richness of expression. When you open your ears to the sounds around you, the world becomes an unending symphony of footsteps on crunchy gravel, wind slithering through tall grasses, and cars flying by in the night. Good sound design is based on events in the real world, and the more attuned you are to sounds in your environment, the better you will be at providing sound for the virtual world.

Your job is always to enhance the visual elements in the project. Be tasteful yet subtle, and your work will best serve its intended purpose.

Nick Peck *owns Perceptive Sound Design, a post-production firm. He plays keyboards in the Bay Area jam band Ten Ton Chicken. You can reach him at nick@tyedye.com or on the Web at www.perceptivesound.com.*

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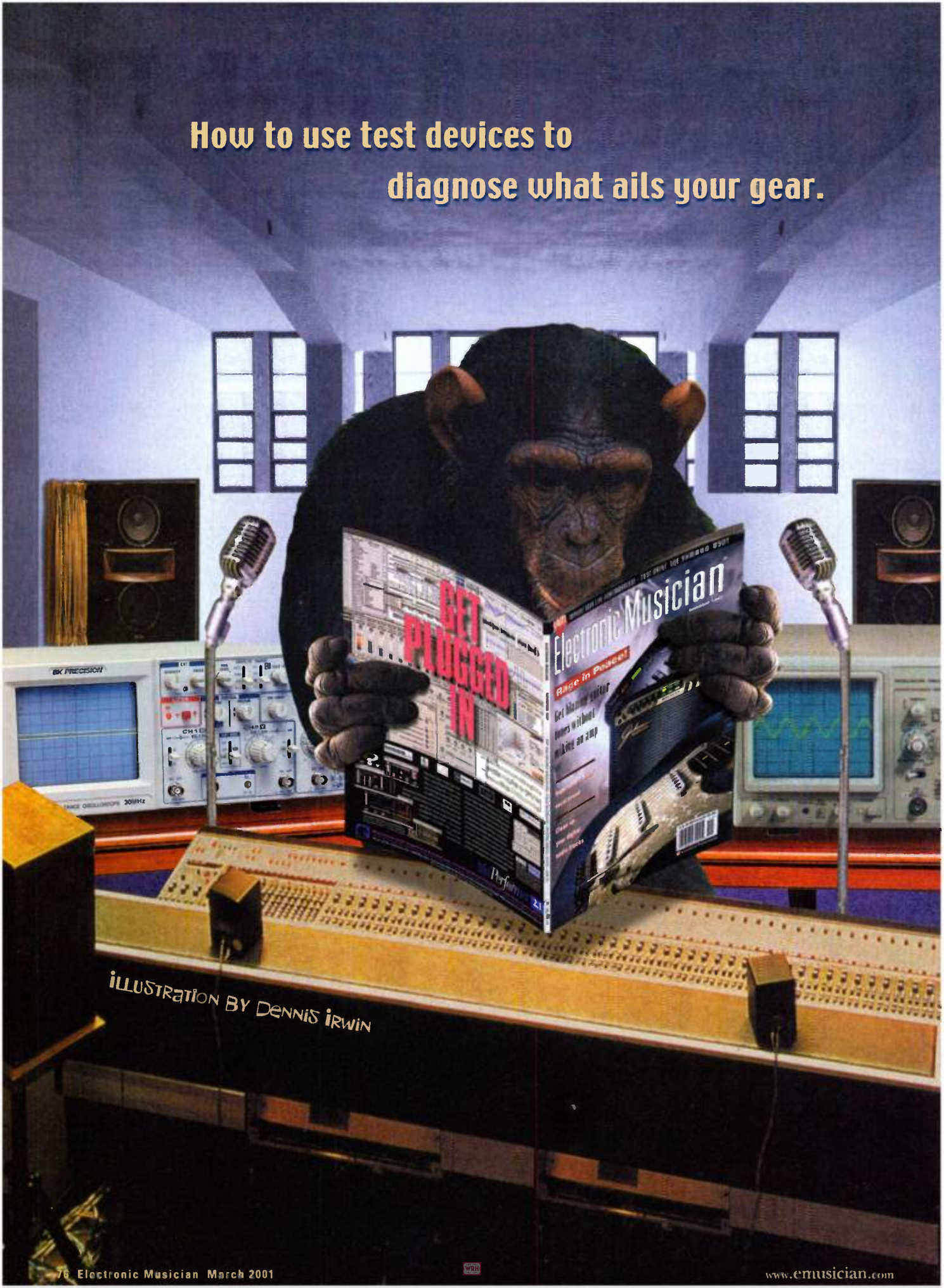

ILLUSTRATION BY DENNIS IRWIN

TESTING, TESTING, 1,2,3

Using audio test equipment is a little like playing an electric guitar: it can take a lifetime of practice to be a true virtuoso, but almost anybody can learn to strum a few chords (think "Louie, Louie"). I don't promise that the CIA will call you to help debug its latest covert monitoring equipment, but this article should enable you to perform some basic test functions with a volt/ohm meter (VOM), a signal generator, and an oscilloscope. In addition, if you've been playing with audio circuits for a while and you know your way around a schematic somewhat, this article will give you a few cool tricks that will save you time and aggravation the next time things go wrong.

By Mike Sokol

However, there is one big caveat. High-voltage electricity can easily be lethal. Even as a trained professional, I nearly electrocuted myself while testing the 600-volt power supply in an Ampeg SVT bass amplifier. If you're unsure about any safety precautions while testing electrical circuits, find someone who's in the business, and let him or her show you how it all works. Here's a list of the things I watch out for.

The majority of electrocutions involve a relatively weak current stopping the heart. Because the heart is regulated by a small chemical/electrical oscillator, it can easily be locked up by a current passing from one hand to the other through your chest. So if you take a shock between your hands, the probability is high that your heart will stop in the process.

Never put two hands in a live circuit. If you have to reach into a chassis with a voltage probe, don't rest your other hand on the tabletop or chassis. If you do and your probe hand slips and makes contact with anything electrically energized, your heart will take the jolt. An old electrician's trick is to put your free hand in your back pocket so you're not tempted to lean it on anything conductive.

Discharge all capacitors before poking around. Most electrical circuits include capacitors that act like reservoirs to store up electric energy. Many of these capacitors retain high voltages for minutes or even hours after an amplifier is powered down. Touching a charged capacitor probably won't kill you, but it can knock you across the room. (I know from personal experience.)

The best precaution is to make a *shorting stick*. You only need a meter probe, which you can get at Radio Shack (www.radioshack.com). This is nothing more than a hollow plastic tube with a metal spike at one end and a wire with a spring clip at the other end. Open the tube and solder a 1 kΩ, ½W resistor between the spike and wire (see **Fig. 1**); if there is a wire running the length of the probe, you'll need to cut it and insert the resistor. Before poking around in an amplifier, attach the clip to the chassis and touch the spike to all capacitors for a few seconds; this will drain any charges and prevent you from being shocked.

Don't trust anyone. That is not paranoia. If you're working on a circuit that's connected to a disengaged circuit breaker in the wall, affix a piece of PVC tape over the breaker with your name and a "Do not turn on" note on it so that no one accidentally flips it on while you're in the circuit. If you're working in another room from the circuit panel, such as a bar or auditorium, add your cell phone or pager number to the note. That way, someone looking for the lighting power can call before flipping the breaker—which could prevent a potentially lethal surprise. As a matter of caution, treat all circuits as live. Remember not to grab anything with both hands.

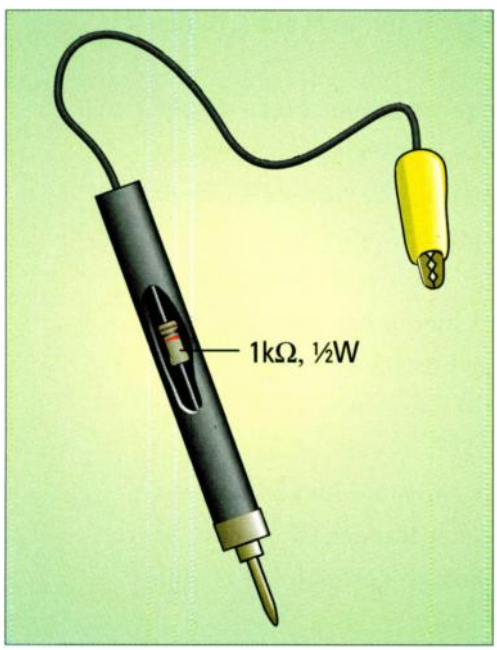

FIG. 1: Make a shorting stick by adding a 1 kΩ resistor to a test probe.

CHEAP TRICKS

The most elementary equipment tests usually involve a VOM, which is a simple piece of test equipment that measures AC and DC voltage, in addition to DC resistance in ohms. To use the VOM, set it to the desired type of measurement, take the two probes that connect to the VOM with flexible wires, touch them to the appropriate points in the circuit, and read the value on the front-panel meter.

The cheapest VOMs from Radio Shack are okay—they will get you by in a pinch—but I prefer a digital meter, from companies such as Triplett (www.triplett.com), that includes an analog-style LCD bar resembling a VU meter (see **Fig. 2**).

FIG. 2: A volt/ohm meter (VOM) is an essential piece of test equipment. It measures resistance and voltage between any two points in a circuit.

That lets you see quick changes in the signal you might miss with a numerical display.

Be sure that the meter includes an audible beep to indicate that both probes are connected to the same conductor. This is called a *continuity test*. For example, to determine if a conductor in a cable is broken from one end to the other, set the VOM to measure ohms and touch the probes to corresponding points on each connector (say, the tips of the ¼-inch connectors or pin 1 of the XLRs). If the conductor attached to those points is not broken, the VOM will emit a beep. Older VOMs buzzed when the continuity was good, so this technique is often called "buzzing out" the cable.

I've spent many hours buzzing out custom cables with a VOM. However, the easiest and best way to test cables is with Ebtech's Swizz Army Tester (www.cymation.com; see **Fig. 3**). It not only evaluates pin-to-pin connections but also has a neat sample-and-hold

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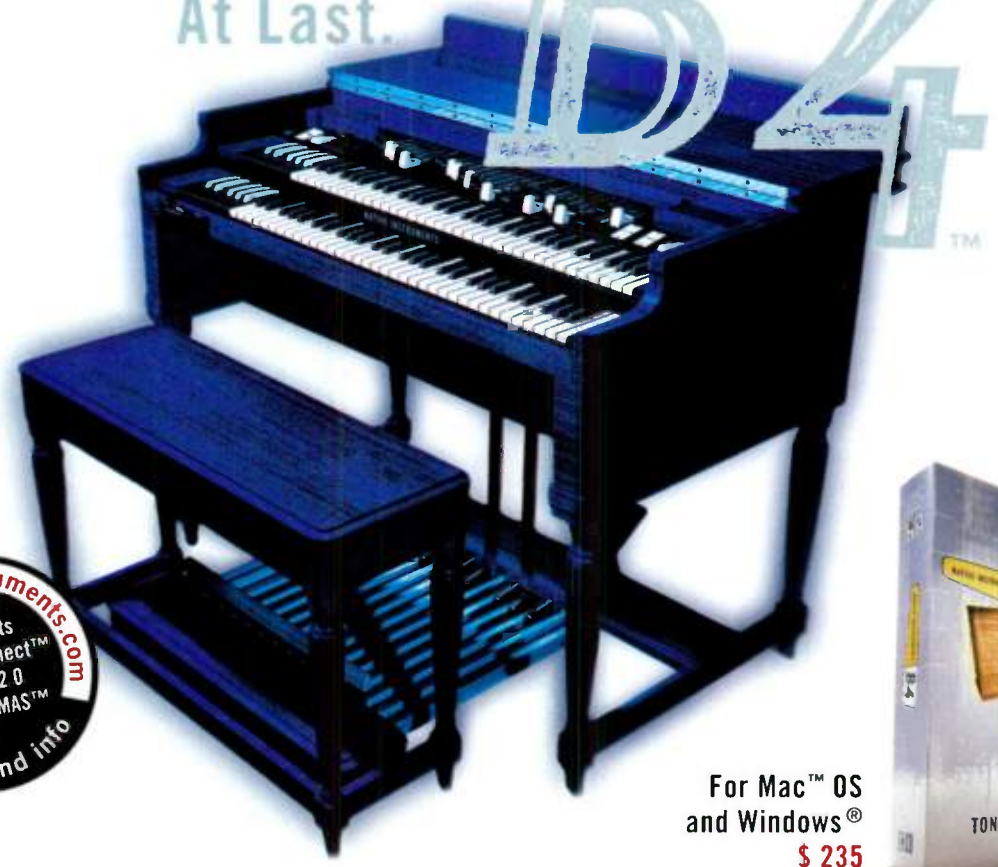

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circuit that reveals intermittent shorts and breaks in the cable. After plugging in the suspect cable and pressing the Reset button, you can flex, pull, and otherwise manipulate the cable to force a failure. Then you only have to look at the LED to see if the cable has any shorts or dropouts.

While you're at it, make sure your XLR cables are pinned out properly—that is, pin 1 to pin 1, pin 2 to pin 2, and pin 3 to pin 3. If you need to use pin-2-to-pin-3 swappers, make sure they are plainly marked. Many audio problems are caused by inverted polarity in a cable, which introduces all sorts of phase-cancellation problems.

All of your TRS-to-XLR adapters should be pinned nominally (the tip of the ¼-inch TRS should be connected to pin 2 on the XLR). I've been on jobs where either pin 2 or pin 3 was hot in a bunch of adapters, and it was a nightmare to get anything to work.

Double-check that the XLR cable's pin 1 (which is connected to the cable shield) isn't connected to the metal shell of the connectors. If they are connected and the metal shell contacts a chassis or a wet stage, serious ground loops can result. In addition, many professional video cameras keep the grounds of their internal power supplies isolated. If you connect them to an XLR cable with pin 1 tied to the shell, you will create a significant audio hum inside the camera.

Make sure that there is infinite resistance between each XLR cable's leads. Set the VOM to its highest resistance setting and test between pins 1 and 2, 2 and 3, and 3 and 1. Any conductivity may indicate that the cable insulation has broken down due to moisture or a physical pinch in the wire. That can cause a crackling noise when 48-volt phantom power is applied to run a condenser microphone. Such leaky cables will probably work fine with dynamic microphones or line-level runs—but Murphy's Law will make sure that bad mic lead will end up connected where it sounds the worst.

FIG. 3: The Swizz Army Tester is an excellent—and simple—way to test cables.

FIG. 4: The Galaxy Cricket tests speaker polarity. It includes a clicker and a listener.

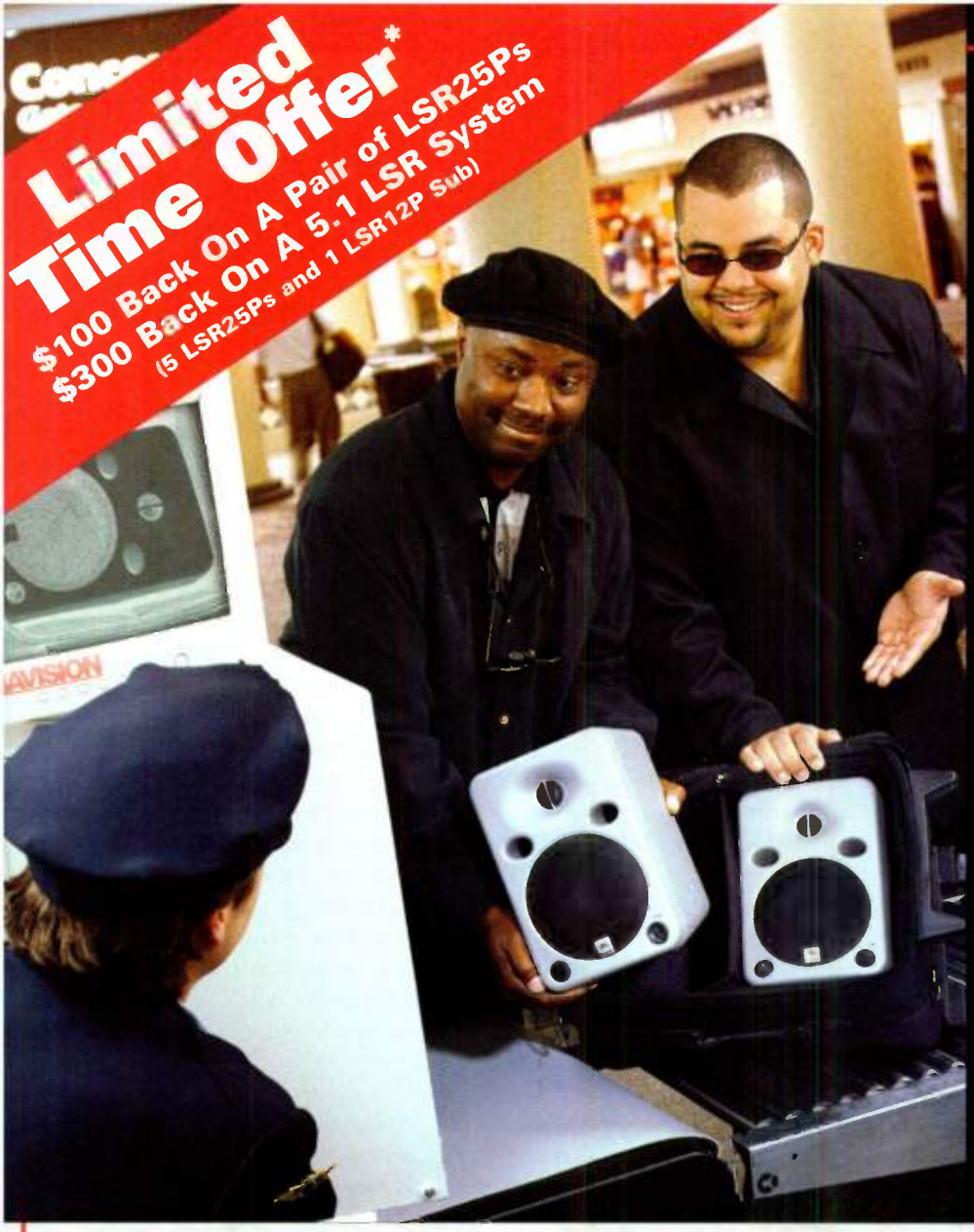
POLARIZED SPEAKERS

Incorrect speaker polarity can be problematic. For instance, if the left and right speakers are wired out of polarity with each other, you will hear a big bass cancellation in the center position between them. The obvious solution is to flip the polarity of one of the cables going from the amp to either speaker.

If you're on a budget, you can temporarily connect a 9-volt battery to the speaker terminals. When the battery's positive terminal is placed on the hot connector of the cabinet (usually labeled with a "+"), the woofers should all move in the same direction, usually out. If one woofer goes the wrong way, you need to reverse the cable leads to that speaker.

The only way to test polarity in a horn-type speaker is with a speaker-polarity tester or *clicker* (see **Fig. 4**). Galaxy Audio (www.galaxyaudio.com) makes one called the Cricket, but other manufacturers offer clickers as well. The tester has two parts: a "clicker," which sends a pulse into your sound system, and a "listener," which has a microphone and an LED that blinks red or green, depending on the polarity. When the listener box is placed directly in front of a speaker, you can quickly figure out whether everything is wired with the correct polarity.

Don't assume that you don't need to test the polarity of factory-wired monitors. Some manufacturers, such as JBL, wire their cabinets in reverse from the rest of the audio universe. I've also seen



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Sammy Peralta loves music. That pure and simple fact comes through strikingly clear as he sits at his keyboard tinkering with half-written tunes. Sammy's background includes work with talents including Tito Puente and Willie Bermudez. "I have to be careful because I can get so lost in the music, I sometimes forget I have a family that would like a little of my attention too".

One last point: Sammy Peralta's new CD ***On the One*** featuring Lenny White was mixed entirely with LSR monitors.

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factory-fresh speaker cabinets in which one of the drivers was wired with reversed polarity. The only way to know is to test.

THE HUNT IS ON

One particularly cool set of test gear is called a *fox and hound* (see **Fig. 5**). This set is great for figuring out wiring harnesses and long cable runs. Like a clicker, a fox and hound comes in two parts: a signal injector (the fox), which puts a chirping tone on the wire, and a receiver (the hound), which detects the signal by induction and plays it on a small speaker. Place the hound next to a bundle of wires; the signal will get louder as you move it closer to the wire to which the fox is connected.

The fox-and-hound test is much quicker than buzzing out a bunch of unmarked cables, and it even detects the signal through a shield. The test set costs between $50 and $75 and is available anywhere testers are sold to phone-wire installers. A variety of companies manufacture fox-and-hound sets; mine is made by Triplett.

To test amplifier outputs, I use a headphone pad, a little gadget that you can make (see **Fig. 6**). It's nothing more than an attenuator that lets you connect headphones directly to the output of a power amp; the variable resistor allows you to adjust it for amps with different power levels. I usually put a dual banana plug on the input, which makes it easy to connect to the amplifier's speaker outputs directly. Set the variable resistor for maximum attenuation before you plug it in; then, bring the level up slowly so you don't fry the headphones—or your ears.

This gadget is handy because you can quickly determine if your crossovers are passing the correct signal to the correct amplifier, if the amp is distorting or otherwise misbehaving; or if you've blown up some speakers but the amp is still working. Likewise, you can use it while the performance is in progress, if you need to. However, be aware that you cannot just plug a set of headphones into a power amplifier without a resistive pad. If you try that, your head will probably explode a few milliseconds before the headphones melt down.

FIG. 5: The Triplett fox and hound helps trace cables in a bundle. The fox injects a signal, which is picked up by the hound.

SCOPING OUT WAVEFORMS

Signal generators are important pieces of test equipment, and they come in a variety of sizes and types (see **Fig. 7**). Typically, they have a rotary control on the front to adjust the frequency; a range control to select frequency ranges (such as 1 to 100 Hz or 100 to 1,000 Hz); and a waveform selector for sine, square, triangle, and pulse waveforms.

If you don't have access to a signal generator, you can use a test CD with those waveforms or generate them from an application such as *Sound Forge*. I don't recommend using a synthesizer to generate the waveforms because synths usually have filters and other processors in the signal path that make the waveforms sound more interesting but harmonically less pure. I have posted a number of sine waves for tuning and tape-deck alignment along with other test waveforms on my Web site (www.modernrecording.com/resources/tones.shtml).

Each waveform type is used to test different conditions. Sounding like a pure flute tone, the sine wave is very musical. Because of this waveform's purity, you can easily hear harmonic overtones added by a piece of audio gear. For instance, if you feed a sine wave into a small amplifier that drives a speaker cabinet and then sweep the frequency in the 30 to 300 Hz range, you can listen for cabinet buzzes, handle rattles, and voice-coil rubbing (a light buzz that's harmonically related to the input frequency). However, be careful not to output high-level sine waves at horn frequencies (above 1 kHz or so), because many horn drivers and tweeters will quickly burn out with a few tens of watts of continuous tone.

Square waves combine low-frequency information (the flat part of the waveform) and high-frequency information (the rising vertical part of the waveform), so they can be used to evaluate circuit stability at both ends of the audio spectrum. Square waves are most useful when teamed with an *oscilloscope*, a device that displays AC-signal waveforms (among other things). Hardware models tend to be expensive, but virtual oscilloscopes

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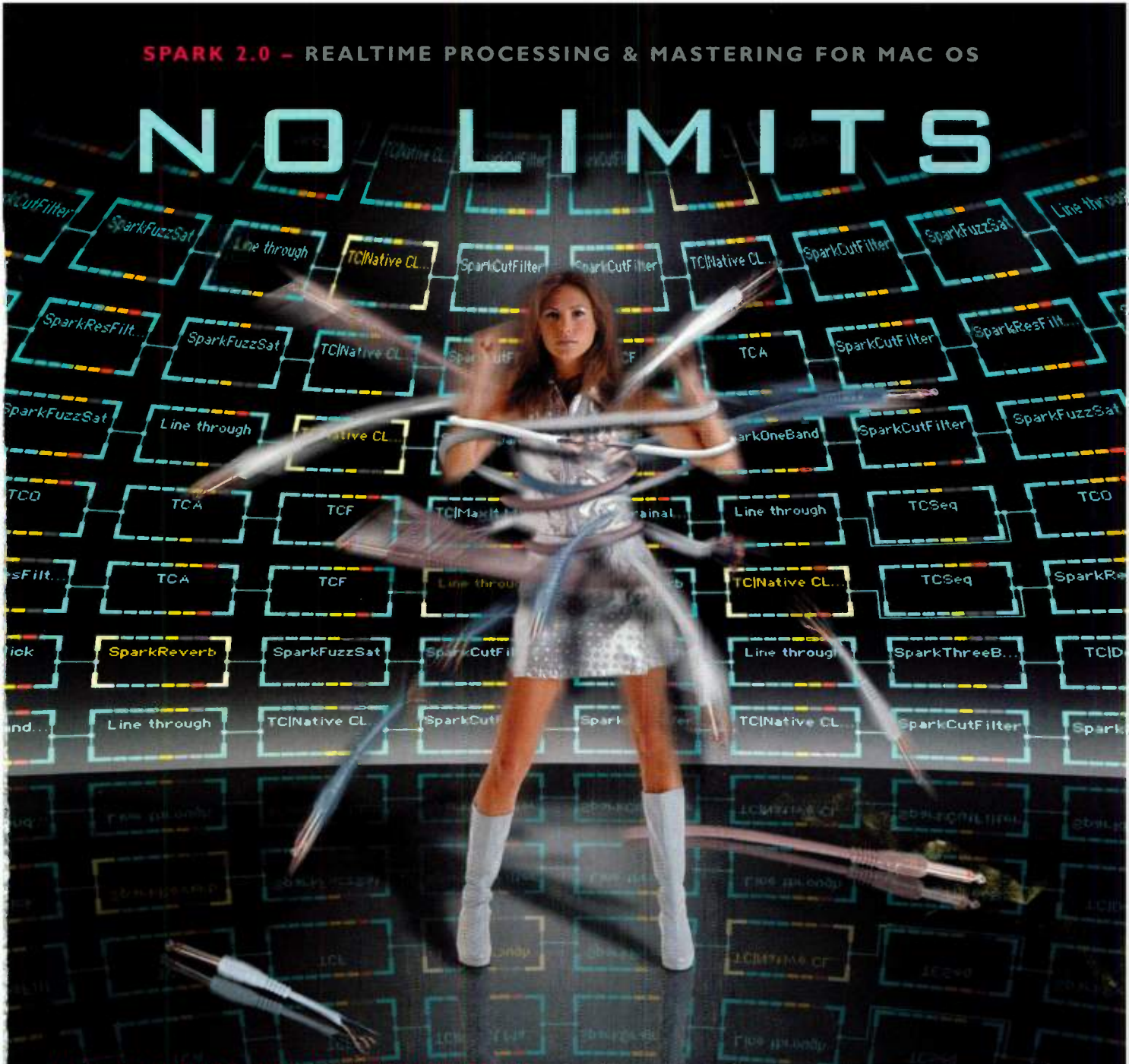

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are a viable alternative. For example, a shareware program that can use any sound card for input is available on the Web at http://polly.phys.msu.su/~zeld/oscill.html. You also can find a commercial product with a hardware interface that connects to the PC's parallel port and uses standard oscilloscope probes at www.designnotes.com/pcs64i.htm.

To use an oscilloscope, feed a test signal to the input of the device you want to evaluate and connect its output to the scope's input. The most obvious thing you can see with square waves on an oscilloscope is circuit *ringing*, a ripple in the flat part of the waveform (see **Fig. 8**). This is caused by unterminated energy bouncing back and forth in a circuit.

One common thing to watch out for is unwanted ringing in transformer-coupled circuits. This can happen when you connect the output of a classic piece of gear, such as a UREI compressor, to a return on a modern mixing console that does not exhibit the expected input impedance of 600Ω. To solve this problem, simply add a termination resistor across the terminal-strip output of the classic unit (see **Fig. 9**). The resistor should be the nearest value that is higher than the unit's transformer impedance; if the transformer's impedance is 600Ω, the resistor should be set to 680Ω. Adding this resistor can dampen the ringing, which improves the sound significantly.

A square wave is simply a symmetrical version of the pulse waveform, spending equal time at the low and high voltage values. Asymmetrical pulse waves are also useful; when combined with an oscilloscope, the waveforms can be used to determine if a particular circuit is inverting the polarity. If the input signal spends the majority of its time at the high voltage and the output signal spends most of its time at the low voltage, the device is reversing the polarity.

That can be a problem in situations you might never even imagine. I once had a digital reverb that sounded funny to me when it was switched to bypass mode. Evaluation with a pulse wave showed that it was passing proper polarity until the bypass switch was engaged. Then it inverted the polarity of the dry signal, which caused all manner of phase-cancellation problems in the mixing board.

RUNNING HOT AND COLD

One of the most basic tests for intermittent circuit failure involves heating and cooling circuit boards. Many circuits are fine on the test bench, but as soon as things heat up in a rack, they start to pop, buzz, or cut out. Back when a can of Freon cost just three dollars, people sprayed it on components to chill them. But Freon was wrecking the ozone layer, so alternative chilling agents became necessary (and the price of Freon shot way up). Those chilling agents are also quite expensive.

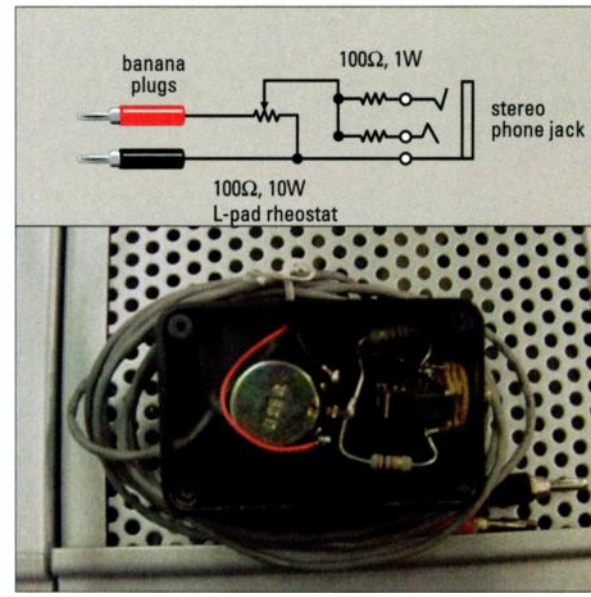
These days I wrap circuit boards in a conductive plastic bag (the type that electronic parts are shipped in) and place them in a freezer for a few hours. Then I reinstall the circuit board in the chassis and warm the components with a hair dryer while I listen for problems. If the problem originates from a heat-sensitive component, you should be able to figure it out with this method.

Flex a circuit board and components with a plastic ballpoint-pen barrel to help you pinpoint cold solder joints and breaks in circuit traces. As you do this, monitor the circuit with a signal generator and oscilloscope or just listen to music through it. By flexing the components one at a time, you should be able to reproduce the intermittent dropout, buzz, or other sonic problem.

In any event, do not use a pencil to push on a live circuit board because graphite is conductive and can cause electrocution. An electrical engineer at a company I worked for accidentally killed himself by touching a high-voltage point with a pencil. So remember, only use plastic for touching live circuits, and never put two hands in a live chassis.

Any solder joint in a circuit is a potential source of problems. By carefully inspecting a circuit board with a bright light and a magnifying lens, you can see any physical cracks in the joints. Cracks are especially common in road gear because vibrations experienced in transport put extra stress on solder joints in heavy components.

Look for any solder joint that's not perfectly shiny and bright. Anything that looks cloudy or grainy is probably a "cold" joint, the result of the component or wire being moved while the solder was cooling. In addition, component leads are not always properly cleaned and tinned (coated with solder) prior to installation. This



MIKE SOKOL

FIG. 6: To test a power amp output, you can build this headphone pad, which connects directly to the amp's output and lowers the signal to headphone-safe levels.

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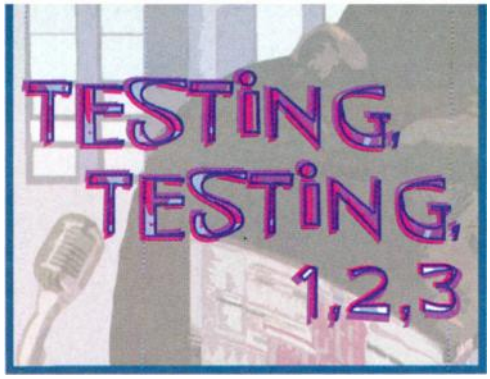

results in a mechanical connection that will eventually loosen because of temperature changes and vibrations. A proper solder joint creates a molecular-level bond between the wires that can't be vibrated apart.

If you need to fix solder joints, use a soldering pencil in the 25 to 35 watt range for normal electronic parts and a 150-watt gun for things such as heavy transformer wires. While heating the joint, add a small amount of 60/40 rosin-core solder. Never use solid-core or acid-core solder; solid-core solder doesn't flow properly around the joint, and acid-core solder uses an acid flux that will eventually cause all your wires to turn green and corrode. This voids all manufacturer warranties, and you might as well throw away the board once it's been attacked by the acid flux.

When you add solder to a joint, it should flow perfectly and look like a pool of mercury while cooling. If you accidentally move the wire, the solder pool will crystallize and turn cloudy. Simply reheat and hold it still. If there isn't enough heat to melt the solder in a few seconds, you need to use a bigger soldering pencil or gun. Holding a low-power pencil on a joint for a minute while trying to get the solder to melt will damage components because this heats up the whole part rather than doing a quick heat and go.

FIG. 8: By sending a square wave through a piece of audio gear and reading its output on an oscilloscope, you can see whether the circuit is ringing. If so, a ripple appears in the flat part of the waveform.

BREAK IT DOWN

One of the most important aspects of troubleshooting is getting yourself into the proper mindset. The first rule of fixing things is "Do no harm." You will do much more harm than good if you dive willy-nilly into a piece of gear before defining the problem and finding a way to isolate it.

The most complicated system in the world is only a bunch of simple machines that interact. Your job as a diagnostician is to figure out a set of tests that quickly point to the problem area. Then devise more tests to isolate the failed component. After you repair or replace the component, you're back in business.

For example, assume that one side of your P.A. system is down. You're sending a stereo feed from your mixer through a stereo equalizer, then to a two-way crossover, and finally to a pair of stereo amps connected to the right and left speaker stacks. If you think about the signal chain, several possible failures could cause one side to fall silent: a failure in one output channel of the console, in one side of the equalizer, or in one side of the crossover. It is also possible that one of the amps might have failed or all the speakers on one side might have blown, but these are less likely. To be sure the amps aren't the problem, swap the cables connected to the amp outputs; if the problem shifts to the other side, you've got a bad amp channel.

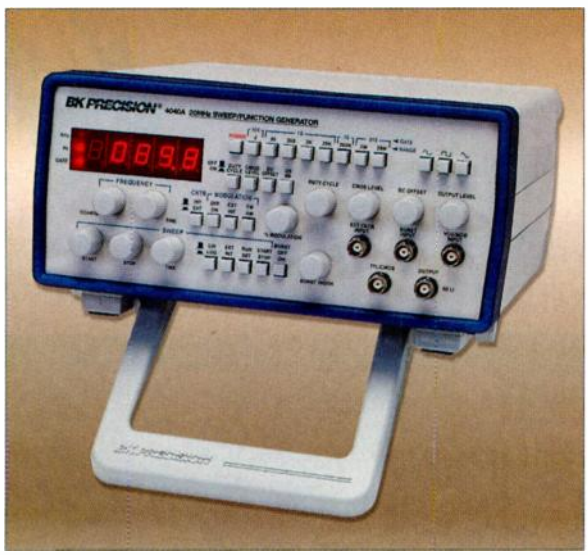

FIG. 7: A signal generator is an excellent source of test waveforms such as sine, square, and pulse.

Assuming it's not the amps, start at the beginning of the chain and work your way along it. Mute the amps so you don't pop any speakers and then switch the left and right outputs from the console to the equalizer. If the opposite speaker bank starts operating and the previously working side quits, you have nothing coming from one side of the console. That could mean anything from a failed driver chip to something really silly like a muted output channel. Don't overlook the obvious, especially with a console you're not completely familiar with.

If swapping the console outputs doesn't change the working stack, return the outputs to their original configuration and move down the chain to the next piece of gear. Maybe the equalizer or crossover has a dead channel. If the EQ uses XLR connections at the inputs and outputs, pull the male and female connectors and patch them together, taking the equalizer out of the signal path. If the speakers on the bad side suddenly start working, you've isolated the problem gear. If the speakers still refuse to work, the problem is probably in the crossover.

When all the electronics seem to be working properly, it's time to test the speakers. You can do this with a VOM set to its lowest resistance range, usually 200Ω. Touch the meter's leads directly to the speaker's input terminals.

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Most speaker failures involve an open voice coil (a break in the voice coil's windings), which will result in an infinite resistance (or at least a value in the megaohm range).

Remember that DC resistance will always be lower than the rated impedance of a speaker because of all that reactive stuff, such as coil windings, internal crossovers, magnets, and so on. As a result, a speaker rated at 8Ω might actually read 4 to 6Ω on a VOM. However, it shouldn't read 0Ω (a short), which can happen when the voice-coil windings touch the side of the magnet gap and short out. You can test for this type of failure by gently moving the speaker cone in and out with your finger while metering it. If you feel a rough spot and the meter reads a short, the voice coil is warped and the speaker needs to be reconed.

By the way, don't overlook something as simple as a bad cable. A $10 cable can bring a million-dollar sound system to its knees, so keep some good patch cables handy to swap into the problem area.

If you have problems with something such as a compressor that's connected with the separated ends of a TRS insert cable, try running a signal through the affected channel, pulling the ¼-inch TS plugs from the input and output of the processor, and touching the tips of these plugs together. If the entire cable is correctly wired, the signal should pass through when you touch the tips together and stop when you separate them. Many times, an insert patch doesn't work because the user simply plugged the input and output connectors into the wrong jacks on the processor. This simple cable check will enable you to quickly isolate a bad cable, bad processor channel, or reversed patch.

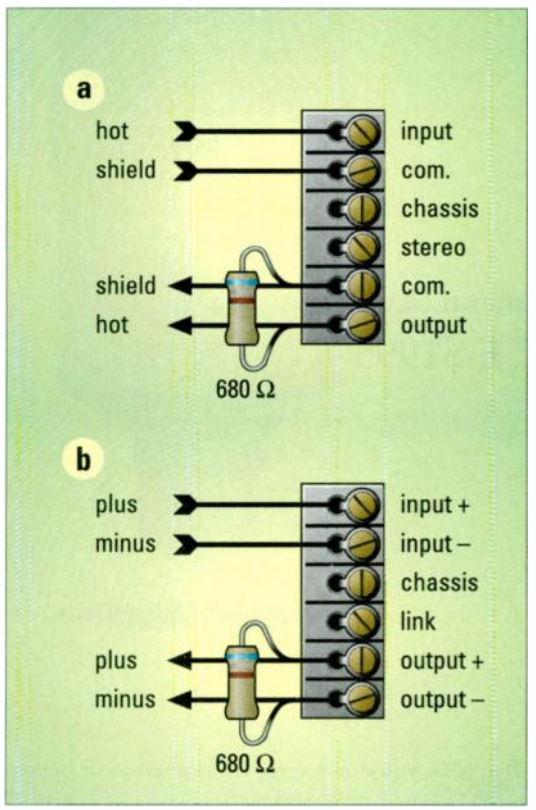
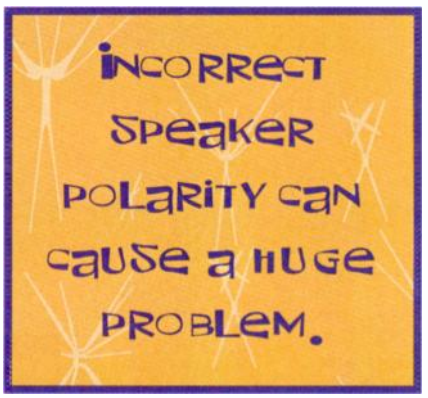


FIG. 9: Stop ringing in a piece of classic transformer-coupled, low-impedance audio gear connected to a high-impedance mixer input by placing a resistor across the output terminals. This works well with balanced outputs (above) and unbalanced outputs (below).

SURROUND CALIBRATION

If you have a surround-sound studio system, it's important to confirm that all six speakers are at the proper relative levels and wired with the same polarity. (For more on surround systems, see "You're Surrounded" in the October 2000 issue of **EM**.) In a stereo system, level and polarity are easy to verify. If the stereo image is leaning to one side, one of the amplifier channels is set with too much gain. If the speakers are wired with opposite polarity, the bass sounds "phasey."

But with 5.1 surround, it's much more complicated. Knowing that the center and rear speakers are at the same polarity as the front left and right speakers is critical. In addition, you must make sure the low-frequency effects (LFE) channel has 10 dB of extra gain relative to the main speakers. To do that properly, you need a sound-pressure level



(SPL) meter, available from Radio Shack, and a polarity tester such as the Galaxy Cricket.

If you're using a mixing console with any sort of joystick panner for the surround speakers, find a source of full-bandwidth pink noise, such as the console itself, a real-time analyzer, or a test CD. After muting all speaker outputs, patch the pink-noise signal into one of the channel strips and pan it to send a –20 dBFS (decibels full scale) signal to all six outputs.

Next unmute the left-front speaker and place an SPL meter at the listening position, pointed directly at that speaker. Adjust the gain of the corresponding amplifier channel so the SPL meter reads 85 dB SPL, C-weighted. Now mute the left-front speaker and repeat this procedure for the center, right-front, left-surround, and right-surround speakers one at a time. Be sure to point the SPL meter directly at the speaker being tested while keeping it in the center listening position.

This is a good time to ensure each channel is being sent to the correct speaker; the left-front output of the console should go to the left-front speaker, the right-rear output to the right-rear speaker, and so on.

After the five main speakers are set to

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JUST LISTEN.

the correct level, mute them all and feed the pink noise to the subwoofer only. According to 5.1-surround specification, the gain of the LFE channel should be set so the SPL in the room is 10 dB higher than the other speakers. Because there are just two to three octaves of bass information, however, this much gain will only register about 4 dB above 85 dB SPL with a typical meter. So set the subwoofer's amplifier gain so the meter reads 89 dB SPL on the C-weighed scale (4 dB more than the rest of the speakers).

This technique for calibrating the LFE level is an approximation at best. To set it properly, you need to use a real-time analyzer (RTA). After setting each main speaker to 85 dB SPL, turn on the subwoofer only. With the same –20 dBFS pink-noise signal, set the gain on the subwoofer so the RTA reads 95 dB SPL—10 dB above the 85 dB SPL reference—at frequencies below about 100 Hz. You should probably do this at least once in your studio and note the reading your SPL meter gives you at this level. Use that reading to verify the speaker levels before every surround mixdown session.

Next, verify speaker polarity by removing the pink-noise source and applying the clicker's test signal to the same channel strip, feeding it to all the speakers simultaneously. (Turn the volume down low first so you don't blow any drivers.) Put the clicker's polarity listener up to each speaker and verify that they share the same polarity. If a speaker's polarity is reversed, swap the speaker leads and try again.

Many surround setups, especially those for movie soundtracks, utilize *dipole* surround speakers, which have pairs of drivers facing front and back. In these speakers, the front-facing drivers are supposed to be reversed in polarity with respect to the rear-facing drivers. A variation of this design called a *bipole* speaker uses front- and rear-facing drivers that are not reversed in polarity, though these are less common.

WRAPPING IT UP

The key to using test equipment properly is to take your time and break the most complex problem into a series of small tests. Then evaluate each part of the sound chain to determine the source of the problem. That is not always easy when an angry producer is hanging over your shoulder and wondering why there's a buzzing noise in one of your monitors. But if you are accustomed to testing your cables and gear in your own studio, there is a great chance that you can do it elsewhere and under pressure. Just be sure not to skip any safety precautions and to focus on the individual part being tested.

Mike Sokol *is a live-sound and recording engineer with 30 years of experience on both sides of the console.*

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MUSIC FOR A Maharajah

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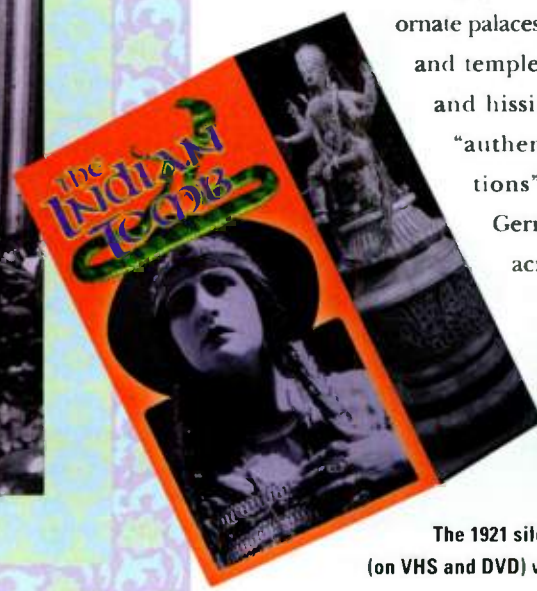
Joe May's 1921 action/thriller masterpiece *The Indian Tomb* has long been considered one of the greatest cinematic epics of all time. Budgeted at more than 20 million German marks (a very large sum at the time), this lavish two-part adventure film takes place in an atmospheric Indian setting of romantic imagination, complete with mystical yogis and dancing girls, ornate palaces and temples, roaring tigers, and hissing cobras. (The "authentic Indian locations" were filmed in Germany at May's 50-acre "film city.") The legendary Conrad Veidt heads a large cast while having a field day as Ayan, the charismatic, sadistic Maharajah of Bengal. Written by Fritz Lang and Thea von Harbou, the serial-style plot has Ayan carrying out an elaborate plan of revenge against his wife and her English lover. With the aid of a reluctant high yogi, Ayan brings a European architect to his palace under a cloud of secrecy to design a massive tomb, just so the evil Maharajah can supply the occupant! The architect's girlfriend (played by Mia May, Joe May's wife) follows him to India and immediately catches Ayan's licentious eye. Thus begins a roller-coaster ride of tense climaxes and assorted plot devices that include sexual extortion, man-eating tigers, exotic festivals, and chases across a crocodile-infested lake (see **Fig. 1**).



BY ERIC BEHEIM

The 1921 silent film *The Indian Tomb* is now available in a digitally restored version (on VHS and DVD) with a new score compiled and produced by Eric Beheim.

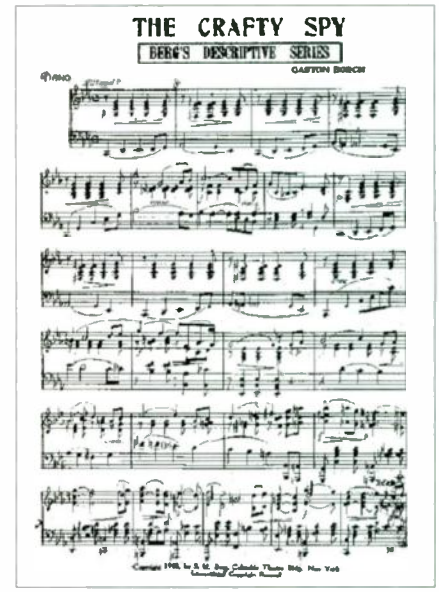
music for a maharajah

Long unavailable, a new, definitive version of *The Indian Tomb* has been digitally mastered for the home-video market by David Shepard, who is internationally known for his work in preserving classic silent films.

In restoring this visually stunning masterpiece to its former glory, Shepard faced many challenges, not the least of which was coming up with more than three and a half hours of appropriate music to match the exotic and sometimes fantastic action taking place on screen. When budget considerations ruled out such niceties as original scoring and the use of a live orchestra, Shepard turned to me. Working with a modest collection of home-studio gear (located in one corner of the family room), I previously supplied Shepard with electronic scores for the films of diverse screen personalities such as Buster Keaton, Boris Karloff, Clara Bow, and Koko the Clown.

As most people know, the so-called silent films were never actually shown in silence. Even in the earliest nickelodeon days, theaters provided a piano or small string ensemble to play during the performance. In many instances, the music bore little or no relationship to what was happening on screen. Later, as films became longer and more sophisticated, it became apparent that the right choice of music was a key factor in making the proper emotional connection between the film and its audience. Early blockbusters such as *The Birth of a Nation* (1915) and *The Battle Cry of Peace* (1916) had specially prepared scores performed by full orchestras.

At about the same time, music publishers began to offer special incidental mood music specifically written for picture use. By the end of the silent-film era, "photoplay music" was available for almost any situation or setting likely to be encountered on screen (see **Fig. 2**). Rendered obsolete by sound films, most of this specialized material disappeared into landfills or scrap-paper drives years ago. While I was still a music performance major in college, I unexpectedly came across a stack of silent-movie music in a thrift store. For 25 cents, I bought the entire lot. When I showed the music to my classmates and professors and declared that someday I would like to synchronize music to silent films, they thought I was crazy.



COURTESY ERIC BEHEIM

FIG. 2: This 1918 composition by Gaston Borch is typical of incidental mood-music selections produced during the silent-film era.

Undaunted, I began to collect anything related to the music used in silent-film theaters. My collection received a significant boost a few years later when a man in Iowa answered an ad I'd placed that asked for old film music. His reply stated that he had a complete library—more than 4,000 orchestral arrangements—that had come from an old theater and asked if I was interested. I couldn't get the check to him fast enough!

At first I prepared scores and parts for classic films such as *Phantom of the Opera* and Buster Keaton's *The General.* Through my company, Cine-Phonic Music Service, I rented those scores to major symphony orchestras in the United States and Canada for silent-film concerts. In 1996, Shepard called me to do an electronic score for a video release of Douglas Fairbanks' 1922 smash hit *Robin Hood.* Before I even finished the project, I realized that I had found my calling.

Although it was longer and more complex than my previous assignments for Shepard, the score for *The Indian Tomb* followed the same general process as the others. Here's a brief description of how I created the score.

COURTESY DAVID SHEPARD/WATER BEARER FILMS

FIG. 1: The lavish production of *The Indian Tomb* is marked by elaborate sets and plenty of high drama.

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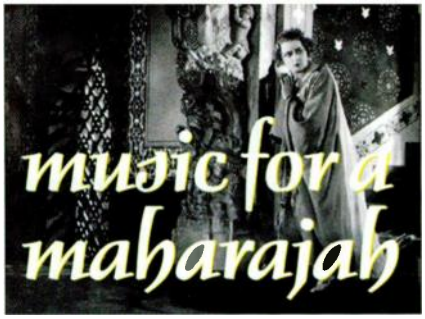

THE BREAKDOWN

Work officially began with the arrival of the film on a videotape workprint striped with SMPTE time code. The tape included a time-code window that allowed me to identify the exact location of any scene or piece of action in terms of hours, minutes, seconds, and frames. After screening the film once so I could become familiar with its plot, characters, and action highlights, I went back and broke down the movie into sequences. The running time for each sequence was then computed in minutes, seconds, and frames.

That information became the first draft of the cue sheet that would eventually serve as the blueprint for the entire soundtrack. The cue sheet, however, was subject to frequent revisions during music selection and editing. Often, I combined several sequences or broke down a long sequence into shorter segments. The final cue sheet consisted of 205 sequences running from a few seconds to several minutes in length.

THE SELECTION PROCESS

Silent films that are released on video have scores using everything from rock and avant-garde to new-age music. But because I'm a stickler for authenticity, I insist that the score be composed of music that is contemporary with the film. *The Indian Tomb* initially played theaters in 1921 and 1922, so I limited myself to the music that would have been in print at or before that time.

In keeping with the film's Eastern setting, I began my search in the Oriental section of my music library. Rimsky-Korsakov's *Antar Symphony* provided the haunting theme that I used for Veidt's Ayan character. I underscored the big festival scene, featuring multitudes of soldiers, elephants, and horses, using "March of the Mogul Emperors" from Elgar's *Crown of India Suite*. "Patrol of the Boxers" by Irenee Berge, composer of many silent-film themes, accompanied a journey by elephant train to the tomb site.

To acknowledge the film's German origin, I worked in quotations from Wagner's *Ein Faust Overture* and the funeral music from *Gotterdaemerung*. Rimsky-Korsakov's *Le Coq d'Or* and Ippolitov-Ivanov's *Caucasian Sketches* provided material for some of the more mysterious moments. One important action sequence uses "Orgies of the Spirits" by Ilynsky, a selection often heard in 1930s adventure films such as the Flash Gordon serials. I also employed selections from Delibes's opera *Lakame*, Cesar Cui's *Orientale*, and

The Indian Tomb
Conrad Veidt/1921
Music Score Compiled by Eric Beheim
1/2/00

1. AT SCREENING PART I (1:38:21) Lakame Selection (Delibes arr. Roberts/1917)	1:00:00:00
2. T: "In the Shadow" (:58:09) Weird Oriental Theme (Levy/1918)	1:01:38:21
3. D: CU Prince (:16:03) Antar I (start at B) (Rimsky-Korsakov/1897)	1:02:37:00
4. D: Uncovering grave (1:01:05) Weird Oriental Theme (Levy/1918)	1:02:53:03
5. D: Body of Yogi in grave (:33:04) Antar I (Largo) (Rimsky-Korsakov/1897)	1:03:54:08
6. D: Removing body from grave (:47:05) Weird Oriental Theme (Levy/1918)	1:04:27:12
7. D: Inside temple (1:33:19) Le Coq d'Or (Lento) (Rimsky-Korsakov/1907)	1:05:14:17
8. D: Roof of temple (3:41:15) Sunrise and Incantations (Borch/1917)	1:06:48:06
9. D: Prince and Yogi (1:52:22) Antar I (from B) (Rimsky-Korsakov/1897) (2 sections)	1:10:29:21
10. D: Yogi rises (:58:07) Indian Invocation (Herbert arr. Langey/1911)	1:12:22:13
11. T: Part 1 (:47:29) To a Lotus Flower (Berge arr. Jungnickel/1920)	1:13:20:20
12. D: Irene behind Herbert (1:16:05) Ah, Moon of My Desire (Lehmann/1909)	1:14:08:19
13. T: "Irene's father" (:11:29) Song Without Words (Mendelssohn)	1:15:24:24
14. D: Irene and Herbert (1:13:14) To a Lotus Flower (Berge arr. Jungnickel/1920)	1:15:36:23
15. D: Irene's father (:22:04) Song Without Words (Mendelssohn)	1:16:50:07
16. D: Empty chair (1:58:17) Eastern Romance (YOGI THEME) (Rimsky-Korsakov arr. Langey/1918)	1:17:12:11
17. D: Irene's father (:32:06) Song Without Words (Mendelssohn)	1:19:10:28
18. D: Herbert and Yogi (:15:25) I-10C (English Horn solo)	1:19:43:04

FIG. 3: The cue sheets prepared for *The Indian Tomb* include SMPTE times on the far right to indicate the location of each sequence. The numbers in parentheses show the running times in minutes, seconds, and frames.

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Berge's *Oriental Suite* to good effect.

Rounding out the score were silent-film standbys such as "Weird Oriental Theme" by Sol P. Levy, "Sunrise and Incantations" by Gaston Borch, and "La Foret Perfide" by Gabriel-Marie. (The fact that all of this music was in the public domain was another bonus for producer Shepard.) As I selected music, I matched it to sequences on the cue sheet. Often, the music for a given sequence had to be changed several times before I felt that it evoked the proper mood and feeling (see **Fig. 3**).

Using Coda's *Finale* music-notation program, I sequenced the individual instrumental parts using the original orchestrations. For its audio output, my PC-based home studio relies primarily on a collection of E-mu sound modules, including a Proteus 2000, two Proteus/2s, and an ESI-32 sampler. The outputs from the sound modules are routed to a Mackie 1402 VLZ 14-channel mixer, and the combined signal is sent to a Lexicon Alex effects processor, which adds the approximate ambience of a medium-size theater (see **Fig. 4**).

My ultimate goal was to emulate the sound of a small pit orchestra that might have been used in a medium-size theater in the 20th century's second and third decades. Through trial and error, I came up with a combination of instrument voices that blended well together. For *The Indian Tomb,* this nucleus "orchestra" had to be expanded to accommodate the lush Oriental numbers, which often made prominent use of bassoon, oboe, and English horn. With so many instruments playing at once, extra care was required to balance individual voices to achieve as realistic a sound as possible. Following the sequencing and mixing process, I edited the music to match the running times listed on the cue sheet.

THE RECORDING

My hard drive recording setup consists of a second PC equipped with a Digital Audio Labs CardD Plus sound card and an Opcode Music Quest MQX-32M card that generates the SMPTE time code. Innovative Quality Software's *SAW* multitrack program provides the four stereo tracks that I need as well as the time-code support.

Throughout the recording session for *The Indian Tomb,* my son Bret, a high school senior and computer whiz, handled the recording and editing chores while I looked after the virtual orchestra. Recording was done at a sampling rate of 48 kHz. Once the music was finished, a few key sound effects were dropped into place. After playback in interlock with the video, only a few minor corrections were left to be made.

The final step was to record the composite soundtrack (music and sound effects) onto DAT for final delivery. As an extra courtesy to the producer, I used *SAW* to generate a 44.1 kHz WAV file of the composite track, which I used to burn an audio CD. I provided that CD along with the videotape to Shepard as a producer's advance copy. By starting the CD and videotape together, Shepard could preview the soundtrack in reasonably close sync with the picture before going into the studio for the final playback. (We jokingly refer to those CDs as Vitaphone disks, after the early sound-on-disk playback system used in theaters during the late 1920s.)


PAMELA HENDRICKSON

FIG. 4: Eric Beheim's home studio fits into a corner of his family room. A plaque displayed nearby says it all: This or Something Better.

With *The Indian Tomb* safely transferred, I had one last chore to perform: coming up with a name for the "orchestra" for the credits at the end of the film. For past projects, most of the names have been in-jokes having to do with the movie or its star. (For *Robin Hood,* my electronic musicians were billed as the Elton Thomas Orchestra, a reference to the pen name Douglas Fairbanks used for his film scripts.) After much serious consideration, I finally decided that *The Indian Tomb* orchestra would be billed as the Tiger Hof Kapelle. In the film's original German titles, Tiger Hof was the walled enclosure in which Ayan kept his pack of murderous pet tigers.

FADE OUT

With all of my *Finale* orchestra files backed up onto a CD, I began to think about the music for my next assignment for Shepard: a trio of films about the Civil War produced by Thomas Ince from 1913 to 1915.

My success in producing composite soundtracks for commercial video releases of silent films demonstrates that owning a room full of expensive gear is not necessary to be a viable player in the electronic-music field. Primarily through trial and error, I've learned how to make the most of a very basic studio setup, which has provided me access to an area that I find personally and financially rewarding.

I'm constantly working to refine my technique and to gain an even better understanding of what my modest system is capable of. Following the completion of each project, I make a point to reinvest some of the profits into system upgrades and perhaps one or two new components to increase my studio's capabilities.

Eric Beheim *leads a multifaceted career as a professional musician and informational video producer. Major symphony orchestras in the United States and Canada have performed his silent-film scores.*

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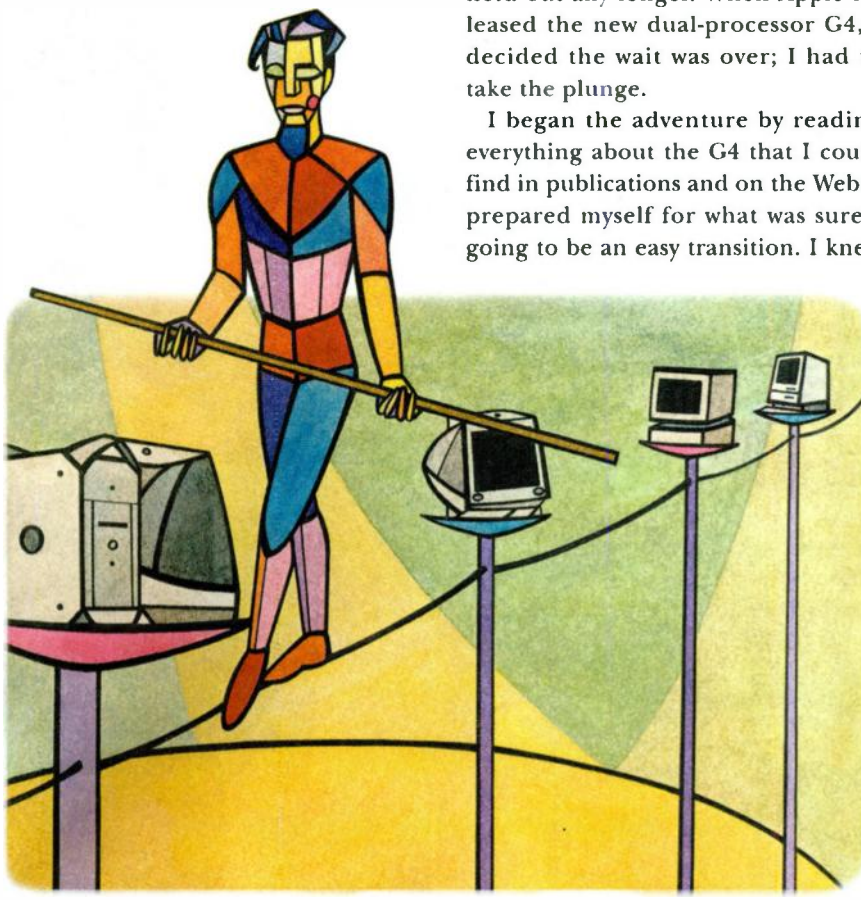
By Rick DiFonzo



JESSICA KASPER

After my trusty Power Mac 8600 had provided me many years of faithful service, I reluctantly demoted it to the role of homework computer for my daughter. I couldn't hold out any longer. When Apple released the new dual-processor G4, I decided the wait was over; I had to take the plunge.

I began the adventure by reading everything about the G4 that I could find in publications and on the Web. I prepared myself for what was surely going to be an easy transition. I knew what I needed, I knew what to expect, and I thought I was ready. I wasn't. The changeover still took more than a week to complete. Perhaps by sharing my experience, I can help some of you avoid the annoyance of navigating endless tech-support phone menus and listening to horrible music while on hold.

TALES OF THE UNEXPECTED

I already knew I'd need some adapters. I budgeted for gobs of RAM, and I also ordered a speedy FireWire CD burner and a second internal ATA hard drive for additional storage and recording. I planned to spend three or four days installing, learning, and experimenting with new software. A few things, however, I hadn't counted on.

Apple has made some pretty dramatic changes since the PPC 604 machines appeared on the market, but not all of them are immediately apparent. The switch from SCSI, serial, and ADB to USB and FireWire has been well documented for some time, but less known is that in 1997 Apple updated its PCI specification from version 2.0 to version 2.1. As a result, many PCI cards manufactured before then must be upgraded to function in the newer machines. In some instances, it's no big deal: for $85, Digidesign replaced the outdated chip

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FIG. 1: The Magma PCI Expansion System houses six PCI slots in a rack-mountable chassis.

on my SampleCell II card and sent it to me the same day. If you're using PCI cards manufactured before 1997, ask the manufacturer if they need to be upgraded.

I didn't realize that Digidesign's tried-and-true *MasterList CD* software requires a SCSI CD burner. I quickly found that my FireWire CD-R drive was useless, and I needed to fill a PCI slot with a SCSI card if I wanted to use *MasterList CD*. You may encounter a similar problem with older SCSI-dependent programs that have not been upgraded recently, and because the new Macs offer only three PCI slots, any such complication may come as an unpleasant surprise. If you want to use SCSI devices and you are willing to purchase your computer directly from the Apple Store (store.apple.com), you can get a customized G4 with a SCSI card and drive instead of the internal ATA drive.

Fortunately, there are solutions to the slot-shortage problem. One solution is an external PCI expansion chassis, a card cage that provides multiple PCI slots though it uses only one PCI slot in your Mac. One of the more popular expansion boxes is the Magma PCI Expansion System, which comes in portable, tower, and rack-mount versions that house 1 to 13 PCI cards (see **Fig. 1**). The 6-, 7-, and 13-slot models are approved by Digidesign for use with Pro Tools cards. However, expansion chassis are pricey; the 6-card Magma cage lists for $1,450 (though as of this writing, it is on sale for $1,195), and the Digidesign-approved 13-slot version lists for $1,995. (For a list of contacts for all companies named in this article, see the sidebar "Upgrade Tool Makers.")

My old friend *MasterList CD* threw me yet another curve when I realized that its copy-protection scheme requires a floppy drive for authorization. You may run into the same problem if you're still using older software. Digidesign recommends two drives: the Imation SuperDisk ($149), which works with regular 1.44 MB floppies or 120 MB SuperDisks, or the Newer Technology uDrive ($89), which only supports 1.44 MB floppies. I opted for the less-expensive uDrive (see **Fig. 2**) because it's unlikely that I'll ever need SuperDisks.

Of course, I could have avoided all those problems by simply switching to another CD-burning program. For example, Roxio's *Jam* doesn't require a floppy drive for authorization. It supports USB and FireWire CD burners and reads *Sound Designer II* regions and playlists. Check the company's Web site for a list of supported drives.

Aside from the more common problems you might expect during an upgrade, be prepared for an occasional hardware glitch to slow down the transition. In my case, the new machine's DVD-RAM drive had intermittent problems and often refused to open. That meant spending an afternoon on the phone with Apple tech support while trying all sorts of software reinstalls. In the end, unplugging the drive's power cable and reseating it solved the problem.

PLUGS AND PORTS

If you're new to USB, you should expect to purchase a USB hub right away. The new Macs have two USB ports. One is normally used for the keyboard, mouse, and perhaps a copy-protection dongle. However, to add a scanner, printer, floppy drive, MIDI interface, CD burner, and a control surface, you'll need at least one USB hub with plenty of ports, and you are probably better off connecting a hub

FIG. 2: Newer Technology's affordable uDrive adds floppy disk capabilities to the new Macintosh computers.

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to each of the Mac's two USB ports.

Several inexpensive models are worth considering, including the $100 Belkin USB BusStation 7-port hub. The BusStation system is modular, and you can replace some of the USB modules with SCSI, serial, or ADB adapter modules, which allows you to use the hub as a USB-to-whatever converter. (Note that the USB-to-serial adapters will not work with externally clocked devices such as MIDI interfaces, and the USB-to-SCSI adapter operates at USB speed, which is slower than SCSI.)

If you want to continue using your existing serial MIDI interface, check out the GeeThree Stealth Serial Port ($49.95), a small card that replaces the Mac's internal modem and provides a single serial port. No other serial adapter I've found works with a serial MIDI interface. The Stealth works like a normal serial port because it uses the internal modem slot and is based on the same serial chip set used in earlier Power Macs. Installation is a bit trickier than snapping a PCI card into place, but if you have steady hands and can operate a screwdriver, you shouldn't have much trouble.

I didn't experience any performance lag with this setup, and I saved a few hundred dollars by installing the Stealth Serial Port and using my old MIDI interface. (If you buy your Mac from the Apple Store, you can save $100 by ordering it without the internal modem. If you need a modem, you can easily find an inexpensive external USB modem.) As my studio grows, I can always add a new USB MIDI interface and use both.

However, to use the computer's internal modem or multiple MIDI interfaces, you'll have to spring for a USB MIDI interface. If you also need another port for, say, your old serial printer, you'll have to get a USB-to-serial adapter. The $80 Keyspan USB Twin Serial Adapter works fine.

For my sequencer, I use Emagic's *Logic Audio*, which employs a hardware dongle for copy protection. As luck would have it, USB dongles were temporarily out of stock when I set up my new Mac. So to use my old ADB dongle, I had to get a USB-to-ADB adapter. At $39, the Griffin iMate USB-to-ADB adapter is an excellent choice (see **Fig. 3**),

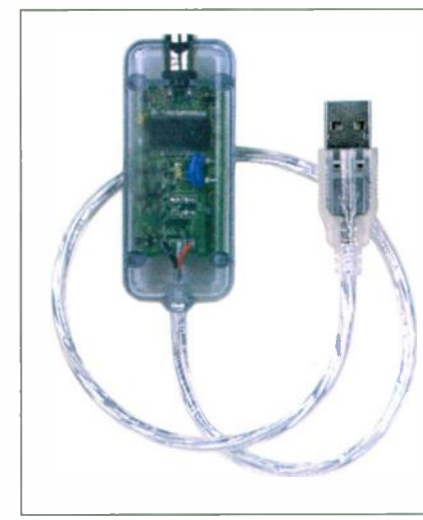

FIG. 3: The Griffin iMate USB-to-ADB adapter lets you use some old ADB dongles with the newer Macs.

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although I was stymied when I first tried to use it. Unlike most of the peripherals and adapters I installed, the iMate didn't come with driver software. Because the packaging had no mention of a driver and no disks were included, I assumed a driver was not necessary—wrong again. I couldn't figure out why *Logic Audio* wouldn't recognize the dongle and boot. After hours of hair pulling, a response from an Emagic newsgroup informed me that I had to go to the Griffin Web site and download the appropriate driver, which could be a problem in studios without Internet access.

THE MACHINE

As you've probably heard, the new G4s are wonderfully fast. Installations race by, applications boot in a matter of seconds, and Web pages load in a fraction of the time that it took with my old machine. Apple's newly redesigned keyboard with full-size function keys is a small improvement over its compact predecessor, but it still takes some getting used to when changing from a full-size ADB keyboard. Moreover, not everyone likes the feel of the new keyboard's key switches. Fortunately, several companies offer excellent replacement USB keyboards. Although keyboards are readily available through mail-order houses, buying one at a local dealer where you can compare the feel of several keyboards is best.

The standard ATA hard drives that come with the G4 should be big enough and fast enough to get you going. In fact, I haven't had any problems at all recording to the boot drive. You might want to add drives later, though, and doing so is easy and inexpensive.

The ATA specification differs from SCSI in that it only allows two drives per bus. With fast 30 GB drives going for $170 or less and two ATA buses to connect them to, you can store a lot of audio without breaking the bank. (Keep in mind that ATA drives must spin at 7,200 rpm or faster and should have an average seek time of about 9 ms or less to record audio reliably.) FireWire boasts higher performance, but FireWire drives cost more. USB drives are simply not fast enough right now, and there is some question about whether Apple will implement USB 2.0, which offers greater bandwidth. (For more about hard drives and storage options, see "Desktop Musician: U Store It" in the February 2001 issue of **EM**.)

SOFTWARE SURPRISES

If you're upgrading from Mac OS 8, you won't see a huge difference in the OS 9 interface other than *Sherlock* 2.0 and some spiffy new desktop patterns. However, QuickTime 4 is very cool, with improved streaming-video and audio capabilities.

The Open Music System (OMS) extension, required for some MIDI programs, reportedly has some stability problems running under Mac OS 9.0.4, but I didn't encounter trouble with my setup. If you are using Digidesign gear or other hardware and software combinations that require OMS, you may experience problems. Check the manufacturer's Web site for compatibility issues and possible workarounds. When starting up *Pro Tools LE*, for example, I was forced to manually configure OMS to recognize my rig. It wasn't a big deal, though, and I've experienced no trouble since.

The new dual-processor G4 has great potential, but don't forget that the second processor offers no speed gain unless the software you are running is optimized to take advantage of it. (That will not be the case with programs written for Mac OS X, which is inherently multiprocessor-aware.) Emagic's *Logic Audio Gold* and *Platinum* 4.5.1 and Steinberg's *Cubase VST* 5.0 split the tasks of processing MIDI and audio information between the two chips. Steinberg estimates that with *Cubase VST* 5.0, you get a 50 to 60 percent increase in processing power over a standard single-processor machine.

I can state from personal experience that *Logic Audio*'s dual-processor support really makes a difference. I can easily get 24 tracks going at once, along with tons of time-based plug-ins and virtual synths. Mark of the Unicorn's *Digital Performer* will support dual processors in the next revision, which probably will be available by the time you read this. *Pro Tools* 5.1

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Web www.roxio.com

does not support dual processors as of this writing.

Some veteran Mac-based musicians are concerned about the demise of Digidesign's *Sound Designer II*. If you want to use that venerable program on a G4, you must have an Audiomedia III card installed. You will also need to purchase the $59 upgraded installer version 2.8.3 from Digidesign.

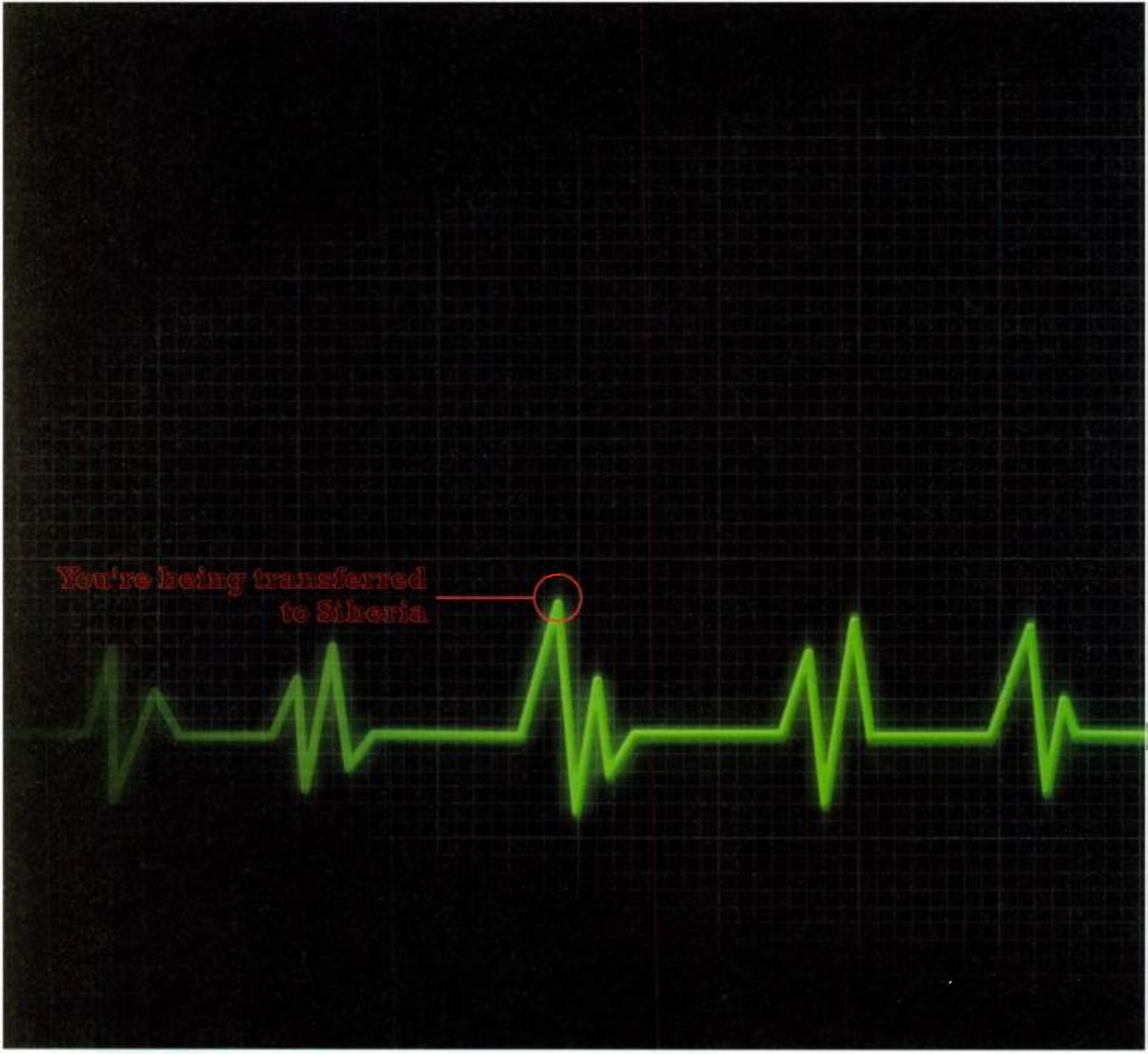
SHOULD YOU?

If you want to greatly enhance your recording, mixing, and general computing experience, the upgrade process is certainly worth the possible headaches and extra expense. Although I managed for a long time to do a lot of interesting work on an old machine, having all this new power has changed the way I think and work. You can always go back to your old ways, but having a wider range of options at your command can be inspiring.

Many of the problems I faced stemmed from the fact that I was not only upgrading from an ancient Mac but also installing new audio hardware and software at the same time. If you don't have to upgrade your audio card right away, wait. Settle in with the new system and update other elements of your studio later. That will avoid an avalanche of confusion and enable you to focus on one problem at a time.

Life on the bleeding edge of technology can be dangerous. Brand-new hardware and operating systems often introduce new problems for the software that you use. Unfortunately, waiting around may not be the best policy because the upgrade path just gets steeper as new technologies continue to evolve. If you're still clinging to your tattered old computer, it's time to jump on the technology merry-go-round and hold on tight—you're entering yet another season of new product releases.

Rick DiFonzo *is a New York City composer, producer, and guitarist who has written music for the small screen, radio, records, and bowling alleys.*



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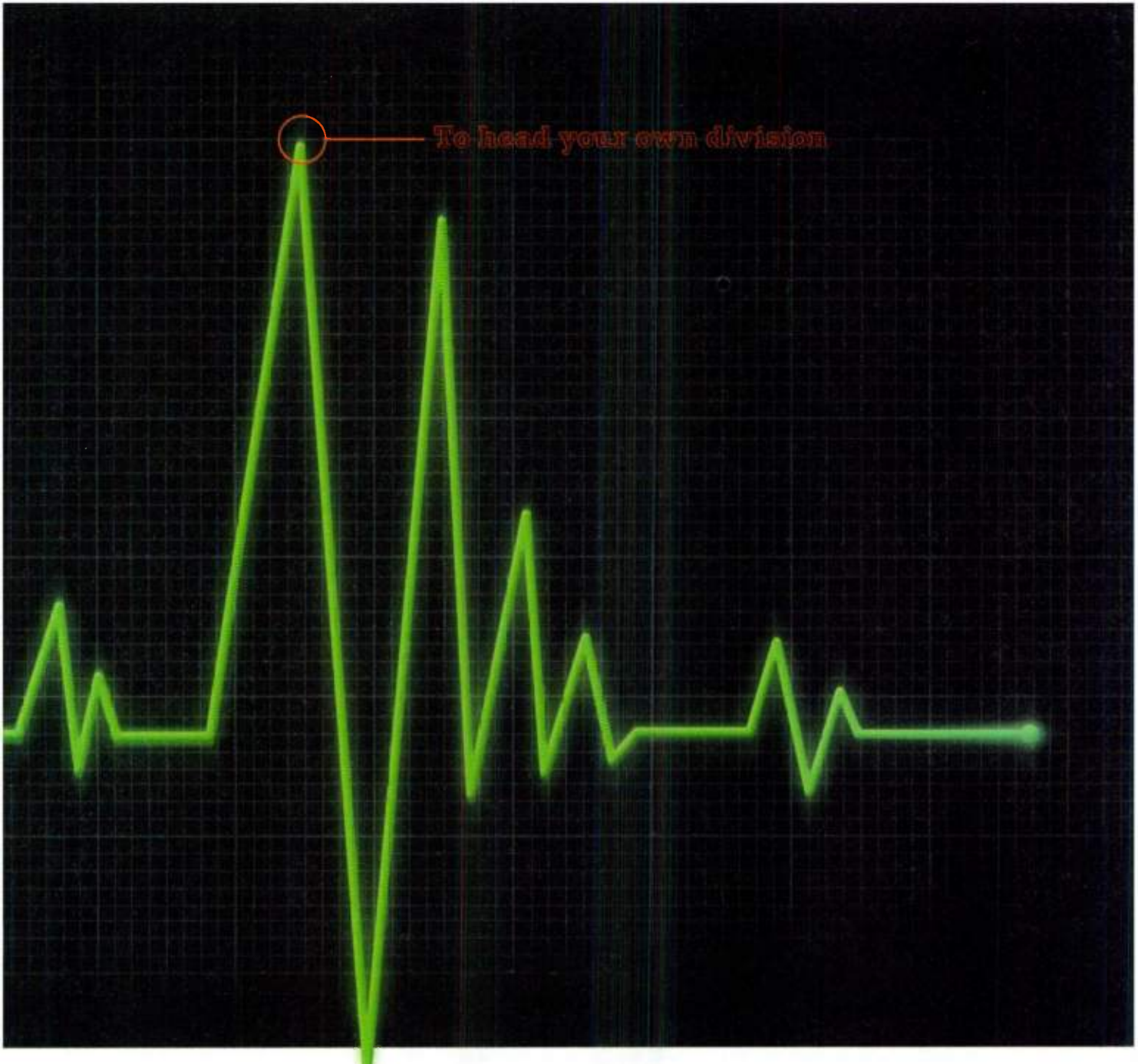


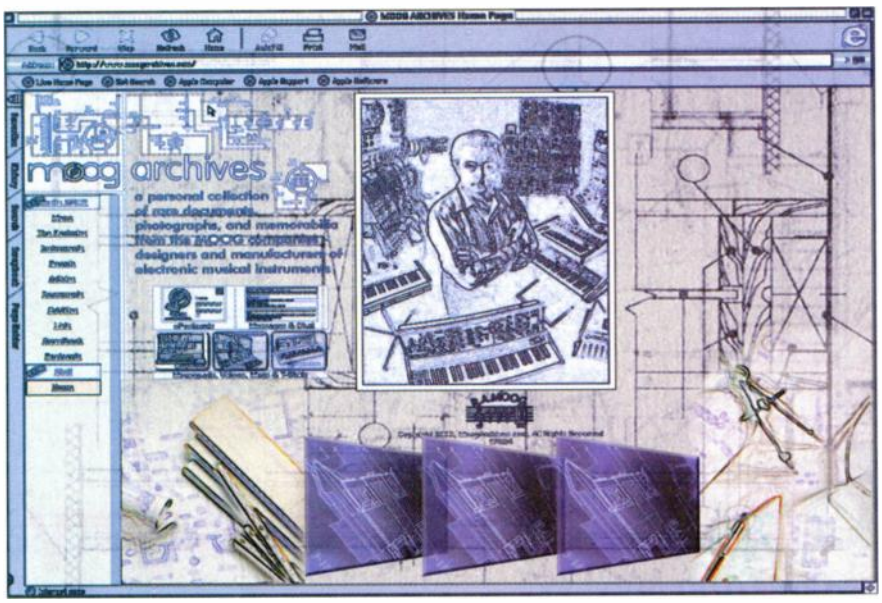
What's in a Web Page?

If you've ever wondered how Web pages work, here's your answer.

By Alan Gary Campbell

As an electronic musician, you may consider the Web to be part of your tool kit. Perhaps you have surfed the Internet countless times, but what exactly is a Web page, anyway? Where did the technology come from, how does it work, and where is it going?

What became the World Wide Web was originally an experiment to find ways for researchers to communicate and share data remotely. It worked. Being creative types who didn't know when to stop, those researchers decided to keep development going and make their creation more interesting than the mostly text-based prototype.



LAURA WILLIAMS

ENTER HTML

The early Web's plain text pages were hardly sexy or feature-laden. A method was needed to control the appearance and layout of a page—font sizes, italics, boldface, headings, paragraphs, and so on—while retaining as much of a text file's simplicity and compactness as possible. The developers decided to utilize a set of embedded, text-based commands called *markup* to control these properties. They created a powerful and flexible standard so that one programmer's software could understand another's markup. That standard was dubbed Standard Generalized Markup Language (SGML).

However, SGML had more power and flexibility than was needed for the Web, and it was difficult to use. So a relatively small portion of SGML's potential was tapped to add simple markup to text, and the result was HyperText Markup Language (HTML). Early versions of HTML offered little more than rudimentary text formatting. But one really powerful feature of SGML was retained in HTML: *hyperlinking*, the ability to jump from one page to another with just a click. Hyperlinking and ease of

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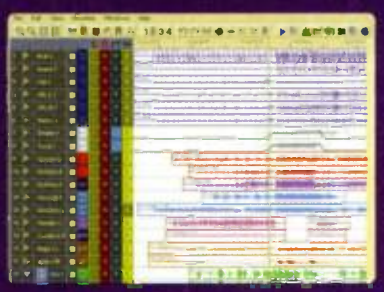

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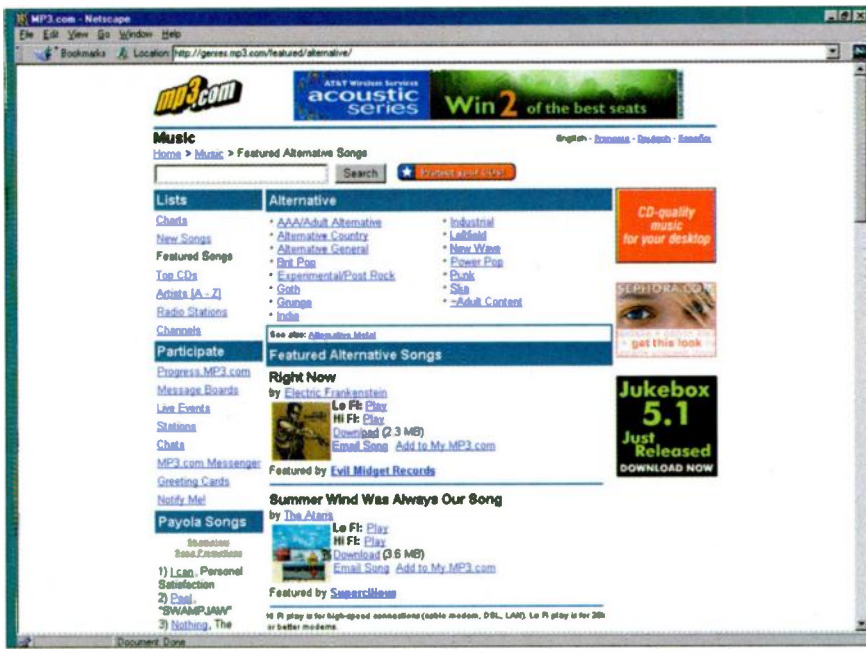

FIG. 1: MP3.com's Featured Alternative page is a content-rich site with several different types of data. It is shown here as displayed by Netscape's *Navigator* 4.

use made HTML very popular very fast.

Of course, the coolest markup in the world is useless if you can't display it. Some of the first software for viewing HTML pages came from—you guessed it—researchers. Browsing HTML-based Web pages with the new software quickly became the norm, so *browser* came to mean the software used to do that. Browser development was soon big business, and features such as font colors and the abilities to display images and to play sounds were added along with new markup to support them.

Unfortunately, features developed faster than standards, and manufacturers frequently disagreed on implementation. Markup that worked well with one browser crashed another. That remains a problem, a trade-off for the rapid development of browser software, which is now so advanced that it resembles a multimedia presentation environment. On the positive side, the most popular browsers, Microsoft's *Internet Explorer* and Netscape's *Navigator*, are distributed for free.

HTML STREET SMARTS

Basic HTML is not rocket science. Its markup consists of simple *tags*, which are really just abbreviations placed between "<>" signs and usually used in pairs. The simplest possible HTML document is:

```
<HTML>Hello</HTML>
```

That's it. The <HTML> tag identifies the start of the HTML document, the text "Hello" is the body of the document, and the </HTML> tag identifies the end of the document. A minimally useful HTML document would probably have at least the following tags:

```
<HTML>
<HEAD>
<TITLE>The No-Frills Test Page</TITLE>
</HEAD>
<BODY BACKGROUND="WHITE"
TEXT="BLACK">
<H1>Welcome</H1>
<P>

Welcome to my no-frills test page.
Coming next week: <I>War and
Peace.</I>

</P>
</BODY>
</HTML>
```

The <HEAD> tags let you delineate information that you don't want to appear in the document body itself—in this case the page title that will display in the browser's title bar. Including a <BODY> tag provides additional options for control over the page presentation. Notice that I've added some instructions—*attributes* in techspeak—inside the <BODY> tag that tell the browser to display a white background with black text. The <H1> tags display the text "Welcome" as a heading, typically in boldface. (There are six heading sizes; H1 is the largest and H6 the smallest.) The <P> tags delineate a paragraph, and the <I> tags cause the title *War and Peace* to be displayed in italics.

You can create any HTML document in a text editor such as *Notepad* or *SimpleText*. Simply save the file as ASCII text with a file name that ends with a .htm or .html extension, then use the browser's File Open command to view your file. HTML tags are not case sensitive, and you can use tabs, blank lines, and extra spaces to make your markup easier to proof and edit. Just make sure the tags have correct syntax and are always nested, not overlapping. (Overlapping tags are considered an error in HTML and might not be interpreted correctly by some browsers.) For example, use <P><I>text</I></P> rather than <P><I>text</P></I>.

INSIDE THE BROWSER

At this point, you have an idea of what basic HTML is and how a browser displays it. But a typical commercial page's markup is much more complex than that of my No-Frills Test Page, and current browsers are pretty complex specimens of software. (See the sidebar "You Can Get There from Here" to learn how a browser finds the pages you request.) For example, **Fig. 1** shows the MP3.com Featured Alternative page (the top part of it, anyway), and **Fig. 2** shows the first 70 lines of the approximately 1,200 markup lines used. (You can reveal any page's HTML using the browser's View Source function, which is usually found under the View menu.)

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As you can see, in addition to the straightforward tags described earlier, there are many others in use, and a Web browser must understand how to display them or at least how to ignore them without problems. Consider something as fundamental as a tag to place an image on a page:

```
<IMG SRC="logo.gif" ALT="Logo
Image" BORDER=0 WIDTH=100
HEIGHT=50>
```

Say this is the first image on the page. The browser attempts to display as much of the HTML as it can before it loads and displays the image. Then it tries to find the image. In the example, the browser searches for the image (logo.gif) in the directory (folder) where it found the HTML file. It checks the file extension and other information to determine the file type. Then the browser decompresses the file (all common Web image formats—GIF, JPEG, and PNG—are compressed to save download time) and determines the color depth and size in pixels. If the color depth is greater than what the computer's video card can handle, it must determine whether to dither or substitute the missing colors.

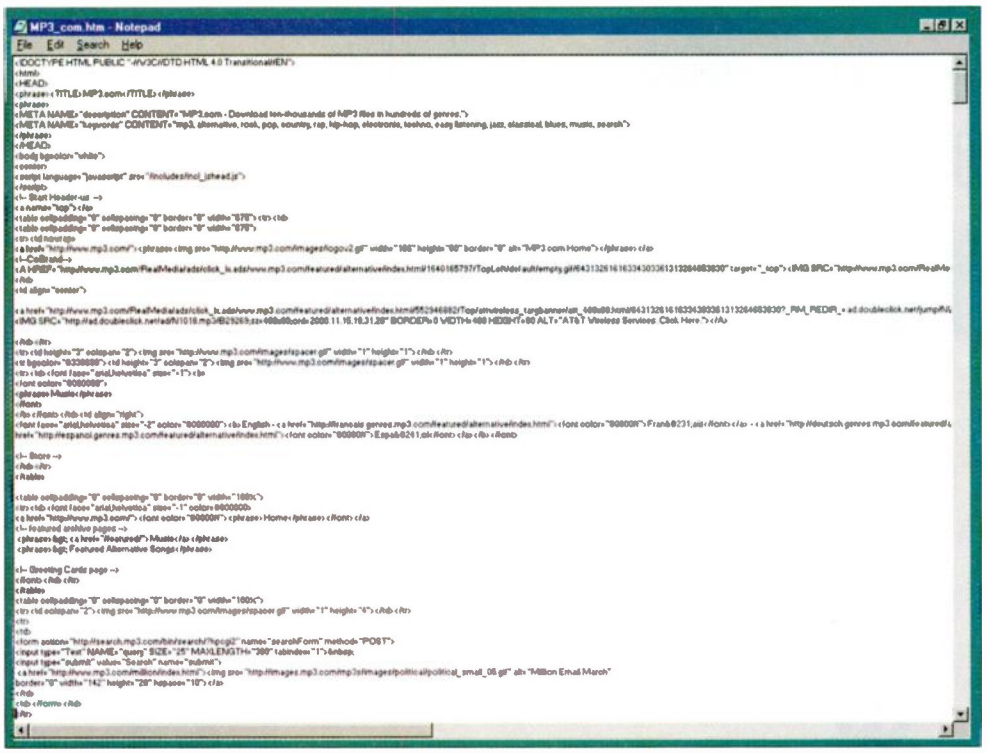


FIG. 2: The markup for the MP3.com page shown in Fig. 1 is extremely detailed; here are just the first 70 lines of the approximately 1,200 lines used to create the page.

If height and width attributes are provided, the Web browser compares them with the actual image size to determine whether it needs to internally resize the image before it's displayed. If those attributes are not

YOU CAN GET THERE FROM HERE

So you painstakingly typed in every character of the mile-long Web address and . . . eureka! You actually found a page; maybe even the one you wanted. Web domains are out there just waiting to send you content. But how does your browser know where to find www.emusician.com?

Computers are good at numbers, so it makes sense for the location of each computer on the World Wide Web (and there are millions of them) to be referenced as a series of numbers, not unlike a postal ZIP code. Those numbers, separated by periods, constitute a computer's Internet Provider (IP) address. For example, www.emusician.com's IP address is 208.242.199.55; if you type that series of numbers and periods in your browser's Location field and hit Return, you'll get **EM**'s home page.

Of course, for humans, remembering www.emusician.com is a lot easier than remembering a string of numbers. Fortunately, there's a hierarchy of computers connected to the Web that do nothing but look up domain names and tell browsers and other Web software where to find them. This hierarchy is called the Domain Name System (DNS). At the top of the DNS are computers controlled by the federal government that tell other computers which one holds the official IP address record for the domain name in question.

When you dial up your provider and then use your browser to request www.emusician.com, the provider checks to see if that address is in its DNS computer. If not, the provider's computer queries others to see if a nearby DNS computer has the information. The provider's computer continues in this manner all the way to the top, querying the computer that holds the official address record. The beauty of this system is that the DNS computers along the way—not just your IP's computer—receive and store the IP address information for some time, which makes subsequent lookups much faster.

provided, the browser attempts to display the image at full size, and it may have to lay out and display the rest of the page again in the process. (That's the reason some pages redraw or stutter while loading—the author neglected to specify the image size attributes.) If the file is damaged or in an unrecognized format, the browser must determine whether to display an error icon, and it may have to lay out and display the page again around that. That's just for one static image.

If the image is a hyperlink, the browser must determine whether to put a border around the image and what color and size the border should be. If the image is the basis for an image map, the browser must also keep track of the mouse position over the image and the link to which each image area points. But those are just the minimum requirements. A Web author may add additional attributes to control layout more precisely or run small programs called *scripts* inside the browser. For example, a script may change the image when the mouse moves over it or display a text message in the browser's status bar.

In addition, the browser must remember not only the image's functions and position, but also the function and position of every element on the page, even while you are clicking on, scrolling, and resizing the browser window. When you consider that a page may encompass not just images and text but forms, audio and MIDI files, and even embedded programs in Web-friendly languages such as Java and JavaScript, the functionality of modern browsers is pretty impressive.

MY BROWSER NEEDS HELP

What if you want to play an MP3 or RealAudio file or view a PDF document or Flash animation, and your browser doesn't understand that format? You'll probably be confronted with a pop-up message that asks if you want to "Pick an Application," "Save File to Disk," or something equally unhelpful. You could upgrade to a newer browser that might support the file type in question, but if you have an older computer or operating system, that may be impractical if not impossible. In that case, you'll have to deal with helper apps (short for applications) and plug-ins.

Configuring your browser to use a helper app really just means telling the browser that when it encounters a certain file type, it should open another program to play or view that file. For example, when you click on a hyperlink to a RealAudio file, *RealPlayer* should pop up and play the file.

Configuring the browser to use a plug-in usually requires downloading that plug-in and going through an installation process. Then, when the browser encounters the relevant file type(s), the plug-in runs inside the browser, often transparently. A Shockwave Flash animation running as part of a Web page serves as a good example. Unfortunately, configuring helper apps and plug-ins can be a notoriously difficult process, even in fairly recent browsers.

WEB OF THE FUTURE

Web development's current focus is on convergence and standardization. It's not unreasonable to expect that the very next generation of browsers will offer unprecedented support for a broad range of tags, media types, and programming functions—just in time for HTML to become obsolete. Extensible Markup Language (XML) and XHTML, an XML/HTML stopgap, are already being deployed to solve some of the problems of separating Web content from appearance. (For more on XML, see "Web Page" in this issue of **EM** and "Tech Page: XML Marks the Spot" in the November 1999 issue of **EM**.)

Do you need to understand these developments right now? Probably not, but the new markup languages will be part of the electronic musician's world in the near future.

Former **EM** *contributing editor* **Alan Gary Campbell** *now writes Web apps for corporate America.*

Getting It Online

You really can make money from your music on the Web.

By Mary Cosola

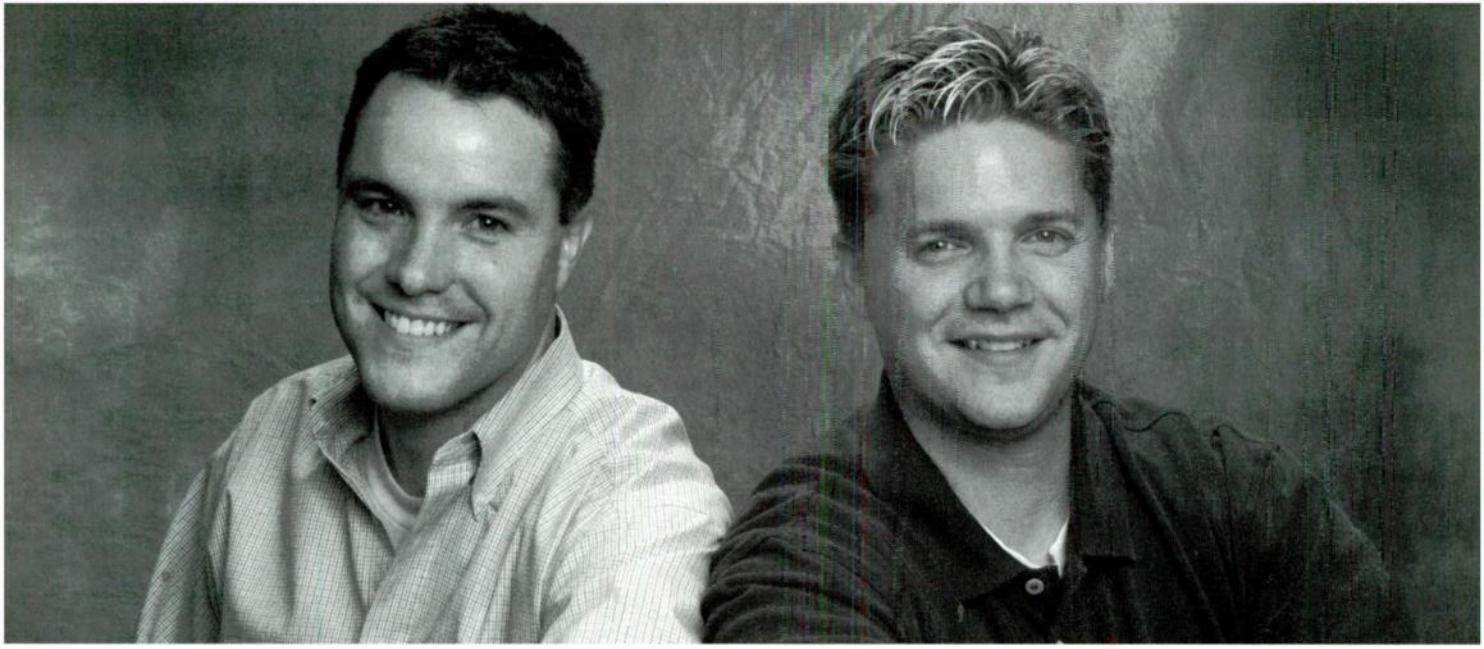
One of the major quandaries in selling music online is finding buyers. Even those musicians who establish decent Web sites and manage to bring in fans have a hard time making any money online from their music. Most musicians resign themselves to the notion that an online presence is just one component—albeit an important one—of a multifaceted approach to marketing music.

The MusicBlitz home page displays featured artists and downloads, as well as menus for finding all the site's music.

But for those musicians interested in selling their songs for use in commercials, soundtracks, computer games, and the like, there is a new online music option that works.

A few bright souls figured out that the Internet offers the perfect alternative to traditional music licensing. Usually, when music is needed for a project—say for a motion picture—the music supervisor must find the appropriate music. One way to do that is to call music publishers, describe the type of song that's needed, wait for the publisher's suggestions, listen to them, decide whether to license any of the suggested tracks, and then fill out the necessary paperwork to pay for the right to use the song. Sometimes the music supervisor knows the song he or she wants but has to track it down and wait for approval to use it—and often pay a very high fee. But new Internet companies are challenging this old way of doing business. These companies make thousands of tracks available for licensing or purchasing through downloads.

Music supervisors, television producers, and game designers are frequenting these sites in ever-increasing numbers as a new way to find music for their projects quickly and easily.



"We Had A #1 Hit Because We Joined TAXI"

If you told me that one day I'd co-write the #1 Country song in America, I probably wouldn't have believed you.

My name is Erik Hickenlooper. My writing partner, Jim Funk and I wrote the Kenny Rogers hit, 'Buy Me A Rose.'

We aren't professional songwriters with a string of hits under our belts. Just a couple of ordinary guys who love to write and record our own songs. We live in small towns in Utah, and we both have day jobs.

But, even though we write Country songs, we've never been to Nashville.

'Buy Me A Rose' was recorded on an 8-track in the back bedroom of an old farm house. We only had one microphone. And every time a cow mooed or a plane flew over, we had to stop the tape. Not very high-tech, but it worked.

Jim and I didn't have any music industry connections, so we joined TAXI. It seemed like the smart way to go. Our instincts proved to be right on the money -- literally.

We landed our first publishing deal through TAXI. That resulted in 'Buy Me A Rose' being cut by Kenny Rogers.

Over the next few months, we watched our song climb the charts until that wonderful week when it hit #1 on all three Country Music charts, including Billboard.

Can TAXI do that for you? Maybe. It depends how good your music is.

TAXI proved to us that if your music is great, they really *can* get it to all the right people.

And TAXI's not just for songwriters. They also work with bands and artists, and can get your music in TV shows and films, too.

But TAXI is much more than a way to connect with the music industry. The written feedback you'll get on your material is like having a team of music industry veterans as your personal coaches.

You'll also get TAXI's great monthly newsletter, and a FREE pass to TAXI's private convention, "The Road Rally." This exclusive convention is phenomenal, and worth far more than what your TAXI membership costs.

So, don't let your music sit on a shelf collecting dust. Call right now for TAXI's FREE info kit. We did, and we got a #1 Hit!

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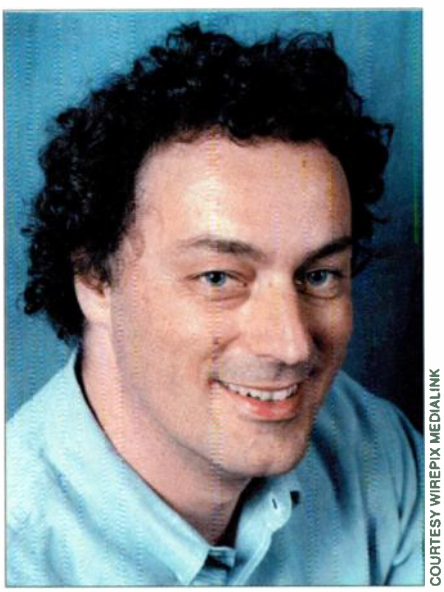
Musicians who want their music represented on these sites must submit songs and await acceptance—or rejection—just as they would when seeking a record deal. The twist is that these sites want to load up their databases with quality tracks, so they evaluate the music on a song-by-song basis. That means musicians aren't faced with selling their bands as whole—including marketability, appearance, and style. They can simply try to sell individual songs. The benefits for all involved are clear: musicians can sell songs whether they are signed to labels, and industry pros can easily search through prescreened, quality tracks. LicenseMusic, iNoiz, and MusicBlitz offer similar services to their clients, but each company operates differently and has different agreements with its musicians.

LICENSE TO SELL

Having debuted in early 1997, San Francisco-based LicenseMusic (www.licensemusic.com) is the oldest established company in the bunch. The idea behind the company was to make already cleared music from all over the world available for commercial projects. "We wanted to use the Internet as a market maker between the buyer and seller of music," says Gerd Leonhard, CEO of LicenseMusic. "There is so much good music that never sees the light of day. We thought it was a good idea to create a marketplace where, say, an Australian producer of a Shockwave animation could go to our Web site and find a cool acid jazz track by a Swedish guitar player."

Customers looking for music to use in a project can search the site in a variety of ways. They can search by genre, or they can perform advanced searches by specifying details such as the desired musical genre, instruments, tempo, setting (for example, action, romance, after-school special, vaudeville), and price range. Tracks that match the given criteria are listed on the screen, and the customer can preview the tracks and download MP3s to audition them as temp tracks. After deciding which tracks to license, the customer clicks on the Calculate License Agreement button and is prompted to choose options for all the licensing possibilities.

"The licensing engine is the nuts and bolts of what we do," Leonhard says. "Traditionally, it can take six weeks to do a song clearance because there are so many variables. Now, we've constructed this engine that computes about 250,000 different licensing deals. Let's say you're doing a motion picture. You can choose worldwide rights, perpetuity, opening credits, all the variables films have, and it spits out a license right on the spot. No lawyers or anything else."



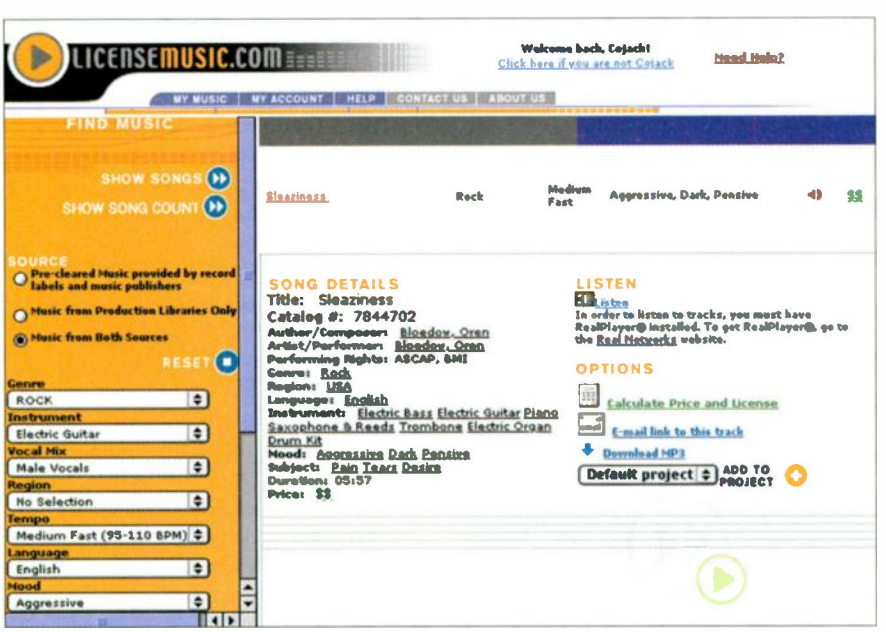
COURTESY WIREPIX MEDIALINK

Gerd Leonhard is the founder and CEO of LicenseMusic.

Leonhard says it took about a year just to do the research on the licensing engine. "It's really the biggest asset of the site," he says. "All of the legwork necessary for a traditional licensing deal—such as calling the publisher, label, and so on—isn't worth it for a $500 deal. The average transaction time is about 12 minutes for a customer to license a song from us."

The company works with unsigned and signed artists, record labels, and music publishers. The contract is very basic: the artist gives LicenseMusic the right to represent him or her in licensing a particular song. The company gets a percentage of each licensing deal. "The artist, label, or publisher keeps all the rights," Leonhard says. "They are just telling us that we can do a deal on their behalf."

The easiest way to submit songs is to



A search on the LicenseMusic site for an aggressive, mid- to fast-tempo rock track for a motion picture soundtrack brought up a few different options, including a track called "Sleaziness." Track information is displayed along the left side of the search result screen, and options such as listening, downloading an MP3, or calculating the license fee are on the right.

Loopfactory

THE SERIES

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The SU200 lets you sample the world and distort it into entirely new forms of music. It's the only sampler in its class that lets you sync and play back loops sampled at different BPMs so you can merge any sounds. Plus...

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- Loop Remix Feature: instantly and in real-time create dozens of variations of any sampled loop
- Resampling: mix together multiple samples with effects into a single new sample

DX200 FM DESKTOP SYNTH

The tonal range of FM is remarkable, from punchy and phat to subtle and expressive. And the sounds you can coax out of this technology with the Yamaha DX200 desktop synth are unbelievable.

- Intuitive interface for real-time control of voice edits make FM programming a breeze
- 16-Step old-skool sequencer with 1 DX and 3 rhythm tracks
- 4-Track "Free EG" to record and playback knob movements
- Comes bundled with voice editing software

AN200 AN DESKTOP SYNTH

The phunky, wild sound of analog synthesis is the domain of the Yamaha AN200. This synth not only delivers the stylized voices, it's got the perfect interface for forming them into your own special vibe.

- Smooth, phat and filthy fully-resonant filters
- Real-time control of voice edits and step sequences
- 4-Track "Free EG" to record and playback knob movements

THEY RUN LOOPS AROUND THE COMPETITION

The Yamaha SU200, AN200 and DX200 – the Loop Factory series – are the best tools for creating dance music loops, bar none. Each has its own individual personality and combined together they are an unstoppable creative force. Check them out at a Yamaha dealer near you and see how easy it is to create the freshest grooves around.

YAMAHA

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which site visitors use a secure server for credit card purchases. The customer can get the purchased music by download, e-mail, or FTP. A fourth option, which is usually best if a lot of content is purchased, is to have a CD burned and sent overnight. Because the music is bought and not licensed, there is no need for an elaborate license agreement.

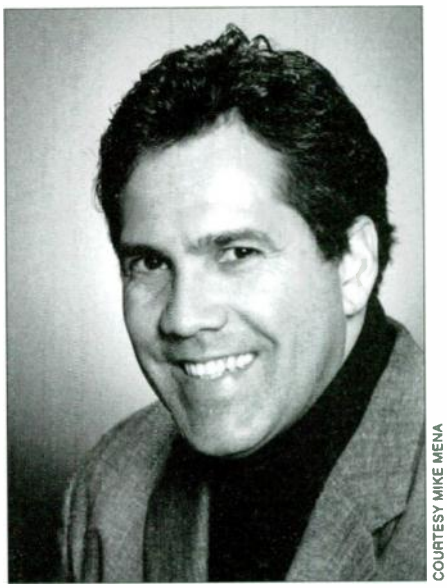
"It's a buy-out deal. We don't purchase the music, but we sell it on our site," Frederick says. "Customers can't recycle it and say it's their own. They can, however, use it in another production. For example, say you're a game developer and you're using the music you've bought from us for the main theme on a game. You then want to develop several mission packs for that game and reuse the music—you can do that."

Musicians who want to get their work on iNoiz must submit a demo through the mail, e-mail, or FTP. The company gets a lot of submissions based on referrals and word of mouth, but Frederick says that they are open to unsolicited music as well. "We'll take anything from any channel because some of the best musicians are people working out of their home studios," he says. "That's what we want to focus on—untapped sources."

Once a song is evaluated and accepted, a percentage deal is negotiated based on the amount of music the artist provides and the music's production level. For example, a MIDI track will be priced differently than a fully orchestrated piece recorded with a large session orchestra. The more a song costs to produce, the more iNoiz pays the artist for it and therefore the more the company ends up charging the customer.

The company accepts music from unsigned and signed artists alike, but the basic deal remains the same. "It is the artists' responsibility to see that their percentage gets split up based on what they have worked out with their label and their publisher," Frederick says. "It keeps it simple for us, the artist, and the end user. That's the whole idea."

Once the deal is set, the artist provides the company with the song and all the necessary database information for indexing the song on the site. The song is then reviewed by the quality control department, encoded, entered into the database, and uploaded to the site. The artist is paid only when the track is sold, and the company mails out payments quarterly.



COURTESY MIKE MENA

Mike Mena is the vice president of artist development for MusicBlitz.

Frederick says that his brother's and his experiences as musicians and in film and television production were the inspiration for establishing iNoiz. "We have often been in situations where we're in an editing suite working on a project, and the client is looking over our shoulders, tapping his watch, wondering why it is taking so long to go through CD after CD to find the right piece of music," Frederick says. "Or we used catalogs, where you have to look through, find the description, order the music, and wait for it to come. Or you try to license a song, and the licensing process can be complicated and time consuming. When you are working on a project with a tight budget, time is very important. The last thing you want to have to do is to wade through a lot of garbage. You want to be creative without having to deal with that stuff, so we wanted to promote the creative process by taking the complications out of it."

SELLER BEWARE

Some artists might not consider that when selling songs for use in television, movies, and advertising, they might have little say in where the songs are placed. That could be a problem if the song ends up as the backdrop to something objectionable.

Mike Mena of MusicBlitz says that because the company places the songs individually, that is not a concern for the company's artists. "The artist has the final say in everything," Mena says. "We take it on a case-by-case basis because you never know what an artist might be opposed to. We want everyone to be comfortable with the deal and where their music ends up."

LicenseMusic has a policy against licensing music for use in political campaigns, alcohol and tobacco advertising, and pornography. "The only leap artists take is that by giving us the right to license their song, we might sell it to an ad agency that does an ad for something they don't like," Gerd Leonhard says. "You don't have control over each single transaction."

Jamie Frederick of iNoiz says that the company has considered how to handle the situation of objectionable song usage, but it has yet to institute any policies. "Like any physical music library that can be bought, there is no control over how the music is used," Frederick says. "We felt that it would complicate things, and we just don't have the resources to police such enforcement. We do, however, have the right to deny a composer's piece of music because of bad taste, such as racist or overly sexual lyrics."

PUTTIN' ON THE MUSICBLITZ

Los Angeles-based MusicBlitz (www.musicblitz.com) operates much more like a traditional record label than iNoiz and LicenseMusic do. It scouts talent, signs artists, produces albums, and sells CDs through record stores. The company differs from regular record labels in that it also signs artists to single-song deals, usually for the purpose of shopping the song to music supervisors and the like.

"Our primary intent was to record some great new music from artists and to license it to television and film, and to explore other avenues to provide the artists with an alternative to a record deal," says Mike Mena, vice president of artist development for MusicBlitz. "Also, we have a great opportunity to work with artists who don't fit into a major label's plans right now, but they can certainly work with us. I think we've found the stuff that could possibly fall between the cracks. For instance, we're currently working with Taj Mahal. The man is a legend—he's won Grammys, and he doesn't have a record deal. He is somebody who is very appealing for us to work with."

The site is set up like a regular music site, with downloads from MusicBlitz artists offered side by side with free downloads of other artists, such as Eminem. MusicBlitz also offers independent reviews of all the site's music. The idea is to bring in traffic from music fans and industry insiders alike. "We're a lot more selective than other music Web sites," Mena says. "Our goal is not to have the most music on the Net; it's to have the best music. We consider one of our functions for our users is to filter out the large quantities of music they'd have to weed through on other sites."

In addition to encouraging industry professionals to check out the music available on its site, the company also networks with performance rights societies and film and TV professionals. "One of the advantages of working with us is that we can turn things around quickly," says Mena. "If a big record label or publisher is closing a multimillion dollar deal, a licensing deal might have to wait a week or two, which is time a music supervisor might not have." The Internet offers the access speed that the traditional method of mailing a CD can't compete with. "If someone needs a song quickly for a TV project, downloading a version of the song to check out is faster and cheaper than burning a disc and spending $60 in FedEx charges only to decide it's not the right track for the project," Mena says.

As for musicians hoping to get on the MusicBlitz site, most of the demos Mena receives come to him as referrals from colleagues, lawyers, or managers he knows and trusts. However, he is open to submissions from the Web site or through the mail. "No matter how we get it, the music has to have some kind of sex appeal," Mena says. "A cassette with a loose-leaf piece of paper rubber-banded around it is not as enticing as a demo that comes with some great reviews." Artists who are signed to single-song contracts still get the full record company treatment. Whereas LicenseMusic and iNoiz contract for already produced tracks, MusicBlitz follows the traditional record company path of taking artists through the A&R process and putting them in the studio.

Although MusicBlitz operates its own record label, it works with signed and unsigned artists for the single-song deals. "For artists not on our label or those with another label, we just own that one song," Mena says. "For the most part, publishers are pretty agreeable to working with us because it's just another form of promotion for them as well."

WRAP IT UP—I'LL TAKE IT

For most musicians, selling songs online for use in television and other projects won't be a major revenue generator, but it does offer a way to get their songs heard and to make money. For all the clichés about the Internet "leveling the playing field" for musicians, it seems that online music licensing is finally doing just that.

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

The Earl of Whirl

Capturing the motion and emotion of the Leslie speaker cabinet.

By Brian Knave

Just as Helen of Troy's beauty launched a thousand ships, the sound of the Leslie "tone" cabinet has launched a thousand emulations. But even the best of them—and there are some good ones—pale in comparison to the real thing. What makes the Leslie sound so compelling? In a word, motion. The sound from a Leslie cabinet moves not only outward but also in a circular direction around a pivot—sort of like the sparks from a fireworks pinwheel. As the sound spins out at varying speeds from the Leslie's rotating components, the listener hears a complex blend of Doppler effects (including frequency and amplitude modulation), phase shifting, frequency masking, and ambient reverberation. That highly animated sound is further colored by the Leslie's tube-driven amplifier, which often is driven to distortion.

Can't fit a Hammond B-3 into your home studio? That didn't stop the author, who recorded organist Dan Eisenberg by putting the Leslie in the studio (see photo on p. 126) and the B-3 in the backyard.

BRIAN KNAVE

It's no wonder most musicians and producers prefer recording the real thing to using an emulation. But is it really possible to accurately capture the magical sonic phenomenon you encounter in a nightclub while listening to the likes of a Jimmy Smith, Joey DeFrancesco, or Larry Goldings (to name just a few masters of the classic Hammond B-3/Leslie combination)? No, not really. No matter how well you record a Leslie, the experience of being in the room with the Earl of Whirl is simply not going to translate fully through a pair of stationary speakers. But take heart: many of the elements that make up the Leslie sound can be documented faithfully in the recording studio.

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If you're into Drum 'n Bass, Jamie Myerson needs no introduction. America's premiere D&B producer has finally made a sample CD, and it slams! Construction kits & loops, from hard two-step to dreamy Lookin' Good-style beats. There's other Drum 'n Bass CDs out there, but how often do you get someone of Jamie's caliber making one?

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Underground Garage

New from one of the *Millennium Garage* team, *Underground Garage* is a "must buy" for anyone who's serious about garage! 30 construction kits filled with the toughest beats, bass loops, keyboards, single notes & multisamples plus exclusive sounds from Ray Hurley & DJ Para. Even if think your House/Garage collection is full, pick this puppy up!

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From the soaring melodies of Goa, to the pounding machine rhythms of Berlin, *Megaton Trance Bomb* delivers everything you need to create trance. This ever-changing music requires up-to-the-minute sounds, & we've got them right here! Construction kits, drumloops, synth arpeggios for that custom sound: everything you need to bliss out right now!

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BRICK HOUSE

Of the many Leslie models that were built, the most popular are the 145, 147, and 122, which I will focus on. Those models share several features, including a three-compartment cabinet design. The top compartment houses a rotating treble horn with identical-looking bells that face opposite directions from each other. Note that only one of the bells channels sound; the other is a dummy that acts as a counterbalance so the assembly spins smoothly without wobbling. Also worth noting is that the horn assembly is mounted off-center in the cabinet—an important detail in some miking setups.

The cabinet's closed middle compartment contains the high- and low-frequency drivers and 800 Hz passive crossover. The treble unit is a ¾-inch Jensen compression driver (the kind used in P.A. speakers) connected to a vertical tube that feeds into the rotating horn assembly. Bass frequencies are handled by a 15-inch speaker that fires downward into a rotating wooden drum located in the cabinet's lower compartment (see **Fig. 1**). The lower compartment also houses the Leslie's 40-watt monophonic tube amplifier.

The treble horn and wooden drum are turned by rotors driven by two dual-speed motors, letting the player choose between slow and fast rotation speeds—for chorus and tremolo, respectively—independently for each element. Interestingly, the wooden drum turns clockwise and the treble horn counterclockwise. In addition, the two units' rotation speeds, though roughly the same at slow and fast settings, differ during ramp-up and ramp-down times because of the wooden drum's greater mass—yet another factor that can deepen the complexity of the sonic stew.

These and other construction details can be critical to the recording engineer because Leslie owners tend to modify their cabinets in various ways. Organ players who kick bass, for example, often don't want chorus or tremolo on the low notes, so they might disable the motor that turns the wooden drum. Other players, for whatever reason, sometimes disable the top motor. Obviously, in either case, the recording engineer who is unaware of the modification is at a disadvantage. If the top motor is disabled, for example, the engineer needs to manually align the sound-producing horn with the mic.

BRIAN KNAVE

Two AKG C 460 microphones capture a stereo image of Eisenberg's Leslie model 122 while an AKG C 414 B/ULS brings in the lows.

Other Leslie mods include mechanically linking the top and bottom rotors so that they turn at the same speed (this gives a more dramatic Leslie sound); switching out the power amp for a more powerful one or disabling it and using an external combo preamp; using different treble and bass drivers (typically because the stock ones were blown); installing a different crossover; or even bi-amping the treble and bass drivers. Although none of these modifications should influence the recording as much as disabling a motor, any will alter the sound and therefore may also affect how you record the Leslie.

You may encounter numerous other original Leslie models, including the older models 45, 47, and 22 (which are identical to the 145, 147, and 122 except that they have single-speed rather than dual-speed rotors). Other original Leslie models, such as the 51, do not contain amplifiers and must therefore be connected to an external amp. In addition, since CBS bought the Leslie company in 1965, several new models have been produced, including the 122A, 122XB, and 147A, as well as larger models made for multichannel organs. It is important when recording new or unusual Leslies to survey the construction carefully in case the unit employs a design change (a side-firing woofer, for example) that would affect sound production.

TIGHT SHIP

Like a neglected drum set that squeaks and rattles each time it is struck, a rickety old Leslie can be a Pandora's box of extraneous noise. Therefore, in addition to checking for any modifications, another important step is to listen carefully to the cabinet while the musician plays at full recording volume. (If the instrument is an organ, make sure the player uses the same drawbar settings that will be used during recording.) If you hear creaking, buzzing, or other unwanted noise—I specify "unwanted" because some of the resonances and distortions produced by a Leslie cabinet may be desirable—have the musician play chromatic scales slowly from lowest note to highest so you can pinpoint the note or notes that set off the noise. If you don't hear the offending sound on single notes, try chords.

After determining which note or group of notes cause the noise, locate where the unwanted sound is coming from. The last Leslie I recorded had an obnoxious sympathetic buzz that I traced to some loose wood plies on the bottom-most panel. After discovering that I could stop the buzz with my

hand, I put a piece of foam rubber on the spot and weighted it down with a 10-pound dumbbell. That didn't completely stop the buzz, but it did damp it sufficiently for the recording.

Other common sources of noise are the Leslie's motors, rotors, belts, and pulleys—any parts that move, basically. Hopefully, the player maintained those mechanical parts well. If not, you need to track down the noises and squelch them or move the microphones back from the source to minimize pickup of the unwanted sound.

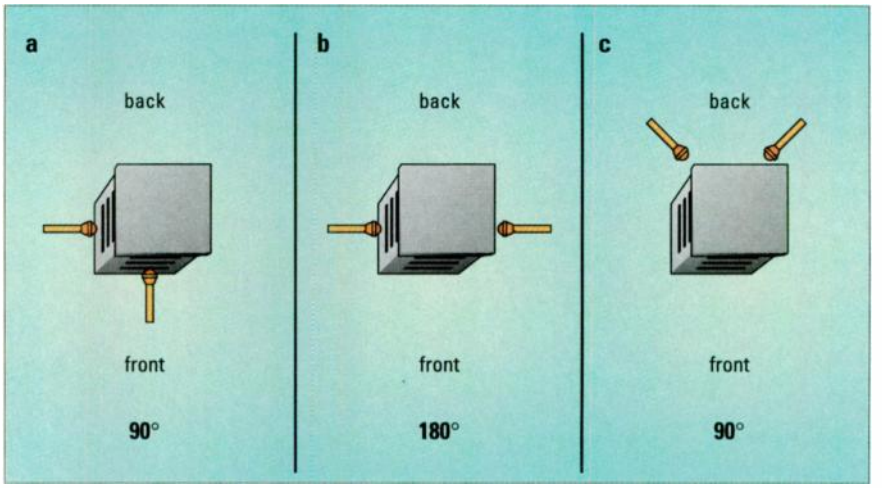

FIG. 2: These diagrams illustrate three different ways to position a pair of mics to capture a stereo image of the Leslie's rotating treble horn.

FREE REIGN

In addition to its historical role as bedfellow to the B-3 and C-3 organs, the Leslie cabinet also gets coupled with other instruments such as guitar, vocals, harmonica, other organs and keyboards, or anything that strikes the player's or producer's fancy. Likewise, there are 101 ways to record a Leslie, depending on the application and your creative bent. Variables include the usual suspects: number of mics (and tracks), mic selection, mic placement, and ambient sound.

As for mic selection, Leslies don't put out much high-frequency content above 12 kHz, so you can obtain good results with almost any decent microphone, whether dynamic or condenser. Indeed, many classic B-3/Leslie tracks have been recorded using just a couple of Shure SM 57s up top with perhaps a Sennheiser MD 421 positioned at the bottom of the cabinet to capture the low end.

Some positioning issues come up no matter how many mics you use. A primary question is whether to aim the mic(s) directly at the source(s) through the Leslie cabinet's open back (with the upper and lower panels removed) or through the louvers on the cabinet's front or sides. Miking from the open back gives a slightly brighter, more open sound but is prone to capturing more extraneous noise, including wind from the spinning horn or drum and mechanical noise from the rotors. For that reason pop filters or foam windscreens may be advisable when miking from the back, especially when using condenser mics. Miking through the louvers provides a mellower sound and cuts down significantly on wind noise from the rotating elements. As usual, the best approach is to try several miking arrangements, compare the results, and choose the setup that sounds best for the song.

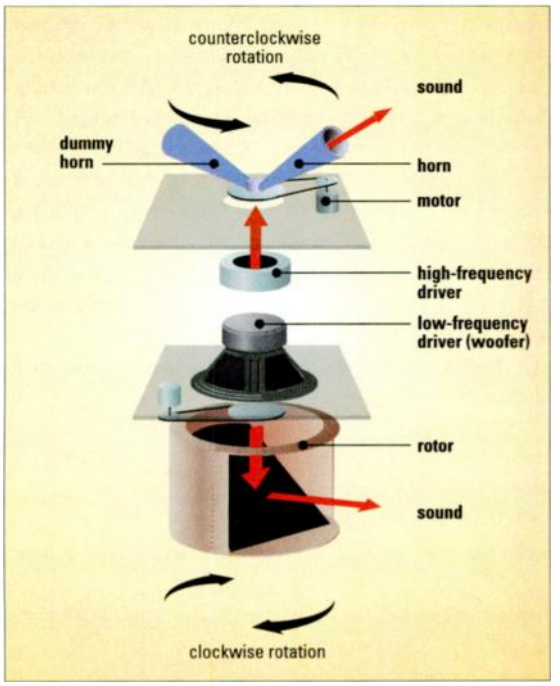

FIG. 1: This conceptual diagram shows the basic inner workings of a Leslie. Note that one of the treble horns produces no sound.

ONE FOR THE MONEY

For a dense mix with scads of instruments, a mono Leslie track may be sufficient. Fortunately, Leslies usually sound so great that even a strategically placed single mic can capture a killer sound.

The trick here is to record in a great-sounding, reverberant room with the mic positioned far enough away from the cabinet to capture a natural sound. Simply move around the space and use your ears to determine where to put the mic while bearing in mind that you are listening for a blend of ambient and direct sound. A small tiled bathroom makes an excellent Leslie chamber, especially for capturing a huge sound on a single track. In that case, I would probably use a large-diaphragm condenser set to the omnidirectional polar pattern or a single-point omni such as the Earthworks QTC1.

TWO FOR THE THROW

There are two approaches to miking a Leslie with two mics: put one mic on top to capture the treble and a second one on the bottom to record the bass, or use a matching pair positioned to capture a stereo image of the rotating

Perfect Picture

98/2000/ME

E WDM
Enhanced Windows Driver Model

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EGO SYS has concentrated energies on developing new driver for the new Windows OS format, such as Windows 2000, Windows ME, and even for Windows 2001(a.k.a. Whistler). Unlike non-WDM Windows 2000 compatible drivers, this newly developed driver using **E-WDM** technology from EGO SYS would work for all of digital audio drivers like MME, DirectSound, ASIO, and GSIF. Until EGO SYS succeeded to develop **E-WDM** drivers, other so-claimed Windows 2000 compatible drivers were not perfect solution for the new OS formats. This makes **E-WDM** driver very unique and special. When there is a new OS released, 3rd party developers have to develop the new driver following the new standard and format. However, no one ever succeeded to achieve new digital audio drivers in WDM especially in multi-channel mode.
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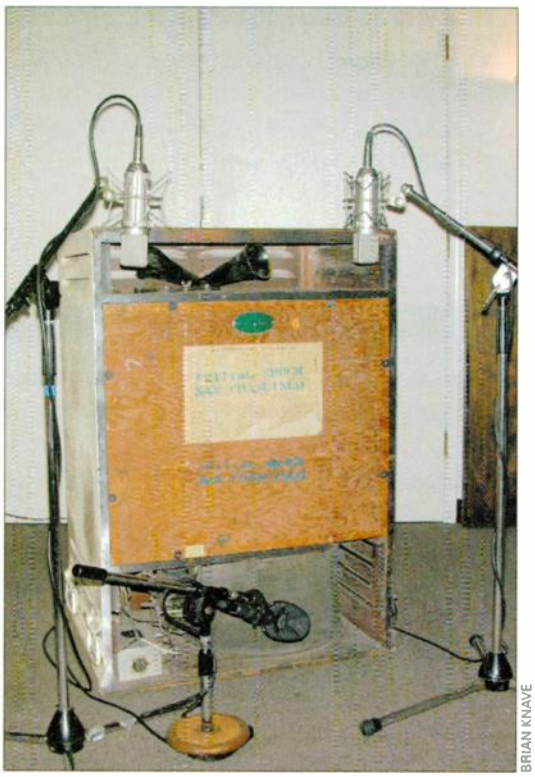
treble horn. Obviously, the first is the better option if you have a B-3 player kicking bass (assuming no other bass instrument is in the mix). In that case, a mic with a good low-end response—for example, a ribbon microphone or a large-diaphragm mic—is the best choice for the bottom. Try panning the bass track dead center and the treble track to one side or the other, say, at ten or two o'clock.

A number of setups will work for the matched mic pair on the rotating treble horn. If miking through the louvers, try positioning the mics on two sides of the cabinet, at either a 90- or a 180-degree angle to each other (see **Figs. 2a** and **2b**) and at an equal distance from the cabinet (typically four to eight inches away). For stereo miking from the back of the cabinet (with the top panel removed, of course), position the mics at a 90-degree angle to each other with both aimed at the treble-horn assembly (see **Figs. 2c** and **3**). Remember to take into account the off-center mounting of the horns.

Panning the tracks hard left and right will create the fullest sound and greatest sense of movement between the stereo mic signals. However, such extreme panning may sound unnatural, especially if you try to create a soundstage with a discernible location for each instrument. Pan the two tracks closer together for a more natural sound.



BRIAN KNAVE

FIG. 4: A matched pair of Neumann U 87s captures a stereo image of the rotating treble-horn assembly while a Coles 4038 ribbon mic documents the lows from a Leslie 51.

THREE'S COMPANY

Although two is the minimum number of microphones required to capture some "motion" from a Leslie, at least three mics are required to capture both treble-horn motion and the unit's full frequency range. When using three microphones, position the matched pair as described earlier and train the third mic on the rotating wooden drum that channels sound from the 15-inch woofer (see **Fig. 4**). That drum can put out a fair amount of wind, especially when spinning at full speed, so if miking from the back of the cabinet, you may need to employ a pop filter or

BRIAN KNAVE

FIG. 3: When using the stereo-miking setup illustrated in Fig. 2c, angle the mics as shown so the live horn will be on axis with each mic during its rotation. Note, too, how the mics are positioned to accommodate the off-center mounting of the treble-horn assembly inside the cabinet.

windscreen. (Note that newer Leslies use a Styrofoam drum, which is lighter and produces less wind than the wooden drums.) If the player disabled the lower motor, a pop filter won't be necessary; however, make sure to turn the drum so that the port is facing the mic.

Again, for the fullest sound and most motion, pan the stereo tracks hard left and right and put the bass track dead center. To locate the organ (or whatever) on a more believable soundstage, group the three tracks closer together—only two or three "hours" apart—again with the bass track in the center and the stereo tracks on either side.

FAB FOUR, FAB FIVE

To create the greatest sense of motion, add a second bass mic—preferably the same model as the first—and record onto four tracks. As with the stereo pair up top, position the two lower mics at a 90- or 180-degree angle to each other. (The configuration of the bass mics to the treble mics—whether on the same, opposite, or adjacent sides of the Leslie—is not critical, but you may want to experiment just the same to find the combination you like best for the song you're recording.)

Afterward, play with the panning to get the desired movement. You could pan each stereo pair hard left and right, of course, but you might also try panning the bass tracks inside the hard-panned treble tracks, perhaps at ten and two o'clock. Or try panning them the other way around, with the bass tracks hard left and right and the treble tracks inside. Still another approach would be to overlap the tracks, with the treble pair panned hard right and ten o'clock and the bass pair panned hard left and two o'clock, for example.

For the most versatile and potentially realistic sound, add a room mic to the mix for a total of five microphones. That mic—a tube condenser would be a nice pick—should be positioned several feet from the Leslie, if not on the other side of the room. Heck, if you want to go all out—say, if you're recording a solo B-3 record—add an XY condenser pair to capture the ambient sound in stereo for a total of six mics and six tracks. If you think that sounds like a lot of mics to keep track of, think of the options you'll face when a B-3 player carts in *two* Leslies—the way a Hammond was meant to be played, according to the connoisseurs.

SPIN CYCLE

As you can see, there are any number of ways to record and mix a Leslie. This article is hardly exhaustive, but it has outlined some basic tried-and-true techniques and hopefully also provided some helpful tips for engineers new to recording the wondrous Leslie speaker cabinet.

Associate editor **Brian Knave** *looks forward to recording and mixing a Leslie for 5.1 surround sound.* *Thanks to Myles Boisen, Barry Cleveland, Dan Eisenberg, Jim Pugh, and George Petersen.*

REVIEWS

YAMAHA

AW4416

Features, flexibility, and quality combine in a competitive DAW.

By Mark Nelson

Yamaha's impressive AW4416 Professional Audio Workstation takes portable digital studios to a new level. With features worthy of a professional recording studio—including an extraordinarily flexible 24-channel digital mixer and a 16-track hard disk recorder that can record on all 16 tracks simultaneously—it offers quality and control that are tough to beat for the price.

The AW4416 records 16- or 24-bit uncompressed audio at 44.1 or 48 kHz. You get stereo digital input and output, built-in SCSI for backup, dynamic and scene automation, motorized faders, and a bevy of handy sampling pads. There's plenty of processing power, too, including two onboard multi-effects processors and 4-band parametric EQ and dynamics processing on each channel. If you fill the slot in the front of the unit with an optional CD-RW drive ($200), you can take your music from inception to CD without leaving your desk.

FIG. 1: The AW4416's buttons and switches are grouped according to use, with utility and mixer functions on the left and recording functions on the right. Eight sample pads are directly below the central display. A second display for metering and counter functions appears in the upper right corner.

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The machine does have its limitations, especially with regard to editing, but it is impressive nevertheless.

AT FIRST GLANCE

Right away, I was impressed with the AW4416's appearance (see **Fig. 1**). Banks of buttons and knobs spread across the surface of an attractive silver wedge. A soft rubberized mat surrounds the faders and transport controls. Labeling is clear and easy on the eyes. Everything feels solid, too; the buttons offer a reassuring resistance, switches emit a satisfying click, and the knobs appear robust. This machine came to play.

I like to dig right in before cracking a book, so it was fortunate that Yamaha taped a note in a conspicuous place, warning of disaster if I powered up before installing the hard drive. Up to 64 gigabytes can be installed on easily swappable 2.5-inch IDE drives. Installing the drive and the optional CD-RW burner was not too much of a challenge, but give the job to a tech if you're easily intimidated.

AROUND AND AGAIN

Inputs 1 and 2 on the rear-panel jackfield sport both XLR and TRS connectors with insert points (see **Fig. 2**). The remaining inputs just have TRS jacks, but their levels range from –46 to +4 dB, still good enough to function as mic preamps. (Yamaha used TRS jacks to save space.) This arrangement works if you're wiring up a room with a patch bay and the necessary cables, but I'd like the option of plugging all my mic cables directly in to the AW4416. In addition, the lack of full-featured microphone inputs meant that when doing multitrack remote recording, I had to use an auxiliary mixer to record a rhythm section, which was kind of a blow to the all-in-one philosophy.

All channels feature dedicated trim pots and peak LEDs on the top panel. A rear-mounted switch handles phantom power for inputs 1 and 2, but I wish there was a status LED.

Input-channel 8 has a second unbalanced ¼-inch jack for electric guitars, basses, and other electric instruments. Naturally, I checked this out right away. I plugged in my trusty Gibson ES-150 and inserted an amp-simulation effect from one of the onboard processors. The quick-and-dirty crunch sounded pretty good, though it wouldn't fool a purist.

FIG. 2: The AW4416 offers a pretty good package of inputs and outputs, including full-featured mic inputs with phantom power and inserts and six TRS mic/line inputs. Channel 8 has an additional high-impedance input. Any combination of inputs and outputs may be routed to the S/PDIF digital I/O.

Directly below the inputs are a pair of RCA tape-out jacks, four unbalanced ¼-inch Omni assignable outputs, and a pair of +4 dBu balanced TRS monitor outs. Why does Yamaha use both balanced and unbalanced connections? Is the unit intended for consumers or pros? The answer, of course, is that it's suitable for both types of users.

▼

I was surprised to learn that the automation didn't extend to the inputs.

Rounding out the rear panel are two option slots, a stereo headphone jack, S/PDIF in and out, word clock in and out, jacks for a footswitch and a 9-pin mouse, a serial connection for a pre-USB Mac or PC, and a SCSI port. There's also a trio of MIDI jacks.

A glance at the front panel reveals the AW4416's heritage. If you're familiar with Yamaha's digital mixers, particularly the 02R, you should feel right at home.

Measuring approximately 3.5 by 4.5 inches, the main display handles anything from waveform displays to information about routing to silly little graphics. Kudos to Yamaha for getting so much information into such a small space. There are three ways of accessing different pages: pressing a selection button repeatedly, pressing function keys immediately below the main display, or using the optional mouse. A second display is dedicated to showing levels for the recorder tracks and main outputs, counter location, clock source, and other useful details.

The AW4416 has more than a hundred buttons and knobs, including four knobs dedicated to hands-on control of pan and parametric EQ. Some buttons control seemingly unrelated functions; for example, the EQ button also accesses pages for fader and mute groups. Likewise, the Dynamics screens contain pages for channel polarity and track offset. It may take a while to get used to this sort of multilevel functionality.

The 60 mm motorized faders serve a variety of functions, operating inputs 1 to 24, aux sends or returns, and returns from the recorder. Change a scene or mode, and the faders instantly jump into position, eliminating any doubt about their settings. Watching while they magically retrace a complex mix is downright hypnotic.

FLEXIBLE ROUTING

Flexible signal routing is the AW4416's key strength. Mixer input choices include those you'd expect—analog ins, S/PDIF in, the sampling pads, the option slots—and one you might not: an internal metronome (see **Fig. 3**).

The multiplicity of output routing options conjures up some interesting scenarios. You can use the option slots to transfer all 16 tracks to a pair of ADATs, which lets you archive your tracks to inexpensive S-VHS tape and to send them to another studio, for example. You can route an aux send through S/PDIF to and from an outboard effects

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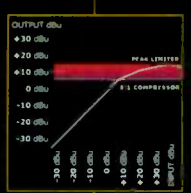

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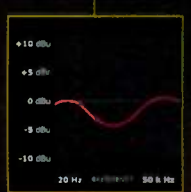

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TAKING IT ALL IN

Thanks to the default busing assignments, basic recording tasks are relatively painless. These defaults are all many users will ever need, but almost any kind of busing and routing option is easy to set up and store for later recall (see **Fig. 4**).

The transport section features friendly tape-style controls. A matrix of buttons manages loops, auto punch, and up to 99 locate points per song. Auto punch is superbly easy to use: I was able to set punch-in and punch-out points, then roll back and fix a mistake in less time than it took to write this sentence.

With the ability to simultaneously record on all 16 tracks, the AW4416 is a serious contender in the desktop-studio world. Quick Record makes recording

AW4416 Specifications

MIXER	
Channels	16
Faders	16 channel; 1 stereo master (60 mn, motorized)
Analog Inputs	(2) XLR balanced mic (Ch. 1 and 2); (16) balanced TRS ¼" mic/line (Ch. 1–16); (1) unbalanced ¼" hi-Z (Ch. 8)
Analog Outputs	(2) RCA L/R main; (4) ¼" unbalanced Omni (assignable); (2) ¼" TRS balanced L/R monitor; (1) ¼" stereo headphone
Channel Inserts	(2) TRS unbalanced ¼" (Ch. 1 and 2)
Digital I/O	(1 pr.) RCA, stereo S/PDIF
Other Ports	MIDI In, Out/Thru, MTC Out; (1 pr.) BNC word clock in/out; (1) SCSI mini-DB50 (backup only); (1) DB9 mouse port; (1) 8-pin DIN to Host
Dynamics Processors	One per channel; provide compression, expansion, gating, and ducking
EQ	One 4-band parametric per channel
Effects Processors	(2) dedicated to aux 7/8, or insert into any input channel or stereo bus
Sampling Rate/Resolution	44.1 kHz, 48 kHz/16-bit, 24-bit
ADCs/DACs	24-bit, 645 oversampling/24-bit, 1285 oversampling
Internal Processing	32-bit
RECORDER	
Hard Drive	2.5" IDE, 64 GB maximum
Max. Songs per Disk	30,000
Physical Tracks	16 mono plus one stereo mixdown
Virtual Tracks	8 per mono physical track
Recording Resolution	16-bit; 24-bit
Markers/Quick Locate Points	99/8 per song
Simultaneous Record/Playback Channels	16/16
Scene Memory	96 per song
Undo/Redo Levels	16
SAMPLING PAD SECTION	
Trigger Pads	16 (8 pads × 2 banks)
Playback	8 mono voices
Max. Playback Time	Approx. 90 sec. (16-bit, 44.1 kHz)
Sequencer	16 tracks
GENERAL	
Option Ports	(2) mini-YDGAI (24-bit)
Display	320 × 240-pixel, backlit LCD; fluorescent color display
Power Supply	120 VAC internal (IEC connector)
Frequency Response	20 Hz–20 kHz (+1, –3 dB, mic/line in to stereo out)
THD	<0.02% (@ 1 kHz, Omni out)
Dynamic Range	104 dB (typical)
Dimensions	22" (W) × 5.8" (H) × 18.1" (D)
Weight	26 lbs.

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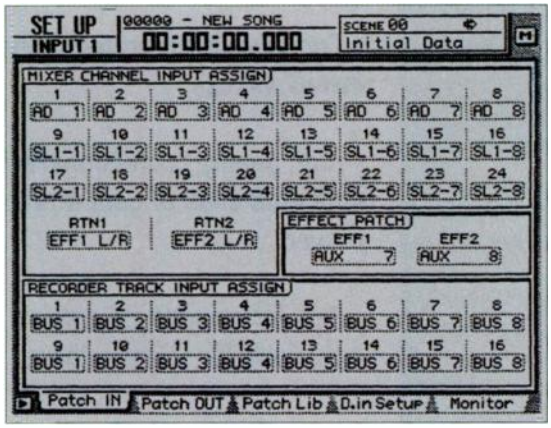

COURTESY YAMAHA

FIG. 3: Any combination of analog and digital inputs can be assigned to the faders. The 16 recorder tracks accept signals from channel direct outs or buses 1 to 8. Information about the current song, counter location, and scene is displayed at the top of the screen.

completely trouble free: pressing one button arms all 16 tracks and routes the inputs directly to the recorder, with a choice of inputs from the analog jacks or the option slots. The Quick Record feature wasn't especially useful, though; rarely do I find myself in a situation where I don't need to assign tracks in a more organized manner.

Each of the 16 physical tracks has up to 8 associated virtual tracks, and there is a stereo track for the finished mix. The virtual tracks let you record multiple takes and later choose which to include in the mix. This greatly expands your options. Of course, only one virtual track may be played at a time for each physical track.

Virtual tracks can be a real problem solver. I knew I could construct a lead vocal track from alternate virtual tracks by copying them to two or more open tracks, then editing those down to a single track. Once I'd filled all 16 tracks, though, I was initially perplexed by how to accomplish this task. Eventually, I simply copied them to another track's virtual tracks, made my edits, and replaced the original vocal track with the edited version.

Up to 30,000 songs can be stored on the internal drive. That ought to be enough for just about anybody. A 12 GB disk affords about 140 minutes of 44.1 kHz recording time at 16 bits. A song is stored as a Yamaha-specific type of WAV file, but support should be implemented for standard WAV import and export by the time you read this. Song files can be written to and restored from the optional internal CD-RW drive or any supported SCSI drive.

Songs with multiple word depths and sample rates peacefully coexist on the hard drive. The CD-RW drive doesn't handle sample-rate conversion—selecting 48 kHz brings up a reminder that you won't be able to burn a CD—but multiple dithering options are supported for the stereo digital and option-slot outputs.

A few dithering choices are downright odd, such as 17-, 21-, and 23-bit rates. Even members of Yamaha's tech support scratched their heads when asked what these rates are good for. One nice feature is that the AW4416 dithers on the fly when burning a CD, so you can take advantage of 24-bit mixing.

SLICING AND DICING

Editing takes place on three levels: Tracks, which run the length of a song; Parts, sections of contiguously recorded audio; and Regions, by which Yamaha means a unit of time. Incidentally, this wasn't the only time I encountered a specialized meaning for a common term. Another example is "optimizing," which refers to clearing a song's undo/redo buffer rather than performing a hard disk management routine (see **Fig. 5**). I wish Yamaha would stop using common terms in nonstandard ways; it is unnecessarily confusing.

I was not impressed with the implementation of the track-editing features. I was particularly disappointed to find that audio is muted when you hit the Edit button, so you can't hear what you're doing until after you've done it. Regions for editing must be selected in the Tracks page, the place in which recording occurs, rather than in the Edit pages. Repositioning markers requires quite a bit of shuffling back and forth. Because the key to successful editing lies in setting accurate markers, I found myself taking careful, handwritten notes. The AW4416 works more like a digital version of a cassette ministudio than a full-fledged random-access audio editor.

In a typical editing session, you might select the beginning and end points for a Part or Region, choose from several options, hit Execute, exit, and listen to the results. This is somewhat awkward, though admittedly it beats using a razor blade. Editing options include Name, Erase, and Copy, as well as various methods for moving data forward or backward in time. You can also shift the pitch and compress and expand the time base while editing. However, I was not completely satisfied with the results of these two functions.

I didn't get the hang of using the Nudge function, which loops a portion of the audio—the default is 100 ms—immediately before or after the current cursor location. The idea is to listen to the loop and use the Cursor/Jog and Shuttle dials to home in on an edit point. Nudging allows you to find locations for edit markers or auto punch-in/out points. With the counter display set to Measure, you can Nudge one beat at a time—but with a loss of accuracy. This feature was more difficult to use than it should have been.

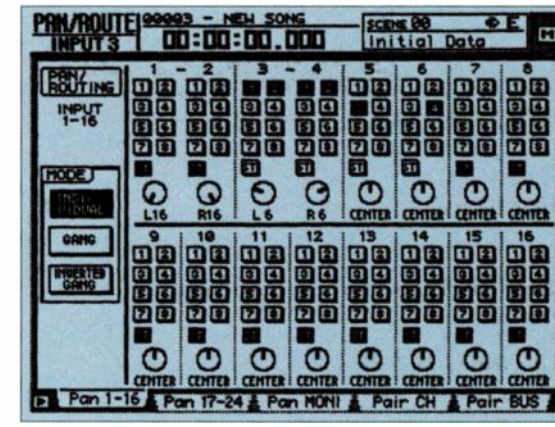

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FIG. 4: Panning and busing assignments for all 16 channels can be set from one convenient page.

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Many operations seem much more complex than they need to be. Because there's no way to connect a QWERTY keyboard, for example, even something as simple as naming a track involves a multitude of button pushes and cursor moves.

SAMPLING AND SEQUENCING

Advance publicity about the eight sample pads generated a lot of curiosity. After all, Yamaha has years of experience in both sampler and sequencer technology. You may be hoping these pads can replace your rack-mount sampler, but I wouldn't make any plans to sell the old warhorse just yet.

Samples up to 90 seconds long can be loaded from an existing track, WAV files, or audio ripped from a CD, and then be triggered from a pad. Without a waveform display, sample editing is extremely limited. For example, trimming a half-second means spinning the data wheel through as many as 24,000 samples. At about 20 samples per rotation, this takes awhile. It would be easier to copy a sample to an audio track, perform the edits, and then load the sample back to the sample pad.

I was similarly underwhelmed by the rudimentary sequencer that Yamaha includes for recording sampler data into a song. Unlike a true MIDI sequencer, pretty much all it does is record on and off points for the pads. With no input quantization, accuracy is especially critical. Editing is limited to erase, copy, and paste. If you're looking for a groove box or an onboard drum machine, forget it.

What are the sequencer and sampler pads good for? On one song, I copied a vocal chorus to a pad, thereby freeing up a track. On another, I flew in ambient swooshes from a sample CD. These capabilities are handy, but the inclusion of a full-function sequencer would make this unit a must-have.

There is hope. When I asked Yamaha if it had any plans for upgrading the sequencer in a software revision, the reply was, "That's a good idea. We'll look into it."

EFFECTS AND DYNAMICS

The two internal effects processors owe a lot to Yamaha's SPX-series effects units. At 32-bit resolution, they sound as good as or better than anything found in the average personal studio.

The processors generate the standard complement of effects, including basic reverbs and enough dual effects in series and parallel to cover most chores. You get modulation-type effects such as chorus, flanging, phasing, rotary speaker, and ring modulation. Stereo delay times can be as long as 1,350 ms. Dynamic filter, flanger, and phaser use an envelope follower to respond to the strength of audio signals, and you also can modulate them using MIDI Note-On Velocity messages from an external controller.

The dynamics processors provide compression, expansion, gating, and ducking. Forty preset dynamics programs are tailored for particular instruments, vocals, and voice-overs. These can be applied to the input channels, the tape returns, and the stereo and bus outputs.

If you're so inclined, you can get deep inside the processors and save custom patches (see **Fig. 6**). However, the effects libraries are stored as song data, so you must remember to copy them to a new location before deleting a song.

The unit's MIDI implementation is good but not great. You can use MIDI to change scenes from an external device, synchronize the AW4416 with an external device, and control the AW4416 using MIDI Machine Control. MIDI data flows through the In and Out/Thru ports or directly to a host computer, using the serial interface (if you can still find a computer with a serial port). A third MIDI Out port is dedicated to sync through MIDI Time Code. Although I had no trouble connecting my PowerBook, I soon learned that the AW4416 behaves better as an MTC master than as a slave.

Unlike the faders in many digital mixers and portable digital studios, the AW4416's faders do not send MIDI controller data. This would be a useful function, and I hope it will be implemented in a future update.

MIX MASTER

I will admit it: I love mixing with the AW4416's automation and moving faders. Of course, as with all love affairs, it has not been smooth sailing all the way.

In attempting a mix involving both recorded and MIDI tracks, I was surprised to learn that the automation didn't extend to the inputs. Ditto for aux sends and returns, making it difficult to fully utilize the internal effects. The only work-around was to create scenes with the desired changes, which caused problems down the road.

Otherwise, the automation is fairly easy to use. I particularly liked being able to customize, on a fader-by-fader

PRODUCT SUMMARY

Yamaha
AW4416
digital audio workstation
$3,599

FEATURES	4.0
AUDIO QUALITY	3.0
EASE OF USE	3.0
VALUE	4.0

RATING PRODUCTS FROM 1 TO 5

PROS: Solidly built. Records on 16 tracks at once. No audio file compression. Swappable 2.5-inch IDE hard drives. Flexible signal routing. 16- or 24-bit recording at 44.1 or 48 kHz. Mix automation with motorized faders. Two option slots. Multiple clocking and synchronizing options. SCSI, mouse, and computer ports.

CONS: Complex user interface. Poorly organized manuals. Sampling pads not fully implemented. Audio muted when editing. Cannot edit stereo tracks. Limited effects automation. Cannot preview playlists. Audio quality less than stellar. SCSI is only for backup.

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Introducing

WaveCenter/PCI

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

Here's a better idea: keep it digital with our new **WaveCenter/PCI** card. It has all the connections you need to integrate your digital mixer into your computer-based studio. Transfer up to 10 channels of digital audio simultaneously using ADAT lightpipe and S/PDIF, with 24-bit resolution on all channels. Use one of the built-in MIDI ports for mixer automation, and the other to connect your synthesizers.

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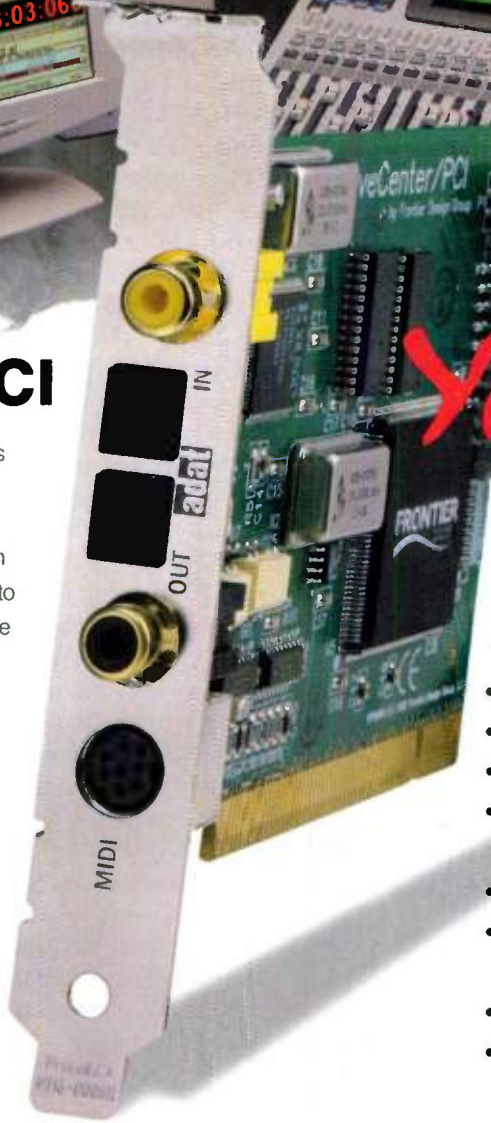

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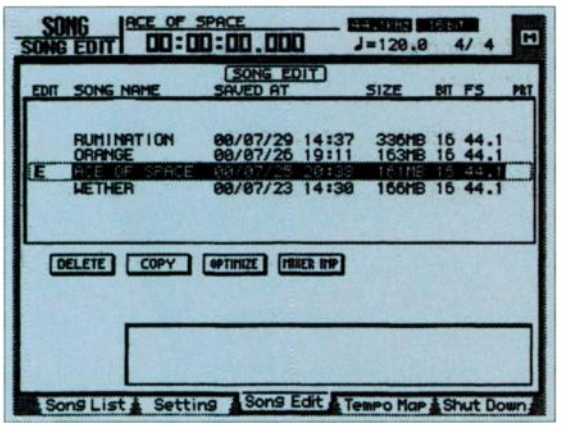

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FIG. 5: Creating a new song involves setting the sample rate and bit depth. Settings for selected effects, dynamics and EQ libraries, scenes, and automixes can be copied from existing songs. The Optimize button clears the undo/redo buffer, freeing up valuable disk space.

basis, the time it takes faders to return to their previous levels after punching out. There is a similar option for setting fade times between scenes. This option is deep because times are set individually for each input and return. Curiously, aux sends are exempt from this automation, too.

Once I was satisfied with a mix, I recorded it to the dedicated stereo track. Next, I looked into mastering and writing a CD, only to find I needed to set EQ and compression before I recorded the stereo track. That was where having multiple scenes became a problem. I had to duplicate my settings in each and every scene used in the mix; otherwise they'd revert back to defaults. If you're going to use the AW4416 to mix down any number of songs, you just have to accept the way it's designed to do things.

It was odd that there was no way to vary the length of time between songs on a CD. With Track at Once recording, the pause is fixed at two seconds, but the Disc at Once mode appears to default to zero. Because discs recorded in one continuous pass are preferred for duplication, this could be a serious shortcoming. The only way to keep songs from bashing into each other is to append time to the end of each stereo track. Even worse, there is no way to preview the playlist. I'd suggest using the internal CD-RW drive for making backups and discs for your friends. Have critical projects professionally mastered.

HOW'S IT SOUND?

As a test, I simultaneously tracked a live band to the AW4416 and a first generation ADAT at 16 bits. Surprisingly, the tape tracks had more sizzle, especially on the cymbals and rhythm guitar. I hadn't expected that; after all, the Yamaha unit is much newer. Recording at 24 bits sounded better, thankfully.

This makes me question the desirability of recording an all-acoustic folk or jazz project on the AW4416. Sessions involving electric instruments and lots of processing worked just fine. Tube fans won't be thrilled with the EQ or dynamics, but they do get the job done.

Don't forget that the option slots open up the possibility of using high-end digital converters. I'd certainly want to have a multichannel outboard preamp on hand, too.

EXPANDING YOUR WORLD

The AW4416's two option slots accept a variety of mini-YGDAI expansion cards, ranging in cost from $269 to $369. Three of the cards are digital audio interfaces. The MY8AT provides eight channels of ADAT Lightpipe I/O. The MY8TD is a similar interface for Tascam TDIF. The 8-channel MY8AE is an AES/EBU interface that requires a breakout cable, an additional $199 bite.

The other three cards are analog interfaces. You get eight inputs on ¼-inch TRS jacks with the MY8AD , and the MY4AD gives you four inputs on XLR jacks. If you need four analog outputs on XLR jacks, you can use the MY4DA.

I borrowed a Lightpipe card from Yamaha to transfer tracks to and from an ADAT XT20. Installing the card and setting the clock source was a snap. I easily flew in rhythm tracks recorded on ADAT at the start of a new song.

To take advantage of some enhanced editing features, I also used a MOTU 2408 audio interface with *Digital Performer*. I synchronized the two systems using MIDI Time Code, using the AW4416 as the master. Because resolution is measured in frames rather than samples, I had to take extra care that my edited tracks didn't shift relative to the rest of the performance, but it worked.

MANUAL WOES

Considering the deep interface, it's too bad the three manuals do not offer enough guidance. The *Reference Guide* is organized in the order that buttons appear on the unit's surface. Worse, it only covers features that appear in the display screen, leaving out large portions of the interface.

The hefty *Operations Guide* takes a more functional approach, but many things aren't fully covered there, either. My frustration grew as I kept riffling through first one guide and then the other.

The *Tutorial* introduces a few basic mixing techniques. However, it's not particularly useful, considering the huge depth of this product. Because it uses a demo song found on a CD-ROM, the *Tutorial* is meaningless without the optional CD-RW drive.

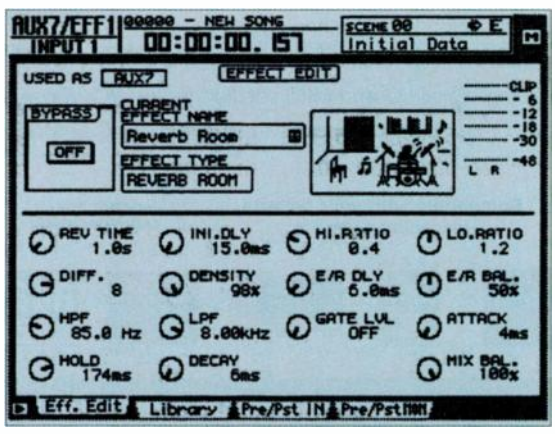

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FIG. 6: Forty-one effects presets are always at hand; the 87 user patches are stored as song data.

DEEP AS A WELL?

In spite of annoying manuals and a very tough learning curve, I have enjoyed using the AW4416. However, it may be too deep for some users. Most folks want something that's ready to use right out of the box—instant gratification is what made cassette mini-studios so cool. Although you can be up and running on the AW4416 relatively quickly, it is also easy to get frustrated and confused. Just ask anyone who was within earshot of me on my first few days.

If you're familiar with Yamaha's digital mixers, you should feel right at home.

Frustrations notwithstanding, in many ways this is the ideal rig for the serious home recordist. You can still get a lot of use from it without ever plumbing the depths.

The AW4416 is one serious machine. It could easily find a home in a variety of pro studios, either as a primary or secondary recorder. In fact, with two expansion slots, pro-level routing, and onboard effects, it's a great buy just as a digital mixer; the recorder is a bonus! By connecting outboard mic preamps, this would be a great live remote rig. Just don't expect to use it on an important session the day you unpack it.

A few features don't live up to their promise. In particular, the sampling pads could be better implemented. Mastering internally also leaves a lot to be desired. But all in all, the AW4416 more than delivers on the promise of an entire studio's worth of gear in one tidy package.

Mark Nelson *'s first studio was the size of a cigar box. A lifelong acoustic musician, he lives and records in Southern Oregon's Applegate Valley. Thanks to Craig Martin and Freeman Studio.*

CREAMWARE AUDIO

POWERSAMPLER 2.03 (MAC/WIN)

Hardware and software blend into a DSP-powered sampling system.

By Eric W. Bell

CreamWare is known for products that combine high-quality DSP-based audio cards with innovative software designs. Among the company's most recent efforts is PowerSampler, which serves up the best features of hardware- and software-based samplers as it avoids their pitfalls.

PowerSampler, which is PC and Mac compatible, is three products in one: a 32 virtual-channel PCI audio card; a software environment with an array of performance, driver, and interface options; and *STS-3000,* a sampling application. This hybrid package offers consistently high sound quality, low latency, and a competitive set of capabilities.

PowerSampler offers distinct advantages over dedicated sampling hardware and software samplers. Its lowest latency, an incredible 2 ms, guarantees that the system responds with the feel of a hardware device. Because it uses the computer's RAM for samples, a computer with 256 MB of RAM yields more sample space than most hardware samplers, which tend to top out at 128 MB. Finally, it's much easier to perform operations by using the computer's monitor than by staring into most samplers' modest LCDs.

PowerSampler has advantages over sampling software as well. Because its DSPs perform the audio data crunching, there is little drain on your system's CPU, even when playing a large number of 44.1 kHz samples. That frees your system to run other applications, such as a hard disk recording program, in tandem with PowerSampler. It also ensures (with a few exceptions) consistent and reliable polyphony regardless of the computer's platform or power. Because PowerSampler has its own audio hardware, the sound quality is not affected by the type or brand of sound card you own.

FIG. 1: The PowerSampler toolbar gives quick access to all areas of the system, including the Environment and Ultra Low Latency settings.

SAMPLING SPECS

The PCI board supplied with PowerSampler has stereo digital audio I/O through an ⅛-inch S/PDIF connector. Its stereo analog I/O is accessible through ¼-inch TRS jacks. The supplied software supports common audio formats up to 32-bit, 96 kHz. With 16-bit, 44.1 kHz samples, PowerSampler offers 32 simultaneous stereo voices.

When connecting digitally, PowerSampler operates as a word-clock master or slave. You can choose from transfer rates of 32, 44.1, 48, or 96 kHz at 16- or 24-bit resolution. To interface the board to your system, a broad variety of software driver options are provided, and a built-in software-mixer window lets you manage signal routing and levels. MIDI In and Out ports are provided, as is an internal MIDI driver for connection to another program running on the same computer.

The two stereo physical outputs (one analog and one digital) may seem limiting, but you can get around that by configuring up to eight stereo channels internally for routing to other audio applications using ASIO or EASI. You can also get an additional eight analog ins and outs using the optional Luna 2496 breakout box ($598), which is connected through the board's Z-Link jack. Adding a 16-channel ADAT interface is another solution, but that requires the optional Z-Link/ADAT expansion plate ($198). There is no RAM on the PowerSampler board—if you want more sample memory, just add RAM to your computer.

Minimum System Requirements

PowerSampler

PC: Pentium II/300; 64 MB RAM; Windows 95/98; 100 MB free disk space

MAC: G3/300; 128 MB RAM; Mac OS 8.6; 100 MB free disk space

One thing you should be aware of: real-world usage varies from the best-case specs quoted in product literature. For example, you lose two to four notes of polyphony for each of the PowerSampler audio drivers (other than stock WAV drivers) that you enable. Also, PowerSampler's mixer uses significant CPU cycles while onscreen. Simply kill the mixer display if you need to counter this—it's still functioning even when not visible.

If you take advantage of the 24-bit hardware support, the board will use a second PCI channel. Finally, if you crank PowerSampler's word-clock rate to 96 kHz, you can't use the sampler software. According to CreamWare, there isn't enough DSP power to support the high transfer rate and the sampler software simultaneously. That isn't a problem, however, if you do your high-rate recording when the *STS-3000* application is not running.

SAMPLER IN A BOX

Along with the sound card, the PowerSampler bundle contains an installation CD; an Akai-format sample CD with various synth, acoustic instrument, and drum samples; two MIDI adapter cables (to convert the card's mini-DIN MIDI jacks to standard MIDI connectors); and an installation booklet.

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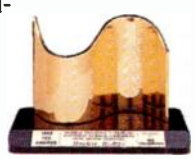

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But if you only leave this page understanding why you should use active monitors, our point will be made: *If your speakers aren't accurate, nothing else is either.*

Hearing is believing.

To discover the details that passive monitors miss, bring a familiar CD or DAT mix into your nearest Mackie Dealer and hunker down for a serious HR824 listening session. We think you'll be pleasantly surprised.

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*"By active, we mean a speaker design where each transducer constantly interacts with its own internal power amplifier... not a "powered" passive speaker with an external amp glued to its back.

The PowerSampler board itself is compact and cleanly laid out. To double the polyphony, connect two PowerSampler cards using their internal S/TDM bus connectors. CreamWare's SCOPE and Pulsar systems can use a connected PowerSampler card in a similar way, but PowerSampler software won't recognize those systems.

PDF-format manuals for the hardware and software load onto your system during the software installation, ready for online perusal or printing.

ENVIRONMENTALLY FRIENDLY

PowerSampler's toolbar (see **Fig. 1**) provides quick access to the STS-3000's Main, KeyGroup, and SampleEditor pages; the Mixer window; and a file browser. (The browser can display partitions from Akai-format CDs.) You access configuration dialog boxes from two menus at the display's top right.

PowerSampler's overall configuration, which includes the enabled drivers, selected sample rates, audio I/O and MIDI options, mixer settings, and voice presets, is called an Environment. You can save and restore Environments, and a default Environment loads when PowerSampler starts. You can change the default to the currently loaded Environment through a menu option.

Audio-driver options in a PowerSampler Environment are vast. You can choose from ASIO (16- or 24-bit, but not both), EASI, TripleDAT, NemeSys *GigaSampler*, 16-bit stereo WAV, and three versions of multiple WAV. Multiple WAV lets you configure up to eight pairs of stereo WAV drivers for routing audio to an internal hard disk recording program or for sending multiple audio applications to PowerSampler simultaneously. The Mac version supports Sound Manager, ASIO, and EASI, and uses Open Music System for MIDI. Under Windows, DirectSound is supported but only as an input. If you enable the ASIO or EASI drivers, keep in mind that these use DSP power, resulting in a loss of three to six notes of polyphony each.

You can route MIDI internally (for triggering the sampler from a sequencer) or externally (to send MIDI data out the MIDI ports), but not both. Get around this limitation by using the MIDI ports on a sound card or standalone MIDI interface in conjunction with PowerSampler's MIDI options.

The Sample Rate Setting dialog box is where you configure the S/PDIF port as a word-clock master or slave and set the sample rate and resolution. The Ultra Low Latency Interface dialog box lets you select from five latency options. At 44.1 kHz, these are 3, 4, 7, 13, and 25 ms. Different latency rates are available at different sample rates. Low latency isn't free, however: the cost is system CPU cycles. If you find things run sluggishly, increase the latency time to free up some CPU muscle.

MIX IT UP

On startup, PowerSampler acts as a standalone audio board. Its flexible 24-channel mixer manages the system's audio capabilities (see **Fig. 2**).

The mixer integrates fully with the *STS-3000* sampler (when it is loaded) but you can also use it for other purposes. For example, you can mix audio channels coming into PowerSampler from its external hardware inputs, and ins and outs from internal sources (such as other audio applications). You can use the various options to mix audio sources and route them to a hard disk recording program, or you can create a monitor mix of your hard disk recording program's own audio outputs.

The mixer's 24 channels are arranged in three groups of eight, and you can view one group at a time. You can also toggle the view to show the effects-control page. Click on the appropriate label to select a channel group.

Right-clicking on a channel strip's Input or Output Routing fields brings up an options menu. Route a channel strip's output to more than one destination to monitor any channel's output without taking it out of its set routing. You can save mixer presets, which is useful because setting up complex routings with large numbers of signals is time-consuming.

In addition to a volume fader, level meter, Pan pot, and Mute/Solo buttons, each channel strip offers Chorus and Delay sends. A stereo-link button gangs adjacent channel strips, and a Mix button routes the channel's output to the Master channel. Naturally, you can route the Master signal to any of PowerSampler's available outputs.

The number of built-in effects PowerSampler offers is meager in comparison with most newer hardware samplers; the only choices are stereo Chorus and Delay. One work-around is to use the sends to route audio out the S/PDIF port to an external digital-effects unit and then back through the returns.

PowerSampler's Delay sounds fine, and I particularly liked it on drum loops with delay times set to match the loop's tempo. The Cross Feedback setting feeds the left delay output into the right channel and vice versa. This yields more variations in rhythm and stereo-field placement while matching the original loop's tempo. You can type values for the delay directly into the data fields, which is easier than using the finicky onscreen knobs. Alternatively,

PowerSampler Specifications

Analog Inputs	(1) ¼" stereo line input
Analog Outputs	(1) ¼" stereo output line
Digital I/O	(1) S/PDIF ⅛" mini phone jack
MIDI I/O	(1) mini DIN input (1) mini DIN output (adapters supplied)
Internal Connectors	(10) S/TDM bus connectors for connecting to additional PowerSampler or Pulsar cards
A/D Converters	24-bit, 96 kHz
Sampling Rates	32, 44.1, 48, 96 kHz

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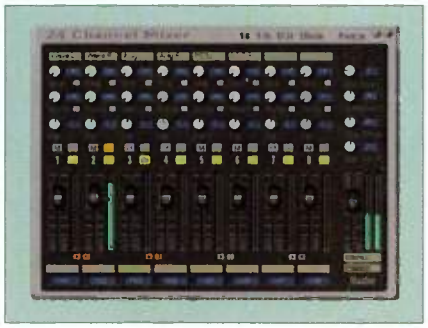

FIG. 2: The 24-channel mixer offers settings for adjusting delay, chorus, pan, and volume levels. You can view eight channel strips at a time.

you can use keyboard shortcuts to adjust most of PowerSampler's controls.

You can route each effect's outputs to any of PowerSampler's available destinations. Each effect has a Mix for sending the effect to the Master output in addition to or in place of the separate outs.

SAMPLER SCENARIOS

The *STS-3000* software is the heart of PowerSampler. It's compatible with a variety of audio and sample formats: Akai S1000 and S3000, SoundFont 2.0, AIFF, and WAV. The PDF manual is complete, with plenty of illustrations, examples, and a tutorial. *STS-3000* lets you group samples on your hard drive and CD-ROM drive for accessing, configure or edit a large number of performance parameters (such as output and MIDI channels, volume, delay and chorus sends, envelopes, and panning), edit samples or record new ones, and more.

STS-3000 allows you to load up to 16 stereo programs simultaneously in RAM. Load times are reasonably quick—much quicker than those of some hardware samplers. A 70 MB program, for example, loads in 90 seconds. Meters on the main STS-3000 display panel's Memory page provide feedback on memory usage. Each program contains one or more KeyGroups, which specify the range of MIDI note numbers that the program responds to. KeyGroups contain four zones into which you can load samples. Using this hierarchical structure, you can create a program that uses a single mono sample or more complex stereo programs containing multiple samples with Velocity sensitivity. Zone crossfading, which blends samples at crossover points between KeyGroups, is supported but uses an additional note of polyphony.

With *STS-3000* you can directly import SoundFont 2.0 or Akai S1000 and S3000 programs with their internal playback parameters intact. A native format called Scope is also supported. *STS-3000* handles 8- or 16-bit mono and stereo sample files. To hear samples, drag one from PowerSampler's browser to one of the four zones in the KeyGroup (see **Fig. 3**). Stereo samples take up two zones. Unfortunately, there is no way to directly audition samples from the browser.

Many program parameters, including loop points, are nondestructive and don't affect the original sample data. Samples are always loaded from their original locations; PowerSampler uses

a pointer system to reference the drives' data. That efficient method lets programs include samples from various drives (including CD-ROMs) and subdirectories.

The Preset panel stores groups of 16 programs—complete with parameters, sample locations, and settings—ready for loading. You can save and recall presets. Although PowerSampler's main panel allows only 16 active programs at once, the Pool panel has room for 128 program references and also has its own preset list. Pool programs are loadable using MIDI Program Change commands. The Load Once switch caches programs in RAM for as long as possible to minimize subsequent reloading.

PARAMETERS FOR CHANGE

Among the options you can choose from on the *STS-3000* Main page are program-level settings for MIDI channel, overall level, panning, output-channel routing, tuning, and transposition. The Priority setting lets you manage which programs are usurped if PowerSampler runs low on polyphony. Additional pages affect settings for entire programs. These include Loudness, Filter, LFO1, LFO2, Pitch, MIDI, Tuning, and Soft (which contains MIDI soft-pedal settings).

The Loudness page features the Modulation Matrix, which is a flexible way to map modulation sources to controllable parameters. Sources include standard MIDI controllers, envelopes, and two LFOs. To set a modulation source, click on a parameter's label and choose from the menu. You can use more than one modulation source to control a single parameter, adding to expressive capabilities.

The two LFOs (each with triangle, sawtooth, square, and random waveforms) feature graphic displays for editing delay, speed, and depth. You can set these values numerically as well. Engaging De-sync allows each of PowerSampler's voices to be modulated independently with the start of the LFO cycle synched to the start of each individual voice rather than all voices sharing one LFO cycle.

Aftertouch, Pitch Bend, and other modulation sources can be employed to affect pitch changes. In addition, each program has an overall tuning control adjustable between ±50 semitones and ±50 cents.

The KeyGroup List panel provides all the tools needed to create and manage zones, load samples, and edit parameters. A MIDI-to-Span function is available for setting KeyGroup ranges using MIDI input.

Global KeyGroup settings include overall loudness and tuning values, zone crossfading, pitch modulation, and detune. A MuteGroup control lets you associate mutually exclusive KeyGroups, which is useful for coordinating open and closed hi-hat sounds, for example.

The Loop page is used to set and edit looping for a sample. You can loop samples with or without the data trailing the loop points being played on key release, from beginning to end, or

FIG. 3: The file browser shows all the available samples and programs on your system. It can also display partitions on Akai-format disks.

not at all. Loop points set here override but do not overwrite settings in the raw sample unless you explicitly save the sample and its settings. The Loop-tuning setting tweaks the pitch of just the looped part of the sample.

I enjoyed exploring the capability to use different resonant-filter settings for each KeyGroup in a program. For each KeyGroup, you can choose which modulation sources will affect the filter's frequency, resonance, and key-follow values. This allows subtle (or wild) changes in filtering for pitches up and down the keyboard.

Two envelope types are also available. As with filters, each KeyGroup has its own envelope settings. Env 1 is an ADSR envelope with attack and release times alterable in accordance with the trigger-note Velocity. Env 2 is a four-stage ADSR envelope with settings expressed in terms of rates and volumes for each stage, providing more flexibility over attack and initial decay than Env 1.

There's simply not enough space here to discuss all the ways you can shape sounds in PowerSampler. Two important pages deserve mention, however: at the program level, a MIDI control page lets you set channels, program numbers, key ranges, and the like; a MIDI tuning page lets you set tuning offsets on a note-by-note basis.

SAMPLE EDITING

PowerSampler's SampleEditor window contains basic tools for recording and editing stereo sample data, such as the familiar Cut, Copy, and Paste options, along with some additional tools (see **Fig. 4**). You can see your sample data on the large graphic-waveform display, which you can split when editing loop points. You can zoom in on the display and scroll during playback.

The AutoSelect feature automatically loads the SampleEditor window with any highlighted zone's current sample. You can't edit a sample file directly, though; it must be part of a program.

You can trigger recording with the Rec button, with a MIDI Note On trigger, or when the sampler's input level crosses the threshold you have set. Threshold-triggered recording solves synchronization problems and saves you the trouble of trimming unwanted leading sample data. The Pre Record option uses a buffer of up to 333 ms so you can add sound to the start of a recording after the fact. This ensures that sharp attacks are captured intact.

The SampleEditor's meters display playback and record levels. A single pot applies a boost of up to 12 dB to both channels. Because recordings are limited to stereo samples, mono signals can be stored in either the left or the right channel.

Editing tools include peak-level Normalize, DC Offset, and Mute, which simply zeroes the selected data. In addition, there are some rudimentary commands to set and edit loop points. The Snap function causes a loop-point edit to "snap" to nearby values matching the slope of its corresponding loop point. This reduces popping and smooths your loop.

The SampleEditor's unaltered playback signal is presented on PowerSampler's Spl L and Spl R outputs (Sampler Left and Sampler Right). To hear the outputs, assign them to the inputs of channels in the mixer.

Although serviceable, the SampleEditor is basic. For example, there is no crossfade loop tool. A number of third-party editing tools are much more comprehensive in ability, but no provision is made for using an external editor. If you edit data outside the program, be prepared for extra mouse work getting your edited samples into PowerSampler programs before you can use them.

POWER DOWN

PowerSampler is a great-sounding tool with most of the features and capabilities of a basic hardware sampler at little more than a software-sampler price. However, some areas in the software need work. To maintain a standard look and feel on PCs and Macs, CreamWare uses an interpreter: an extra layer of code between the user interface and the operating system. This causes problems with the user interface, namely slow redraws, slow response to user input, flickering cursors and display elements, and hard-to-manage windows.

PRODUCT SUMMARY

CreamWare Audio Solutions

PowerSampler 2.03 (Mac/Win) sampling system

$649

FEATURES	4.0
EASE OF USE	2.5
AUDIO QUALITY	4.5
VALUE	4.0

RATING PRODUCTS FROM 1 TO 5

PROS: Low CPU requirements for sound processing. Highly configurable program parameters. Comprehensive driver support. Access to large memory space for samples. Many features for the price. System scalable with additional cards.

CONS: Software problems limit some functions. Sluggish nonstandard user interface. No sample auditioning before loading. Few built-in effects.

Manufacturer

CreamWare Audio Solutions, Inc.
tel. (604) 435-0540 or (800) 899-1939
e-mail info@creamware.com
Web www.creamware.com

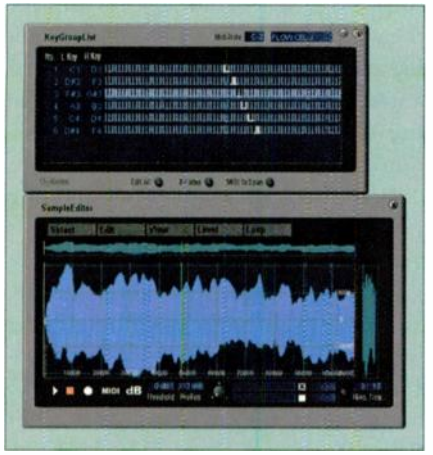

FIG. 4: The SampleEditor's main screen shows a graphic view of your audio data and provides access to numerous editing functions.

Some sluggish operations are simply unbearable. For example, the Sampler Window redraws its display every time you click on the window. In one test, a 64 MB sample caused a 20-second redraw time. A third-party dedicated audio editor took about one-tenth the time to redraw the same data.

Standard Windows interface capabilities, such as cursors, file Open and Save dialog boxes, context sensitive help, and floating tool tips are missing from PowerSampler. And if you use a screen resolution less than 1024 × 768, menus in the Mixer display run off the screen, prohibiting the selection of some options. Mac users will no doubt have their own list of interface beefs.

Despite the manufacturer's claims that you will forfeit very little CPU power when using PowerSampler, some operations do impose penalties. A faster setup than my 366 MHz Pentium test system should give better results.

I also experienced a frustrating lack of response to my e-mail and telephone inquiries to CreamWare's North American support.

PowerSampler's software has made great strides from the 1.33 release to the 2.03 version covered here. As soon as CreamWare squashes those few remaining bugs, there will be even more reason to recommend PowerSampler. If you are looking for an inexpensive sampling solution but want the power and sound quality of a hardware unit, you may find PowerSampler an excellent fit. Musicians working with very large samples will appreciate PowerSampler's ability to use system RAM. If your setup includes other ASIO or EASI compliant programs, you'll appreciate PowerSampler's support for those drivers and their multiple direct audio channels.

If you are not into programming sounds, PowerSampler offers easy access to Akai, SoundFont, and other sample libraries. But if you do like to do your own sampling, looping, and tweaking, PowerSampler's deep set of parameters and settings can help you take your sounds just where you want them to go.

Eric W. Bell *is a writer, programmer, and musician. More music musings are at www.howlingdog.com/howls.htm.*

SYNTHESIS TECHNOLOGY

MOTM SYNTHESIZER

A modular analog synthesizer that surpasses all expectations.

By Robert Rich

Lately I've been asking myself, "What madness would induce a musician to invest thousands of dollars in an analog modular synthesizer when everyone is moving toward software emulations?" Software synthesizers offer more features at a lower cost and don't take up the space of hardware synths. However, they lack the immediacy and physicality of a real instrument.

When I first encountered Synthesis Technology's MOTM system and its wall of conservative black faceplates covered with large gleaming knobs, I recalled the excitement I felt before the days of MIDI. Creating electronic music then was a tactile and visceral experience, and the instruments were big and messy. MOTM stands for "MOTher of all Modulars," and though that's quite a claim, the exceptional quality of Synthesis Technology's system might validate such hyperbole. The MOTM system balances modern audiophile standards with some of the more desired quirks of retro analog synth design.

BACK TO THE FUTURE

Whereas a MIDI-controlled synthesizer, by its nature, operates in terms of discrete note events, a modular synth works from a continuum: sound is created by complex swings in voltage and the interacting patterns of clocks and switches. A patch on a modular synth is an environment for inflection and nuance: a living and changing entity far removed from the preset mentality brought on by recent keyboard developments.

Synthesis Technology's MOTM modules are designed to fit standard 19-inch racks, so you can mount five double-wide modules side by side. The system in this photo includes a row of 6 modules, for a total of 11 modules in a portable SKB case.

The physical accessibility of each function in the various components of a modular synthesizer makes it easy to explore new sound possibilities. Anything that creates a voltage can become a controller, and anything that makes a sound can become part of a patch.

A modular synth can be integrated into the modern studio in ways that were not possible during the instrument's heyday. A MIDI-to-CV converter and sequencer software give substantial control over the modules through MIDI. The recording and sampling capabilities of a digital audio workstation further increase the usefulness of an instrument that requires repatching to get new sounds.

Although the idea of programming with patch cords is primitive by today's standards, it's more direct than programming a preconfigured synthesizer. You don't have to spend hours fussing with routing assignments, SysEx parameters, and device IDs as you do when you set up a hardware controller for a soft synth. Using a modular hardware synthesizer is a truly satisfying plug-and-play experience.

SELF-SERVICE

A modular synth might even inspire you to grab a soldering iron. Although preassembled and tested MOTM modules are available, Synthesis Technology also offers each module in kit form. The kit versions are about 30 percent cheaper than the preassembled ones. Besides the satisfaction you will gain from creating a system by hand, you might have enough money left over to buy a few extra modules. For those new to DIY, the Synthesis Technology Web site has a regularly updated tutorial page that includes a list of supplies and books for getting started, as well as MP3 demos of each module.

Synthesis Technology grades most modules by difficulty (1 is the easiest, 5 the most difficult) and, in some cases, indicates the amount of time it will take to build the module. For this review, I built the MOTM-110 VCA/Ring Modulator module, which has a difficulty rating of 2. It took me about four hours to build.

The documentation that comes with the modules is clear and precise, featuring thorough assembly and testing instructions, schematics, and calibration procedures. The kit includes everything except a soldering iron, tools, and a voltmeter. The wires are even precut, stripped, and tinned. The module worked perfectly the first time I plugged it in, and I calibrated it in a few minutes to within a millivolt of specification. Now I have a deeper respect for the thought

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Electronic Musician 2001 EDITORS CHOICE

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and precision that goes into the design of this high-quality system.

Perhaps you don't want this kind of intimacy with your instrument. Even if you choose to buy your modules completely assembled, you could find yourself getting your hands dirty: the MOTM system encourages hands-on interaction, and you'll get better mileage from it if you have a knack for DIY (see **Fig. 1**).

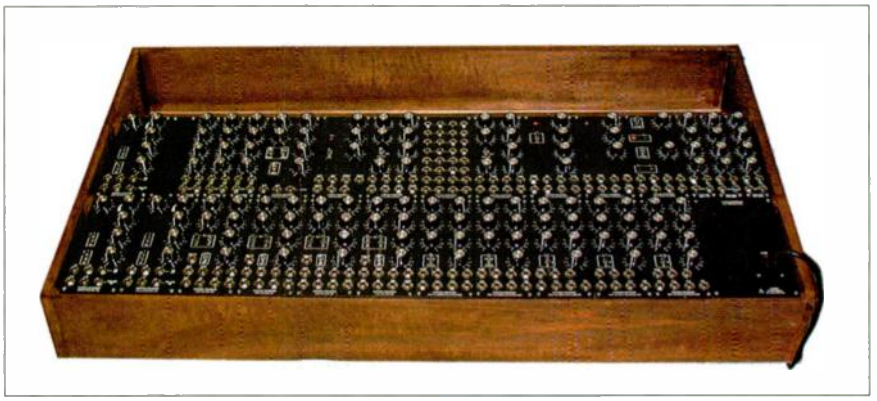

FIG. 1: The MOTM system encourages DIY customization. When Robert Rich's system outgrew its original rack case, he built a wooden cabinet that holds about 40 modules, upgraded the power supply, and rewired the power distribution.

KICKING THE TIRES

The various modules' components—¼-inch-thick aluminum faceplates, large instrument-grade knobs, high-quality Switchcraft ¼-inch jacks, sealed Bourns pots, and low-leakage capacitors—show attention to detail throughout. The circuit boards have solid ground planes and mount to a steel backplate that helps shield against unwanted hum and noise (see **Fig. 2**).

Through its care in the design process, Synthesis Technology has overcome the problems that plague most modular systems. One interesting and useful design feature is the placement of the I/O jacks in the module's lower section, which keeps the patch cords away from the controls. Synthesis Technology's modules have a high signal-to-noise ratio, impressive thermal stability, and low leakage and hum. The sound is neither dry nor clinical; you can easily saturate the filters and make them growl, just as you can with an old Minimoog or a Sequential Circuits Prophet-5.

Synthesis Technology also has a conservative approach to design. MOTM modules will never fulfill the addiction to blinking lights that some analog junkies have; only four of the modules sport LED indicators. According to the designer, each LED adds noise to the system in the

form of added current drain and switching transients. By avoiding gratuitous flashing, each module's sonic performance is improved. It seems like a fair trade-off, but I wish a few of the modules (especially the envelope generators) gave a visual indication of activity.

The MOTM modules fit neatly into a 19-inch rack by mounting to steel brackets (available from Synthesis Technology) that horizontally span the rack rails. Each module is 5U high and comes in multiples of the 1¾-inch width: single-, double-, and triple-width. Five double-wide modules fit the traditional 19-inch rack.

The modules are less than five inches deep, so they are shallow enough to fit into a 12-space SKB pop-up mixer case. Depending on the configuration, those cases can hold 10 to 12 modules (see opening photo).

The MOTM system is continually evolving, and new modules are due for release every few months. Although the current module selection seems a bit conservative, it provides a strong foundation for a system that will grow into a monster in the near future. (Synthesis Technology says that three new modules will be shipping by the time this goes to press.)

The current module selection includes a VCO, an LFO, a sample and hold with noise, a VCA and ring modulator, an envelope generator, a voltage router, a suboctave multiplexer, and three filters.

OSCILLATORS

MOTM-300. The MOTM-300 Ultra Voltage-Controlled Oscillator sports a frequency range of 0.2 Hz to more than 38 kHz, and it exhibits better tracking and thermal stability than any other analog oscillator I have used. In typical Synthesis Technology fashion, this module includes a host of useful features without gimmicks. Each of the VCO's four waveforms—sine, triangle, sawtooth, and pulse—has a dedicated output jack that can be used simultaneously with the others (see **Fig. 3**). The module also includes a 1V/octave input and two FM inputs. The first FM input has dedicated switches for exponential or linear FM tracking and AC or DC coupling. Control knobs are included for pulse width, pulse-width modulation (PWM) depth, coarse and fine tuning, and depth control for the FM inputs.

The MOTM-300's sync feature is particularly interesting. The sync jack emits a pulse that can drive other VCOs. However, the jack can also accept an external sync signal, depending on the position of the Hard and Soft Sync switch. In Hard Sync, the sync jack acts as an input only. In Soft Sync, it acts as both an input and output simultaneously. The harmonic richness that results from Hard Sync, in which the pitches of the oscillators are locked together, is part of what gives an analog synth its characteristic sound. On the other hand, Soft Sync, a rare innovation that first appeared on the E-mu modular synth, latches two oscillators together when they are close in frequency but allows for drift when the frequencies are farther apart.

MOTM-320. The MOTM-320 Voltage-Controlled LFO has the same high stability as the VCO, as well as sine, triangle, ramp, and pulse outputs that you can use concurrently. The frequency range is from one cycle every 30 minutes to 2.8 kHz. Along with a 1V/octave input, the MOTM-320 has a Hard Sync input, an FM input, a rate control, an FM-scaling control, and a wave-shape control.

The MOTM-320 provides voltage-controllable waveshaping through the Shape input. This feature narrows or widens a pulse wave, makes a sine wave asymmetrical, or morphs a sawtooth from an upward ramp to a downward ramp. However, more sophisticated processing is also possible. For instance, to create syncopation in a particular piece, I synched the LFO to a clock and triggered an envelope generator from the LFO sawtooth. That turned the shape control into a continuous groove control that moved the trigger timing earlier or later in relation to the beat. To hear an example of this, visit www.emusician.com/emlinks.

FILTERS

The MOTM system really shines when it comes to filtering. Each of the three filter modules sounds both stunning and unique. For signal processing alone, these filters might inspire an investment in a small MOTM system.

MOTM-420. The MOTM-420 Voltage-Controlled Filter has a 12 dB/octave response and emulates the filter on a Korg MS-20 synthesizer. A switch selects among highpass, notch, and lowpass types of filters. The MOTM-420 has three audio inputs (each with its own level control), two scaleable FM inputs, and cutoff and resonance controls.

You can push the MOTM-420 into full resonance, causing it to cross-modulate with the incoming audio signals in fun and nasty ways. At maximum input levels, the filter sounds good and fat as it begins to distort. The 2-pole response lets some of the harmonics leak through, so the MOTM-420 tends to sound more open and cutting than the MOTM-440 filter.

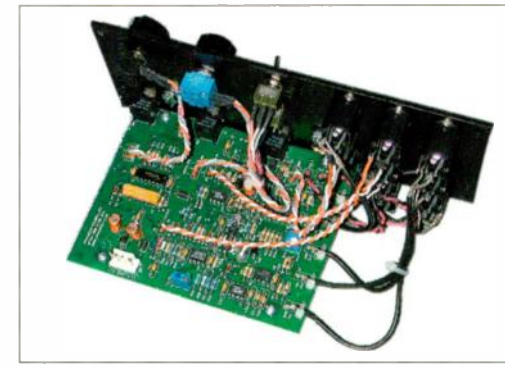

FIG. 2: Synthesis Technology's attention to detail and its modules' high-quality components can be seen in this view of the MOTM-300 Ultra VCO circuit board.

MOTM-440. The MOTM-440 Discrete OTA Voltage-Controlled Lowpass Filter (with VC Resonance) has a 4-pole response that emulates the old SSM filter chips. Filters designed with SSM chips gave the Octave-Plateau Voyetra 8 and the Rev 1 and 2 versions of the Sequential Circuits Prophet-5 their thick, meaty sound. Those filters got their famous growl from subtle overmodulation within the filter itself, and the MOTM-440 manages to improve on that sound with a better signal-to-noise ratio.

Synthesis Technology considers the MOTM-440 to be its signature filter, and I can see why: it is one of the best-sounding filters I have ever heard. Like the MOTM-420 filter, the MOTM-440 has three audio inputs and two FM inputs, and, of course, you can push the filter into full resonance. You can also make it distort, and the resulting overdriven sound differs significantly from that of the MOTM-420. The MOTM-440 also includes a Bass Enhance switch that increases the bass frequencies for an absolutely huge, throbbing low end.

MOTM-410. Based on the Korg PS-3100 filter, the MOTM-410 Triple Resonant Filter complements the other two filters with its interesting formant-shaping abilities. The MOTM-410 contains three sweepable bandpass filters and two asynchronous sine-wave LFOs (one of the LFOs runs 20 percent slower than the other). The filters have a fixed resonance that lets them evoke the vowels of the human voice or impart shifting articulations to the source sound. Each frequency band has a dedicated tuning control and output, so you can pan a sweeping signal across several speakers or sum the output to mono.

The MOTM-410 allows voltage control over the LFO rate (from 0.02 Hz to 100 Hz), modulation depth, and Sweep. Sweep controls all three frequency bands simultaneously. Also, a Mix control varies the amount of unprocessed signal in the output.

A three-position LFO Mode switch determines how each filter tracks with the LFOs. When the switch is in the Single position, LFO 1 controls all three filters. In Dual mode, LFO 1 controls Filter 1, and LFO 2 controls Filters 2 and 3. In Dual Reverse mode, LFO 1 sweeps Filter 1 upward, while LFO 2

MOTM Specifications

VCO Frequency Range	0.2 Hz–38 kHz
LFO Frequency Range	1 cycle/20 minutes–2.8 kHz
VCO Frequency Drift (24 hours)	<0.5 Hz
VCO Pitch Tracking Accuracy	≤0.1% (25 Hz–3.2 kHz)
VCO/LFO Output	–5 to +5V peak-to-peak
CV Range	–7 to +7V
Gate Input Range	2.5–12V (positive)
Dynamic Range	>90 dB
S/N Ratio	>90 dB
Power Supply Voltage	±15
VDC Power Consumption/Module	± 30 mA typical
Module Dimensions (single width)	8.75" (H) x 1.75" (W) x 5" (D)

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sweeps Filter 2 upward and Filter 3 downward.

I wish the module provided an output for each LFO. It would give me two free LFOs to use elsewhere. I also wish I could control the filters' resonance. The circuit's topology doesn't allow for that, but that's okay. As it is, the MOTM-410 sounds fresh and unique; I would need five simpler modules to get the speechlike formant shifts this module provides.

SOUNDS AND PROCESSING

MOTM-101. The MOTM-101 Noise/S&H module performs several duties at once. The noise and the sample-and-hold sections are internally patched together, but you can use them independently, as well.

White, pink, and randomized noise are available simultaneously from the lower row of jacks. The module also includes a randomized vibrato output, created from band-filtered noise centered at 7 Hz. A vibrato control adjusts the filter's Q, which creates a random signal, more or less. Although you can't adjust the frequency of this random LFO, it's useful as it stands.

The sample-and-hold part of the module has an internal clock and a rate control. It can also lock to an external pulse, and a control knob scales the output. A unique feature of the MOTM-101 is the Track/Hold switch. In the Hold position, the module steps to a new voltage every time it gets a trigger. In the Track position, it mirrors the moving input voltage while the gate is low (below 1.5V) but freezes that output when the gate is high. It's like the child's game of red light, green light played with voltages.

MOTM-110. Another dual-function module, the MOTM-110 VCA/Ring Modulator has a simple but high-quality VCA with audio input and output jacks, a gain control, CV input, and a corresponding sensitivity knob. Like the other modules, the MOTM-110 sounds impeccably clean and punchy. The gain knob boosts the signal above unity,

PRODUCT SUMMARY

Synthesis Technology
MOTM Modular Analog Synthesizer
$89–$349 per module

FEATURES	4.0
EASE OF USE	3.5
AUDIO QUALITY	5.0
VALUE	4.5

RATING PRODUCTS FROM 1 TO 5

PROS: Unique sound. Audiophile quality. Great thermal stability. Heavy-duty construction. Can be customized and expanded. Resists obsolescence.

CONS: Large, heavy, and expensive compared to preconfigured and software synths. Envelope generator doesn't have an LED indicator.

Manufacturer

Synthesis Technology/MOTM
tel. (888) 818-6686
e-mail: synth1@airmail.net
Web: www.synthtech.com

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FIG. 3: Looking closer at the MOTM-300 front panel, you can see the ¼-inch jacks conveniently placed at the bottom.

which lets you use negative control voltages for attenuation. With two VCAs, you can crossfade between timbres—a nice touch, though confusing at first.

The Ring Modulator works independently from the VCA, with separate carrier and modulation inputs and associated level knobs (marked X and Y), modulation output, and two AC/DC switches to pass or remove DC offsets. An extra control lets you unbalance the modulator, which has the sonic effect of adding grunge and saturation to the sound while letting the carrier signal bleed through. That adds an extra timbre-shaping twist to the familiar clangorous tones of ring modulation.

MOTM-800. The MOTM-800 envelope generator is the simplest MOTM module. Just as expected, you have front panel control over attack, decay, and release times, along with sustain level. The module has gate and trigger inputs and positive and inverted outputs. The time values are on the fast side and allow some snappy attack transients. Techno mavens will love it.

I wish the MOTM-800 had an LED so I could see when it triggers. I also think Synthesis Technology needs a full-featured envelope module with voltage-controllable durations. In the meantime, the MOTM-800 is affordable and functional.

MOTM-120. The MOTM-120 Sub-Octave Multiplexer includes two digital counters, a digital multiplexer, and four digital ring modulators. The module converts an input signal to a square wave and then divides the frequency by 2, 4, 8, and 16. You can mix the octaves with the original, but that's only the beginning.

The MOTM-120 works with both audio and ultra-low frequencies, and it lets you modulate the first input with a second input. In the Cross mode, the module acts as a ring modulator when A and B inputs are audio. If A is audio and B is an LFO, you hear rhythmic, sequenced pulsing among the four suboctave pitches. When a clock is sent to both A and B inputs, strange syncopation occurs. Things can get wacky fast with this module.

MOTM-700. The MOTM-700 Dual 2:1 Voltage Controller Router allows you to assign one input to two outputs (or vice versa) and use control voltages to flip between the two. The module has two routing sections that can be used independently. The MOTM-700 accepts any frequency, so you can route control voltages or audio signals. You can even force the switches to flip at certain audio frequencies, which creates composite waveforms with some bizarre and messy modulation artifacts. The MOTM-700, by design, invites creative misuse.

POWER AND VERSATILITY

The two remaining MOTM modules are utilitarian in design. The MOTM-900 power supply uses a medical-grade, low-ripple power block with enough juice to drive about 30 modules. The power cord plugs into the front of the power supply module, which makes cabinet mounting easier. The power switch has a mechanical indicator for the on position, though no lighted switch.

For systems using more than 30 modules, Synthesis Technology sells a larger power block that doesn't quite fit behind the front panel. Some drilling and soldering is necessary to mount it into a cabinet.

The MOTM-940 Patch Panel includes eight rows of three jack types—¼-inch, 3.5 mm, and banana jacks—so you can integrate voltage and audio signals from modular synths (such as Buchla, Serge, Modcan, and Wiard) or other instruments that don't use ¼-inch cords. The 3.5 mm jacks accept ⅛-inch plugs, and each of the banana jacks has a second banana jack next to it for grounding purposes.

Also, the bottom of the MOTM-940 panel includes two handy 4-way mults using ¼-inch jacks. Each mult group gives you as many as three output signals from one input. The MOTM-940 is the only panel that doesn't come in kit form. However, it is a panel that you could use in other parts of the studio, wherever there's a need to convert from one plug type to another.

A DREAM COME TRUE

Like most analog modular synthesizer manufacturers, Synthesis Technology is a small company, and its size has certain advantages and disadvantages. On the one hand, it means that a certain clarity permeates the modules and few compromises creep into the design. On the other hand, you might experience delays when ordering modules, and the price won't drop if a large quantity of modules are produced. The MOTM system is a boutique product aimed at open-minded electronic musicians, audiophile sound designers, and serious DIY tweakers. Such a select market doesn't lend itself to mass production, and I suspect these modules will remain a specialty item.

Yet for the high standards they meet, the Synthesis Technology MOTM modules are quite a value. Each module costs less than a comparable Moog or Buchla module did in 1970, which makes this system a bargain after 30 years of inflation. In addition, Synthesis Technology has updated seemingly retro analog gear into an audiophile 21st-century sound designer's dream. The MOTM system is the real deal. There's nothing virtual about it.

Robert Rich *has released more than 20 albums and has created an* Acid *loop library among other stuff. Some of it's pretty strange. His Web site is www.amoeba.com.*

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JOHNSON AMPLIFICATION

J-STATION

A tone toolbox for the recording guitarist.

By Steve Broderson

Adding to a line of products that includes the Millennium, Marquis, and Mirage series guitar amplifiers—each of which uses physical modeling to simulate a variety of classic amps—the folks at Johnson Amplification have now unveiled the J-Station modeling preamp. The J-Station is a standalone tabletop unit featuring models of 12 classic guitar amps, 3 bass amps, and 2 acoustic guitars. It also features Cabinet Imaging technology, which Johnson Amplification says re-creates the characteristics of 12 different amp cabinet/speaker combos. The J-Station's presets allow for extensive editing, which makes it possible to run, say, a Marshall JCM900 through a Fender Twin cabinet—without consulting a custom shop. The rig also contains a respectable number of effects for further tone tweaking.

Although you might find the J-Station in your retailer's stomp box and multi-effects section, its features make it especially suited for the personal-studio recording guitarist. That's not to say it doesn't perform as an effects unit; it does and very well. But the J-Station shines in its ability to funnel giant-size guitar tones straight into a sound card or hard disk recorder—without waking the neighbors.

THE ONCE-OVER

About the size of a drum machine, the J-station is a well-built unit with a hefty metal chassis. The J-Station's control surface provides 11 knobs, 7 buttons, a 2-digit numeric display, and an LED tuner indicator. The six large knobs control the usual amp parameters, labeled, from left to right, Gain, Treble, Mid, Bass, Level, and Master Volume.

A separate knob on the right side of the panel lets you select among 17 amp models: J Crunch, J Solo, J Clean, Boutique, Rectified, Brit Stack, Brit Combo, Black Face, Boat Back, Flat Top, Hot Rod, Tweed, Blues, Fuzz, Modern, British, and Rock. The model name lights up when selected. An 18th selector position, labeled More, lets you gain access to a bank in which future software-updatable amp models (developed by Johnson) can be stored—a nice bit of obsolescence protection. On the unit I tested, the More bank contained models of a '78 Marshall Master Volume and a Hiwatt Custom 50.

The J-Station's effects section has a Data knob for selecting presets and adjusting parameter values, as well as three smaller knobs for dialing in, as labeled, Effects/Speed, Delay/Fback, and Reverb. A Shift button gives you access to the alternate functions. The other six buttons are labeled Effect Type, Comp (compressor), Tap-It, Gate, Tuner, and Store. The compressor, delay, and noise gate are always available, but only one of seven effects—chorus, flanger, phaser, tremolo, rotary speaker, auto wah, and pitch shift/detune—can be used at a time. The selected effect's name lights up in the matrix.

You can gain access to effects parameters at two levels: Top Level and Deep Level. Obtaining access to parameters at the top level is straightforward—simply select the effect and turn any of the three edit knobs. As the name suggests, Deep Level editing is more involved.

The J-Station ships with 30 factory settings and locations for 30 user settings. As a bonus, Johnson's Web site features a Patch Library where users can both upload and download user-created patches. Fortunately, saving user presets on the J-Station is a quick and easy one-click process.

The Johnson Amplification J-Station provides models of 12 different guitar amps (and a slot for extra software-updatable models developed by Johnson), 3 bass guitar amps, and 2 acoustic guitars. Its versatile effects section can be edited at the Top Level and Deep Level.

GOING DEEPER

Deep Level editing requires selecting a preset (a preset comprises amp model, cabinet, effects parameters, and levels) and holding down the Shift key for three seconds. The display then shows a number between 0 and 39, each of which corresponds to a parameter listed in a chart in the manual. The Effect Type and Tap-It keys now function as increment and decrement buttons, allowing selection of parameter numbers. After you select a parameter, use the Data knob to change its value. Deep Level allows fine editing of almost every

aspect of the sound, including EQ, compression, effects, and cabinet emulation (which always comes at the end of the line). The effects' order can also be changed somewhat, and some global changes are available here, as well.

For all its wonders, though, Deep Level editing is a bit tedious thanks to the 2-digit display. However, on the plus side, the top-level knobs are still active, so you can readily make quick adjustments to the parameters those knobs control, even when you're "deep down."

The J-Station's rear panel provides ¼-inch stereo analog outputs, an S/PDIF digital output on an RCA jack, a MIDI In jack that you can switch to handle the Johnson J8 Foot Controller, a MIDI Out/Thru jack, and a ¼-inch footswitch jack for the Johnson J3 Foot Controller.

The J-Station also offers a wonderful extra for Windows users: a downloadable editor/librarian called *J-Edit* that speeds up the editing process with an intuitive point-and-click layout (see **Fig. 1**). All presets are shown in a list, and *J-Edit* also has graphic representations of all editable parameters, including toggle buttons for global settings such as digital dry output and cabinet emulation. After editing a preset, you can name it, save a copy to your computer, and dump the whole thing into the J-Station using MIDI System Exclusive messages. *J-Edit* shows the effects' order at the bottom of the screen, and you can even change your presets' order in the list, which is very handy for live gigs. *J-Edit* makes the J-Station an infinitely more fun and usable piece of gear. Hear that sound? Those are my knees hitting the floor as I humbly ask Johnson Amplification for a Mac version of *J-Edit*. The buzz from Johnson is that Mac support is on the way.

You can also edit presets and effects using MIDI. In addition, the standard MIDI In port doubles as the port to Johnson's J8 foot controller with the flick of a switch. (Also, there is an additional jack for the more limited J3 footswitch.) However, the silk-screened graphic on the switch is fairly confusing—what appears to be the MIDI In setting is actually the J8 setting and vice versa.

HARD DRIVIN' AMP

The J-Station's primary bragging rights proceed from its digital-recording capabilities; indeed, this is the only unit in its class that offers an S/PDIF digital output. Granted, guitar-modeling plug-ins are available for many popular hard disk recording applications; however, the J-station has some definite advantages, in particular the speed and familiarity of using actual knobs and buttons instead of menus and mouse-clicks. Also, anyone who has tracked to hard disk from a plug-in amp modeler has probably run into digital latency: that small but annoying delay between what you play and what you hear. With the J-Station, you can avoid latency by monitoring the unit's analog output or by using headphones.

When tracking with the J-Station, you have the option of generating mono or stereo digital output, both at 24 bit, 44.1 kHz. Make sure your digital-output level is nice and hot, though; some of the J-Station presets didn't generate enough bits into my audio card, which resulted in muddy, weak audio with hardly any pick attack or depth. Fortunately, that problem is easily remedied by going into the Utility menu and cranking up the digital output.

MODEL CITIZEN

Now that I've covered what the J-Station can do, I'll describe how the unit sounds. I tread lightly here because preferences in guitar tone are so subjective. Fortunately, you can get a preview of the J-Station's sonic palette at Johnson Amplification's Web site (www.johnson-amp.com/jstation.htm). There you'll find a virtual tour of the J-Station, including mouse-over sound bites from each of the model presets (very cool), the latest version of *J-Edit* software, and some gee-whiz documents about Johnson's cabinet-imaging technology. Although results with the unit may vary depending on factors such as your guitar, strings, pickups, and playing style, the Web site does provide a well-executed first listen to the product.

I used two different guitars to test the J-Station: an Epiphone Les Paul and a Fender American Standard Telecaster. I patched the J-Station's analog outputs straight into an Alesis RA-100 reference amp and used Alesis Monitor One

FIG. 1: *J-Edit*'s intuitive point-and-click interface greatly simplifies the J-Station editing process.

close-field monitors. I connected the J-Station's digital output into my Mac's Logic Audio card.

I checked out each preset but soon found myself tweaking them because many had heavy gain and effect settings that I was eager to subdue. To compare the raw models as fairly as possible, I created a user patch with no effects and then switched only among the amp models.

The first three models—J Crunch, J Solo, and J Clean—are imported from Johnson's combo-amp line. J Crunch and J Solo sound similar when first dialed up, with slight differences in the midrange and distortion overtones. Boutique and Rectified are high-gain models based on Matchless and Mesa Boogie amps, respectively. Each has a generous scoop in the midrange.

▼

Hear that sound? Those are my knees hitting the floor.

The mids are well accounted for, however, in the next two models: Brit Stack and Brit Combo. Those two models are tonally well balanced, with plenty of sparkle and bite. The Stack model loved my Les Paul, for which it provided gain and overtones galore. Black Face also proved a favorite of mine, thanks to its big body and punchiness. The Tele, with lipstick (neck position) pickup selected, was the hero here. When I used the Tele and Black Face in conjunction with one of the spring-reverb models, I could almost smell the tubes warming up.

Next in line are the two acoustic guitar models: Boat Back and Flat Top. Although not completely convincing at turning my Tele into a miked acoustic, the acoustic models provided usable sounds, both with distinctive high ends. Using the pickups closest to the neck helped the simulations sound more realistic.

Hot Rod is a straightforward combo model that really accentuated my Tele's biting midrange. Tweed has a smallish, compact sound with the personality of a 12-inch-speaker combo amp. Blues is another straightforward model, and Fuzz, with its massive overdrive and sound of worn-out tubes begging for mercy, channels the spirit of Hendrix.

Apart from the Black Face model, the clean amp sounds didn't reveal much personality until the pregain was pumped up a bit. Once I added some bite with the pregain, my guitar sounded more as if it were coming out of a cabinet rather than directly off the board.

As for the Cabinet Emulator, turning it off made for harsh, transistor-like sounds with far too much high end, so I ended up leaving it on for all of the auditions. Overall, it imparted a focused, natural tone to the models and, for the most part, enhanced the sense of an amp being miked.

J-Station Specifications

Frequency Response*	25 Hz–20 kHz (+1, –3 dB)
S/N Ratio*	>101 dBA
THD*	<0.02% (@ 1 kHz)
Programs	30 factory/30 user
A/D/A Converters	24-bit
Sampling Frequency	44.1 kHz
Analog Inputs	(1) ¼" unbalanced guitar
Analog Outputs	(2) ¼" unbalanced; (1) ⅛" stereo headphone
Digital Output	(1) RCA, stereo S/PDIF
Other Ports	MIDI In, Out/Thru
Dimensions	2.25" (H) × 10.5" (W) × 6.5" (D)
Weight	2.1 lbs.

* = all effects disabled

PRODUCT SUMMARY

Johnson Amplification
J-Station
guitar-amp modeler
$449

FEATURES	4.0
AUDIO QUALITY	4.0
EASE OF USE	4.0
VALUE	4.0

RATING PRODUCTS FROM 1 TO 5

PROS: Wide tonal variety. Digital output. User-friendly control surface. Excellent effects section. Cabinet emulation.

CONS: No editor/librarian for Macintosh. Slight tracking delay in Pitch Shift.

Manufacturer

Johnson Amplification
tel. (801) 566-8800
Web www.johnson-amp.com

Next on the dial are three bass models: Modern (based on a Trace Elliot bass amp), British (based on a Vox Top Boost), and Rock (based on Ampeg's venerable SVT). Each was warm and imparted a nice, round sound to my Fender P-Bass. The exception was the Rock model, which stood out because of the edge and growl it injected. That sound will be on my next recording, I can promise you.

EQ AND EFFECTS

The EQ for each amp model is based on what the actual amplifier provides, so the range is a bit limited. Indeed, small adjustments made almost imperceptible changes. I found that I could beef up the low end by really cranking the Bass knob or take some harshness out by backing off the High, but each model's overall sonic characteristics remained pretty much intact. Considering Johnson included effects in the J-Station that are not available from the individual amps the unit replicates, the company might also have included more extensive EQ offerings.

I definitely like the J-Station's effects section. The push-button ease of engaging the compressor and gate is wonderful. Compression is smooth and

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transparent with lots of sustain and no pumping and breathing. The gate does just what it should, silently removing noise without noticeable gating or false triggering. The time-based effects are all pleasing as well. Pitch Shift was the only effect that left a bit to be desired because of a slight delay in its tracking; however, that complaint is minor.

Delay and Reverb levels are readily set by knobs, and both offer a nice variety of analog and digital models. I was especially impressed with the spring reverbs, which are convincing and a lot of fun to mess with.

STAGE MEISTER

I couldn't resist trying the J-Station through my trusty Peavey Classic 50 tube amp. To do so, I first turned off the cabinet emulation. (That is done, thankfully, using a one-step global setting accessible through the Utility menu.) Next I ran the left output directly into the power-amp input, bypassing the Classic 50's preamp (after making sure the J-Station's Master Output knob was turned all the way down, of course). As before, I tweaked the factory presets a bit using both Top-Level and Deep-Level editing.

In combination with my tube amp, the J-Station models were noticeably warmer sounding and had less digital edge. I was again impressed by the effects, which were solid performers in this application. The J-Station gave my stage amp several new personalities. Now all I need is a MIDI pedal board!

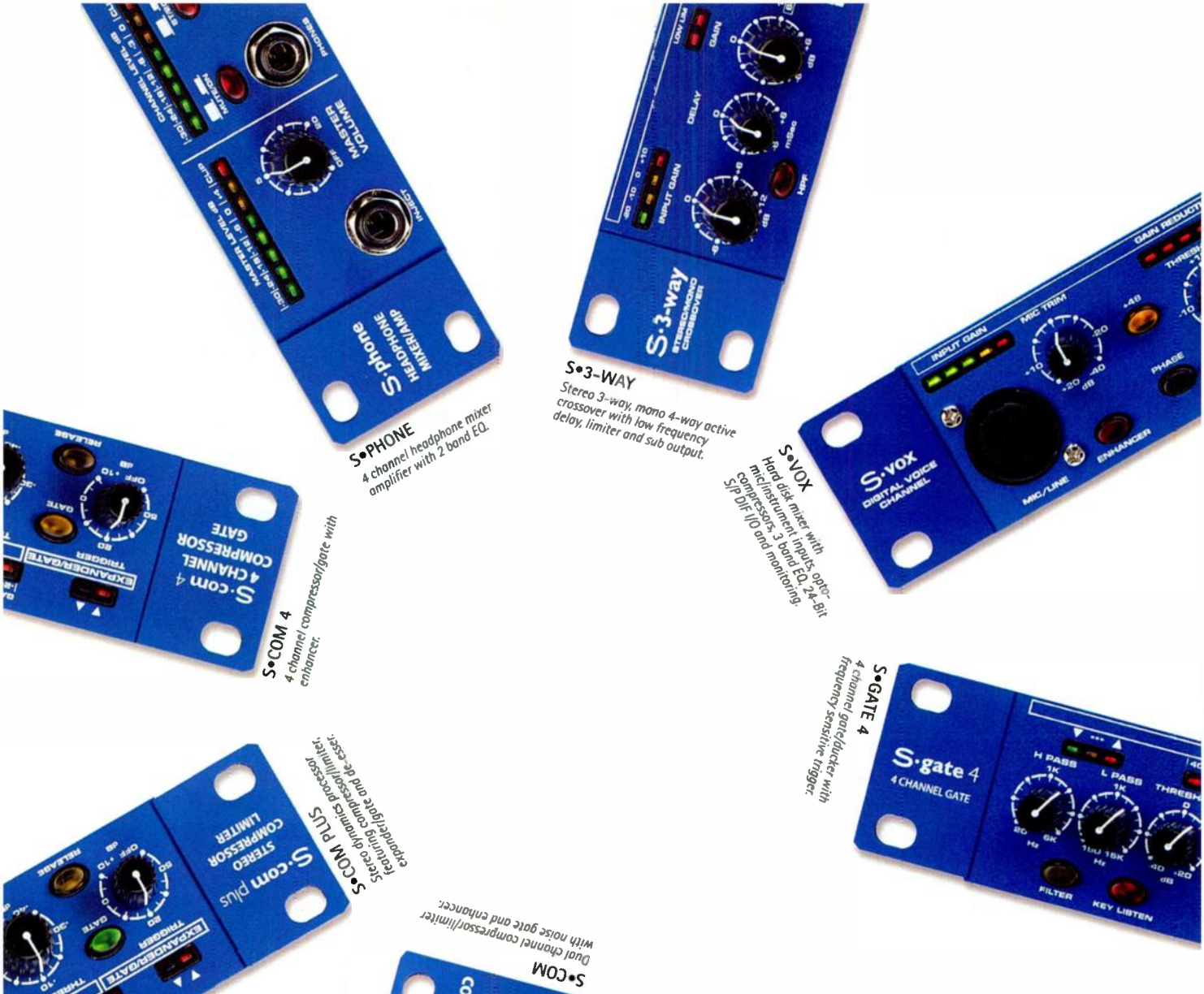
J-Edit makes the J-Station infinitely more fun and usable.

SOLID DEAL

The J-Station is a good-sounding, well-built, and easy-to-use unit. The effects are sonically right on and a breeze to tweak. The *J-Edit* software is an absolute gem. Indeed, I am adding the J-Station to my arsenal both for recording and live performance.

The most difficult thing about the J-Station may be how to categorize it. From what I've seen in catalogs and stores, the unit seems to get lumped into the stompbox and multi-effects category. Retailers would do better to put the unit next to the hard disk recorders. The people who should be most excited about the J-Station are personal-studio guitarists who don't have a vintage amps collection at their disposal—and those whose neighbors won't tolerate wailing guitars at all hours of the night.

Steve Broderson *is a copywriter for an advertising agency in Lexington, Kentucky. His recently formed Studio 246 has produced original music for the University of Kentucky, Ale 8-1 (a local soft drink), and iHigh.com.*



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M O T U

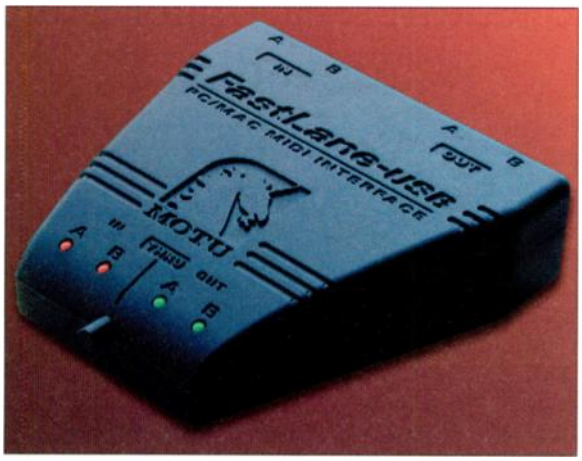
FASTLANE-USB

A modest MIDI interface with a reasonable price tag.

By Geary Yelton

If you purchased a new computer—whether Mac or PC—in the past few years, chances are good it has a Universal Serial Bus (USB) port or two. Because you're reading this, chances are also good you have at least one MIDI device you'd like to connect to your computer. To make that connection, you need a MIDI interface.

The FastLane-USB is a 2-In/2-Out MIDI interface from Mark of the Unicorn (MOTU). It is the successor to the original FastLane, a 16-channel serial interface with one MIDI In and three MIDI Outs designed for the pre-USB Mac. The FastLane-USB has two distinct signal paths, so it's capable of handling 32 channels of MIDI data. It requires no external power other than what the USB cable supplies directly (a 2-meter USB cable is included). The unit's exterior is rigid plastic and comes in black, as well as a variety of colors to match the Apple iMac.



If your computer has a USB port and your MIDI needs are modest, MOTU's FastLane-USB will get you into the game.

BASIC AND BEYOND

The FastLane-USB offers basic features with a minimum of bells and whistles. It has no SMPTE or MIDI Time Code synchronization capabilities, but it does support MOTU's new hardware-based MIDI Time Stamping (MTS). MOTU claims this provides timing accuracy of 1 ms or better when used with *Digital Performer* 2.7. With MOTU's more expensive rack-mountable interfaces, such as the MIDI Express-USB and MIDI Timepiece AV, MTS increases timing accuracy to 0.33 ms.

The interface features a Thru button to let MIDI data pass through the unit when there is no USB connection. The button allows you to play MIDI devices from your controller without rerouting MIDI cables, even when your computer is turned off. However, each input (labeled A and B) is matched to a specific output (A or B). And if you choose to use one instrument to play another without going through the computer, they must be connected to corresponding ports on the interface. There is no way to reroute data from input A to output B or vice versa. The Thru button is never mentioned in the *QuickStart Guide,* which is the only documentation that is included with the FastLane-USB.

Four LEDs on the FastLane-USB monitor MIDI activity. Two red LEDs flash when data passes through the MIDI In ports, and two green LEDs flash when data passes through the MIDI Out ports. A glance at those LEDs will tell you if you have an incorrect connection or a dead MIDI cable. When the Thru button is depressed, however, the LEDs stop working because the interface gets its power from the USB, and pressing the button disconnects the interface from the bus. The LEDs also go unmentioned in the documentation.

PRODUCT SUMMARY

Mark of the Unicorn
FastLane-USB (Mac/Win)
MIDI interface
$79

FEATURES	3.0
EASE OF USE	4.0
DOCUMENTATION	2.0
VALUE	5.0

RATING PRODUCTS FROM 1 TO 5

PROS: Economical. Bus-powered. MIDI activity LEDs. Works with FreeMIDI and OMS. Compatible with PCs and Macs.

CONS: No synchronization. Thru button of limited use.

Manufacturer

Mark of the Unicorn
tel. (617) 576-2760
e-mail info@motu.com
Web www.motu.com

DRIVER, TAKE ME HOME

A CD-ROM includes the software required for the FastLane-USB. The disc I received had no Windows driver, but MOTU says the FastLane-USB now ships with a cross-platform CD-ROM that includes drivers for Windows 98 and Windows ME as well as Mac OS. You can also download drivers from MOTU's Web site (www.motu.com). For this review, I used the interface with a Power Mac G4 running OS 9.0.4.

I encountered some minor problems when I tried to install the FastLane-USB and its drivers according to the instructions, which indicate that first you should connect the interface to the computer and MIDI instruments to the interface, then install the software. The manual says to start the Mac with extensions off and then insert the CD-ROM. Right away I knew the Mac wouldn't recognize a CD-ROM without the Apple CD/DVD Driver turned on.

Nonetheless, I gave it a try to see if anything unexpected would happen, and it did. A few times I received an address error message, and the computer froze—but not on every try. I've been told this kind of freeze on start-up is possible with

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any USB device that is connected when its driver is disabled or missing. When I managed to restart without freezing, a dialog box appeared to inform me that my computer had no driver for the interface and offered to pull one off the Web.

At that point I disregarded the instructions and restarted the Mac with the CD/DVD extension turned on and the FastLane-USB disconnected. Then I ran the Installer and reconnected the interface. I didn't have any further problems.

MOTU says it will revise the manual to instruct Mac users to keep the base set of extensions on and install the driver before connecting the interface.

I opened FreeMIDI Setup and let it find the newly installed interface (the installer will install or update FreeMIDI on your machine as necessary). With the installation complete, I restarted my computer. When I opened *Digital Performer*, everything worked perfectly.

If you install FreeMIDI on a computer that also has Open Music System (OMS), you can select either one; choosing OMS means the MOTU OMS Driver is installed. You can also toggle between FreeMIDI and OMS in the FreeMIDI Setup utility. To use both MIDI systems separately, tell FreeMIDI to allow other applications, and instruct OMS not to run MIDI in the background.

LIFE IN THE FASTLANE

Assuming your computer has a USB port and that 32 channels are enough for your MIDI setup, a handful of MIDI interfaces are available; at $79, the FastLane-USB is the least expensive one you'll find. If your needs grow and you require more MIDI channels in the future, you can connect several FastLane-USBs to a USB hub. Even if you need 64 MIDI channels, two FastLane-USBs are cheaper than any 4×4 interface in stores now. A FastLane may also be the best solution if you already have a multiport MIDI interface and just need a couple more Ins and Outs.

FastLane-USB supports MOTU's new hardware-based MIDI Time Stamping.

MOTU has long been a leading maker of MIDI interfaces for the Mac, and recently it has been meeting the needs of PC users, as well. The FastLane-USB MIDI Interface is a solid product, built for years of dependable service (as long as you don't step on it). If you have a Mac G3, G4, iMac, or iBook—or a PC running Windows 98 or Windows ME—and you can get by with 32 MIDI channels and no synchronization, the FastLane-USB will get the job done with a minimum of trouble and expense.

Geary Yelton *recently married the love of his life (Pam), landed the job of a lifetime (associate editor of* **EM***), and escaped the harsh demands of the corporate environment.*

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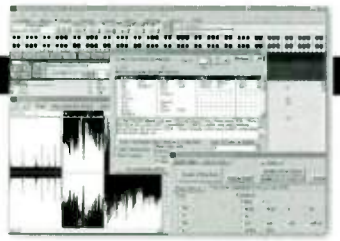

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This interactive program is designed to help you learn to solo like a professional jazz guitarist. Musicians studying improvisation frequently learn by copying other soloists' performances. Since guitar is a "visual instrument", it is often easier to learn to play by watching the fretboard than by reading the music from a staff. The on-screen guitar fretboard in The Master Jazz Guitar Solos PAK shows you exactly which notes and chords are being played, in real time as you hear the music. We've even included "guide notes" for common fretboard positions that are displayed on the on-screen guitar fretboard to help you learn the note names, string and fret positions. The Master Jazz Guitar Solos PAK contains over 150 songs designed to illustrate basic skills to the beginning guitarist while enhancing the skills of more advanced guitarists.

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TC WORKS

TC VOICETOOLS 1.0 (MAC/WIN NT)

An affordable, all-in-one vocal-processing solution.

By Chris Gill

One of the main advantages big studios always have over home and project studios is that they are better equipped for recording vocals. The right equipment for recording professional-quality vocals—microphones, mic preamps, compressors, equalizers, gates, de-essers, and reverbs—usually requires a hefty five-figure investment. Even then, you still need a decent-sounding room to record in. However, thanks to the virtual-studio revolution and vocal-processing plug-ins such as *TC VoiceTools*, the playing field is getting more level every day.

The *TC VoiceTools* TDM bundle provides a variety of processors commonly used for vocal-recording applications in a single package. Actually, the software consists of two separate plug-ins. One plug-in is *TC VoiceStrip*, an all-in-one channel-strip processor offering a compressor, EQ, de-esser, gate, and low-frequency cut; the second is *TC Intonator*, a pitch-correction device. *TC VoiceTools* is compatible with any Power Mac approved for use with a Digidesign Pro Tools 24/Mix system running *Pro Tools* 4.x or 5.x software. I tested the package with a Power Mac 9600/350 with 320 MB of RAM, a Pro Tools 24/MixPlus system, and *Pro Tools* 5.0 software.

Minimum System Requirements

TC VoiceTools

Power Mac; Pro Tools 24/Mix; Pro Tools 4.x/5.x

CRUISING THE STRIP

Replacing a whole rack's worth of processors, *TC VoiceStrip* is the bundle's true workhorse. TC adopted a vintage design concept, providing the look and sound of classic processors with relatively limited controls that are optimized for vocal applications (see **Fig. 1**). Although the plug-in eats up 50 percent of a DSP chip on a Mix card, its channel-strip design provides efficient use of resources because it supplies six different effects on a single insert and doesn't require any RAM. Should you decide that you prefer a certain equalizer or compressor, for example, you can bypass those effects individually.

TC VoiceStrip's 3-band EQ section is tuned especially to the frequencies within the normal vocalist's range, which lets you make reasonable, but not overly drastic, tonal changes. Each band features controls to boost or cut gain up to 18 dB. The low band features a shelving filter with adjustable frequency from 100 to 350 Hz. The mid band's filter is bell shaped, and the frequency range spans 700 Hz to 7 kHz. The high band consists of a shelving filter with a fixed 2.5 kHz frequency. This section also includes a SoftSat switch that provides a subtle, analog-like, soft-clipping harmonic distortion effect. Even at extreme settings the effect prevents hard clipping, offering smooth tones with none of the harsh overtones encountered with digital distortion.

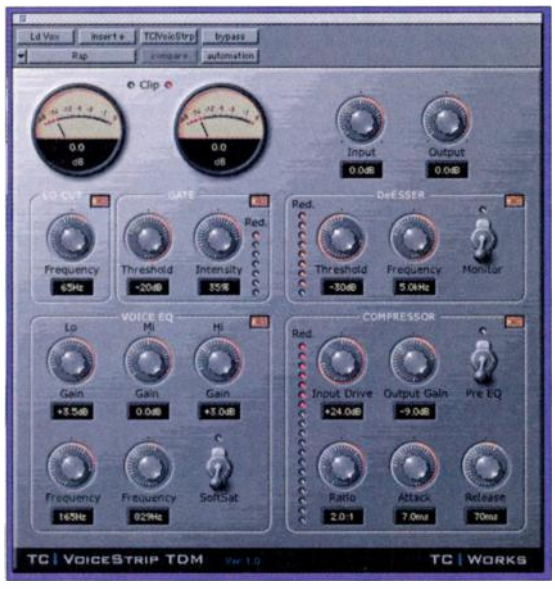

FIG. 1: *TC VoiceStrip* has a vintage-style interface and provides five individual effects modules, including a low-frequency cut, gate, de-esser, EQ with soft saturation, and compressor.

Like the EQ section, the compressor is designed with vocal applications in mind. The feedback-type compressor's processor gets its sidechain input from the output. Controls include Input Drive (24 dB boost or cut—the compressor's threshold is fixed at –24 dB, so the harder you drive the input, the more compression you get), Output Gain, a Pre EQ switch that lets you change the routing order of the compressor and EQ sections, a Ratio control with a 1:1 to 64:1 range, Attack (0.1 to 50 ms), and Release (50 ms to 2 seconds). The compressor maintains a smooth, natural sound no matter where the controls are set, which avoids the unnatural breathing effects often encountered with overcompressed signals.

The de-esser's controls consist of a Threshold knob with a 0 to –30 dB range, a Frequency knob with a 1 kHz to 10 kHz range, and a Monitor switch for listening to the sidechain signal to help you determine the desired cutoff frequency. The de-esser's operation is level independent, so sibilance is always removed—even when the recorded material's level changes.

A standalone gate section offers Threshold (0 to –70 dB) and Intensity (0 to 100 percent) knobs. An independent low-cut filter lets you adjust the cutoff frequency in 5 Hz increments from 60 to 120 Hz. You can also select a DC removal setting optimized for eliminating DC noise.

A NICE VU

TC VoiceStrip's master controls consist of round, porthole-style input and output VU meters, input and output clipping indicators, and Input and Output level controls. In Stereo mode, an In/Out switch appears between the two VU meters, so you can use both meters to monitor input or output levels. Left and Right level controls are provided for the input and output sections.

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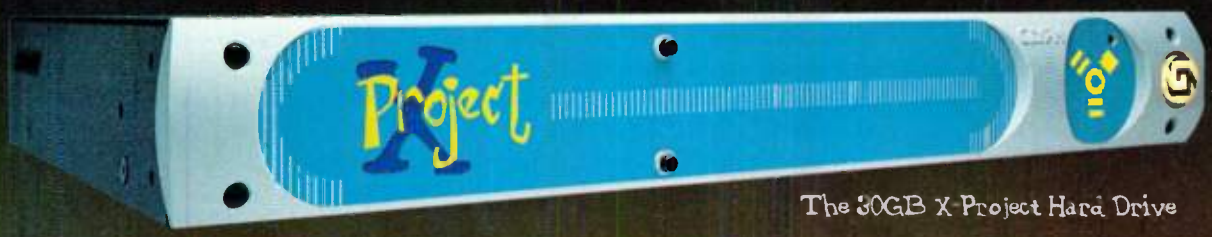

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When the plug-in is bypassed, the control panel turns entirely gray except for the clipping indicators, making it easy to determine at a glance whether the processor is engaged (see **Fig. 2**).

TC Works included 21 presets that provide users with a good idea of *TC VoiceStrip*'s capabilities. Most of the settings are quite tasteful, and you can apply many of them to preexisting recordings with only a few minor tweaks to get good-sounding results. Even the more exotic settings, such as the booming, in-your-face Rap preset or the "telephone voice" ThinFX preset, sound more musical than gimmicky. I applied the plug-in to a variety of prerecorded source material—including a multitrack master of a Lisa Loeb song, a semiprofessional demo by a rock band with a male vocalist, and some a cappella vocals lifted from a hip-hop single—and the plug-in let me give the vocals a smooth-sounding, professional polish in all instances. The analog-style processing rounds off the rough edges that often accompany raw digital recordings, providing a more natural sound quality.

CORRECTIONAL FACILITIES

Whereas *TC VoiceStrip* is the kind of processor that you would use often on entire tracks, its counterpart, *TC Intonator*, is an effect that most users would probably apply sparingly. *TC Intonator* is a pitch-correction plug-in that fixes intonation in real time. Although I have yet to encounter a pitch-correction plug-in that can transform a tone-deaf warbler into Maria Callas, *TC Intonator* does a reasonably good job of correcting notes that fall a few cents sharp or flat of the ideal. If you have hopes of turning your 17-year-old sister into the next Christina Aguilera, however, you're probably better off investing in a voice coach.

TC Intonator provides a good selection of controls that supply a considerable level of flexibility to help you achieve the results you desire. The well-designed interface includes a guitar tuner-style bar meter that shows in red how sharp or flat the signal is compared with the desired note (see **Fig. 3**). The amount of applied correction is shown in green. You can select scales and modes from pull-down menus and create your own scales by engaging the 12 buttons on the keyboard display.

Each button shows the detected pitch in red, which is helpful when you want to select existing notes from a performance as reference points. Four knobs allow you to select the amount of pitch correction, manually adjust the pitch, select attack time over a 2 to 2,185 ms range, and set the low-cut filter's frequency from 55 to 265 Hz. The low-cut filter offers standard fixed-frequency and adaptive modes. Other features include an input signal indicator, an input level clipping indicator, and a tuning reference control for fine-tuning the plug-in to a recording.

FIG. 3: *TC Intonator*'s guitar tuner-style bar meter shows how sharp or flat the input signal is compared with a desired note in red and the amount of applied pitch correction in green. In this example, a flat note is perfectly "corrected" to B.

UNNATURAL LAW

With all these control variables, it's also quite easy to get results that sound somewhat unnatural. Although that may not be ideal for a recording of a jazz or opera singer, it could be just the special effect needed to make a pop or dance-music recording stand out. By setting the pitch-correction window to an extreme range such as ±400 cents, turning up the Amount control to 100 percent, and dialing in a fast attack time, you can create a yodeling sound that warbles between two notes, similar to the vocoder effect heard on Cher's "Believe," Kid Rock's "Only God Knows Why," and Daft Punk's "One More Time." In fact, it's hard to avoid that effect whenever the pitch correction window is set to a wide range. A Note Hold button helps eliminate excessive note wavering when that result isn't desired.

Another fun application is to choose an entirely different scale or mode than the original performance. Again, that approach doesn't provide the most natural-sounding results, but it can often warp a vocal performance in a delightful way. For example, I was able to twist a major-scale melody into a bizarre Arabian-scale line that gave the vocalist's performance an exotic feel—just the ticket for an experimental-flavored remix. While *TC Intonator* was

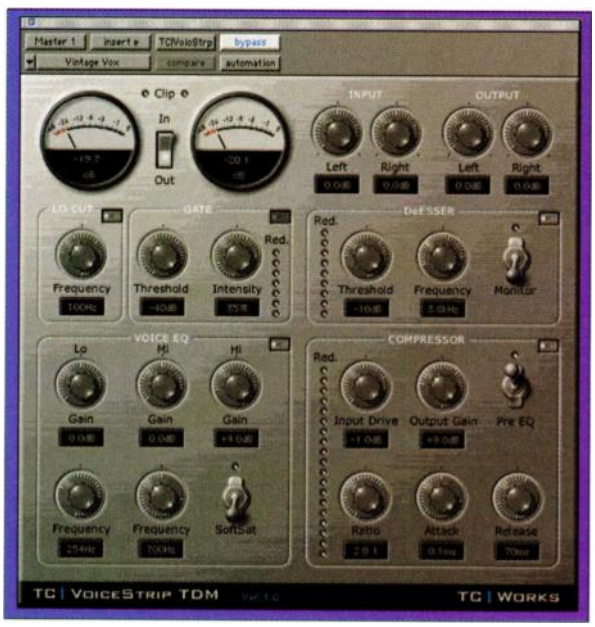

FIG. 2: When you bypass *TC VoiceStrip*, the entire control panel turns gray, which lets you know that the processor is disengaged.

primarily designed for subtle pitch-correction applications, the "wrong" settings provide adventurous producers with some rather rewarding tone-twisting possibilities.

CODED LANGUAGE

Unlike most Pro Tools plug-ins, which use a floppy disk-based key for copy protection, *TC VoiceTools* employs a challenge-and-response code authorization. You can use the plug-ins for 21 days without authorization while you wait to receive your response code, so you don't have to worry about any downtime if you need to use the software immediately. TC Works promises to deliver the code within two days through e-mail or fax (in my case the code arrived less than 24 hours after I submitted the challenge code). Because new Macintosh computers no longer come with floppy disk drives, this approach is a reasonable and welcome alternative. This method also avoids the hassles of losing authorization during a system crash; just type in the response code again, and you're up and running (make sure to write down the code somewhere you can locate it easily).

ALL FOR ONE

TC VoiceTools can provide a significant competitive edge when it comes to recording professional-quality vocals. Although it provides most of the processors necessary for taking the rough edges off of a raw vocal performance, you're still going to need a high-quality mic, a good mic preamp, and a decent reverb to sound even halfway close to the pros. *TC VoiceTools* won't replace a rack full of Fairchild limiters, Urei compressors, and Neve and Pultec EQs—or their software-emulation equivalents for that matter—but it does provide admirable results for a lot less money. Even if you already have outstanding single-application TDM plug-ins, *TC VoiceTools* is still worth consideration because it uses up considerably less DSP. Maybe you can use that DSP-hogging reverb on a few more tracks instead of shelling out more bucks for another Mix Farm card to complete your mix.

While *TC VoiceTools* can greatly improve the sound of your vocal recordings, keep in mind that it is not a cure-all for a poorly recorded vocal. That is one case where the "garbage in, garbage out" condition really applies. To get the best results out of the software, make sure that the performance sounds as good as possible going into your Pro Tools system. Considering the product's affordable price, you might be able to come up with enough money to at least rent a top-drawer Neumann mic and a slick mic preamp to ensure that your performance sounds its best from the start.

If you are really serious about improving the sound of your vocal recordings, give *TC VoiceTools* a try. This is one situation in which an all-in-one solution provides excellent results without any significant compromises.

Chris Gill *is the Los Angeles-based editor of* **EM***'s sister publication,* Remix. *Despite having decent keyboard, guitar, and music-programming chops, he still can't sing worth a damn.*

PRODUCT SUMMARY

TC Works

TC VoiceTools 1.0 (Mac)
voice-processing/pitch-correction plug-in
$699

FEATURES	4.0
EASE OF USE	4.0
AUDIO QUALITY	4.0
VALUE	4.5

RATING PRODUCTS FROM 1 TO 5

PROS: A full range of vocal applications in a single moderately priced package. Elegant interfaces. Analog-like warmth.

CONS: Pitch correction requires care to avoid unnatural results. EQ range may be too narrow for certain applications.

Manufacturer

TC Works/TC Electronic (distributor)
tel. (805) 373-1828
e-mail us@tcworks.de
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ULTRAFUNK

SONITUS:FX 2A (WIN)

A seven-pack of flexible DirectX/VST effects plug-ins.

By Scott R. Garrigus

Sonitus:fx 2a is a group of seven DirectX/VST plug-ins from Norwegian software developer Ultrafunk. Each effect is named after the function it provides: equalizer, modulator, phase, reverb, compressor, wah-wah, and surround. The plug-ins have an uncluttered, intuitive interface and offer an assortment of parameters that you can adjust both numerically (from the keyboard) and graphically (with the mouse). They are available for purchase only online, either separately or as a package, and each comes with a short but informative Help file that doubles as the manual.

I tested the *Sonitus:fx* 2a package on a Pentium III/700 MHz with 256 MB of RAM and Windows 98SE using Cakewalk's *Pro Audio,* Sonic Foundry's *Sound Forge,* Steinberg's *Cubase* 5.0, and a few other host applications. All of the plug-ins performed without problems and were fun and easy to work with.

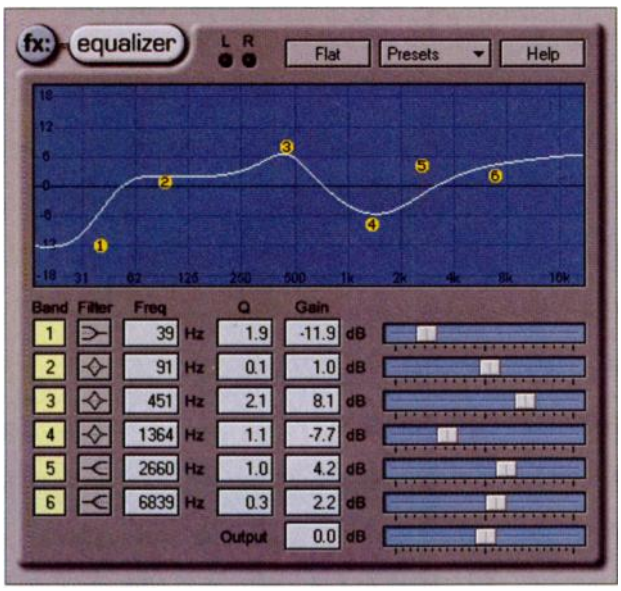

FIG. 1: *Fx:equalizer* provides six bands of parametric EQ, which you can set numerically or graphically.

EQUALIZATION

The *Fx:equalizer* plug-in (see **Fig. 1**) provides six bands of parametric equalization. Controls for each band include filter type, center frequency, Q, and gain. You can choose from five different filter types: Lowpass, Highpass, Peak/Dip, Shelving Low, and Shelving High. The center frequency can be set at up to half the sample rate, with the width of the band controlled by the Q value. The gain of each band can be boosted or attenuated by 18 dB, and you can use the master output gain to normalize the amplitude of the signal after a filter has been applied.

I particularly like the filter graph in *Fx:equalizer.* The six bands are represented by numbered yellow dots. Dragging the dots changes the frequency and gain parameters; double-clicking on a dot toggles its band on and off. You can select multiple dots to modify the parameters in different bands simultaneously while still retaining their original ratios—very nice! *Fx:equalizer* also lets you modify the Q settings by holding down the Shift key while dragging a dot, an unusual but useful function. The amplitude range of the graph can be set to 5, 10, 20, or 40 dB for displaying subtle, precise modulations or drastic EQ effects. The Flat button provides a quick way to reset all the gain parameters to zero.

MODULATION

Fx:modulator can produce various delay-related effects, such as flanging, phasing, chorusing, and doubling. It uses three modes of operation: Flanger mode mixes in one delay line per channel of the stereo signal; Ensemble mode uses three nonsynchronized modulating delays for fatter and more diffuse effects; and String Phaser recreates the "sound of analog sweeping synth-strings" by modulating the signal's phase instead of delaying it.

The plug-in provides a wide range of adjustable parameters, which are Phase, Depth, Delay, Feedback, Mix, Cross Mix, and LFO Rate. You can also set the LFO waveform to Triangle, Sine, Peak, Peak with inverted second half, Twin Peaks, or Peak/Dip. Just as with *Fx:equalizer,* a master gain enables you to normalize the amplitude of your output signal. A simple EQ feature offers Low Cut or High Cut and a single control, which is used to specify its cutoff frequency.

Fx:modulator provides added flexibility by letting you invert the feedback and the mixed (wet) signals either separately or together. Inverted feedback tends to produce a more "hollow" effect, and the inverted wet signal can produce a variety of sounds depending on the source material. In addition, *Fx:modulator* offers a Tape feature that emulates the sound of an analog tape flanger.

COMPRESSOR

Fx:compressor provides variable-knee compression with built-in peak limiting (see **Fig. 2**). The controls include Attack (0 to 400 ms), Release (1 to 4,000 ms), Ratio (0.4:1 to 30:1), Knee (1 to 30 dB), Threshold (–60 to 0 dB), and Gain (–30 to +30 dB). You can adjust the controls by dragging with the mouse or by directly entering numbers. Depending on your settings, the Compression Curve Graph displays the signal level, compression ratio, and knee values over a 60 dB range. The graph also lets you alter the Ratio and Knee values with your mouse. Click anywhere within the graph and drag left or right to change the Knee; drag up or down to change the Ratio.

Fx:compressor includes a built-in peak limiting algorithm as well as a Transient Controlled Release algorithm that automatically adjusts the release time to eliminate fast compression changes

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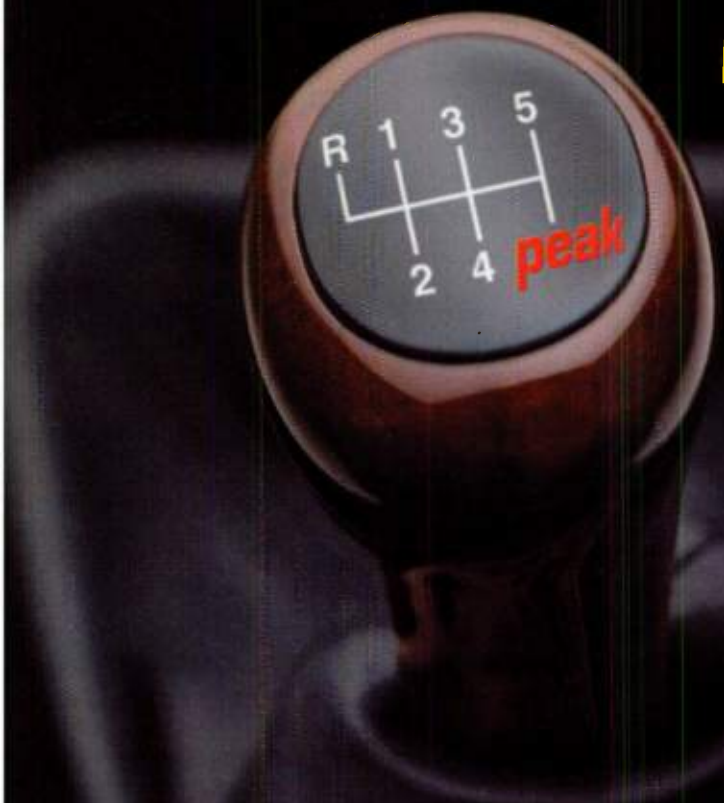

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that can cause "pumping." The most notable aspect of *Fx:compressor* is its two modes of operation: Normal and Vintage. In Normal mode, *Fx:compressor* acts as you would expect, attenuating the gain above the threshold level using the ratio setting. In Vintage mode, however, the compression ratio is gradually reduced to a value of 1:1 as the signal goes above the threshold. This allows the loudest parts of the signal to pass without being compressed as much as the rest of the signal, giving it more warmth and punch, as you would find with a vintage compressor.

WAH-WAH

Fx:wahwah emulates classic wah-wah stomp-box guitar effects. The plug-in operates in three modes: Manual, Automatic, and Triggered. In Manual mode, you control the behavior of the wah envelope with the Wah slider. You can also control the spectrum characteristics of the effect at the high and low points of the wah with the High and Low Filters; both have parameters for Frequency, Q, and Gain. A master gain control is also included.

Automatic mode adds a Tempo control to the features listed previously. The Tempo control sets the wah modulation speed in bpm.

Triggered mode has the same controls as Manual mode and adds Attack, Release, and Threshold settings. In this mode, the wah effect is only applied when the signal reaches the specified Threshold; the Attack and Release parameters control the time it takes to reach the High and Low filter settings respectively. In addition, the Wah slider works differently in this mode. Instead of providing direct control over the wah envelope, it simply determines the range between High and Low Filter settings. If set low, the difference between the settings is smaller; if set high, the difference is larger. You have to hear it to really understand how it works, but it offers great control over the effect.

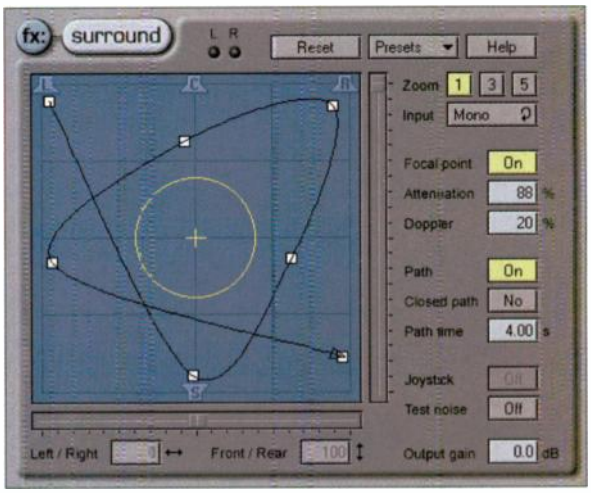

FIG. 3: You can specify your signal's position in the sound field with *Fx:surround*, which encodes surround- and center-channel information into the stereo output.

JUST A PHASE

Fx:phase is the simplest plug-in of the bunch, with only three adjustable parameters, four modes of operation, and a master output gain control. The Phase slider controls the degree of signal shift (±180 degrees), and the selected operation mode determines the way in which the phased signal is mixed into the stereo output. Left/Right mode changes the phase between the left and right stereo channels. In Mid/Side mode, the middle (mono) information is phased in relation to the side (stereo difference) signals. Center/Surround encodes center information in the left channel and surround information in the right channel for use in surround mixes. Surround/Center mode does the opposite, with center information on the right and surround information on the left. The two surround modes automatically apply a phase shift of 90 degrees, which is the optimum setting to avoid phase cancellation when panning in surround sound.

The Width slider determines the size of the stereo image after phase shift is applied: from 0 percent (mono) to 100 percent (stereo) to 200 percent (double the level of side information). The Meter lets you monitor the phase difference between the two channels before (Pre) or after (Post) processing. And the Filter Cycle button lets you choose between an Infinite Impulse Response filter (which is fast and introduces a constant phase shift to the signal) or a Finite Impulse Response filter (which is slower but provides greater accuracy for high frequencies). This plug-in can be used to alter the phase content of your audio signal for a variety of purposes: as an effect, to avoid inverted phase problems, or even to create a pseudostereo signal from a mono input.

SURROUND SOUND

In contrast to *Fx:phase*, the *Fx:surround* plug-in (see **Fig. 3**) is the most complex of the effects and offers the most unusual features. This plug-in lets you position your signal anywhere within the sound field, even behind the listener. It achieves this by creating a

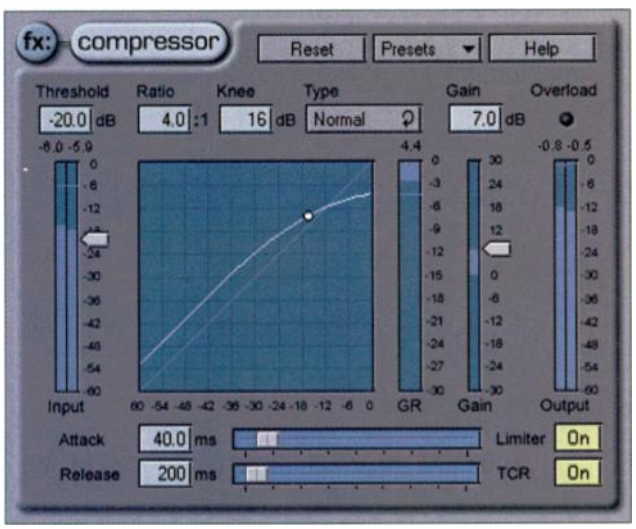

FIG. 2: *Fx:compressor* offers two modes of operation: Normal and Vintage. In Vintage mode, you can emulate compression with more warmth and punch.

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PRODUCT SUMMARY

Ultrafunk
Sonitus:fx 2a (Win)
DirectX/VST effects plug-ins
$200

FEATURES	4.0
EASE OF USE	4.0
DOCUMENTATION	2.5
VALUE	4.0

RATING PRODUCTS FROM 1 TO 5

PROS: Superb sound quality. Excellent user interface. Useful for a wide range of applications. Very low CPU load.

CONS: Documentation could be more detailed and include tutorials.

Manufacturer
Ultrafunk
e-mail info@ultrafunk.com
Web www.ultrafunk.com

Fx:equalizer to the guitar track. A 6 dB boost at 92, 351, and 2,890 Hz filled in the sound. I came across one annoyance while working with the EQ plug-in, though. When I accidentally hit the Flat button, all of my EQ settings disappeared—an undo feature would have been useful at that point.

Next I used *Fx:modulator* to make the horns sound a bit bigger. The Flanger and String Phaser modes didn't get me what I wanted, but the Ensemble mode worked perfectly. It's more effective than a regular chorus effect because it uses three nonsynchronized modulating delays.

Finally, I added some reverb to the entire mix with *Fx:reverb*. I was looking for a tight, warm room, and though none of the presets provided the settings I needed, I had no trouble getting the right sound. The multitude of parameters that this plug-in provides was somewhat overwhelming at first, but after a bit of use, I felt right at home. For versatility, *Fx:reverb* is one of my favorite plug-ins, and it sounds excellent.

To test the plug-ins' parameter automation (available only with a VST host), I gave the effects a try under *Cubase* 5.0. *Fx:surround* benefits in

particular when used under VST: even though the plug-in's Path feature lets you create automated trajectories for sound positioning, you get more precise results using automation. You're also not limited to 25 control points, which is the limit when the plug-in is used in a DirectX host.

Unfortunately, I wasn't able to test the capabilities of the surround plug-in fully because I don't have a surround setup in my studio. But even with only two speakers, I heard changes in the audio. In particular, I noted the signal's phase being changed, which made the sound appear "outside" the speakers. The effect worked best with both high and band-limited sounds. I also tested the plug-in with sound effects. Using a footsteps sample, I created an automated path that gave the impression of someone walking around in the stereo field.

I didn't really find much use for the phase plug-in, though it would come in handy for changing a signal's phase to avoid cancellation.

END PROCESSING

It's hard to find fault with this package. All seven plug-ins have excellent user interfaces and go beyond the call of duty in flexibility and processor efficiency. The overall sound quality is excellent, especially considering that the *Sonitus:fx* 2a plug-ins cost less than half as much as some high-end bundles.

▼

***Fx:surround* lets you position your signal anywhere within the sound field.**

Moreover, the plug-ins have extremely low latency, and when I tested their effect on my computer's CPU load (with *Sound Forge*), they each showed a drain of only about 2 or 3 percent. That's pretty unbelievable for plug-ins of this caliber.

In addition, the Preset Manager used by all of the plug-ins lets you save and load your own effects presets. Unlike most other plug-ins, the presets you create are available within any host application, not just the application in which they were created.

My only criticism is that the documentation could be expanded to include more details about how the settings work. Tutorial examples demonstrating how to use the plug-ins would also help. In general, though, the *Sonitus:fx* 2a suite of plug-ins is a great package and a great value. If you want a collection of flexible and efficient plug-ins with superb sound, get funky with *Sonitus:fx* 2a.

Scott R. Garrigus *is the author of* Cakewalk Power *and* Sound Forge Power. *He is also the publisher of the* DigiFreq *music-technology newsletter. For more information, surf to www.garrigus.com.*

16 voices out of the more complex presets. Although the price is on the high end for standalone software synthesizers, the feature set is unique. With the opportunity to fully test-drive the product before you buy, you can hardly go wrong.

Overall EM Rating (1 through 5): 4.5
Rhizomatic Software; e-mail info@absyn.com; Web www.absyn.com

BIG FISH AUDIO

Helter Skelter

By Dan Phillips

Big Fish Audio markets its *Helter Skelter* audio sampling CD ($99.95) by proclaiming its contents to be "the hardest sounds around" and mentioning Rob Zombie, Marilyn Manson, and Nine Inch Nails. The CD features an interesting amalgam of rock, industrial, goth, hip-hop, techno, and metal.

Helter Skelter gives you 18 construction kits, each stocked with guitars, synths, drums, soloed elements, and individual drum hits. You also get drum, noise, and synth and bass loops; effects; multisampled power chords; and a bunch of scratch effects.

Kink Crimson

The construction kits' quality is uneven. Although a few elements within each kit are good, the mixes are generally marred by a variety of sonic and musical ailments. For instance, many tracks are drenched in reverb, resulting in an overly muddy sound.

Some tracks, such as "Unbeliever," contain slamming drums, but most have drum sounds that aren't hard-hitting enough. The chord patterns are often tired clichés, sometimes resulting in a curious mixture of 1970s rock and 1990s industrial.

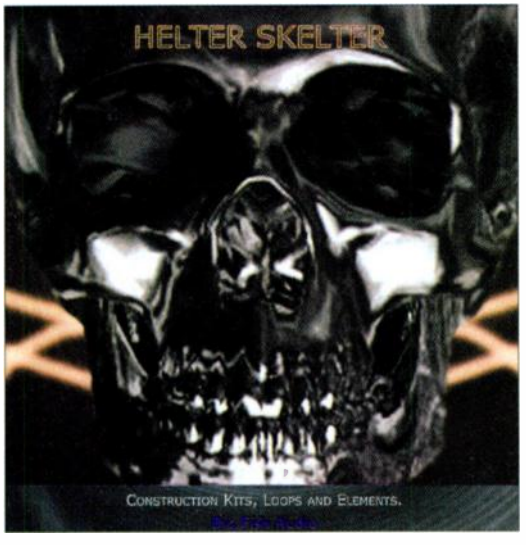
Nevertheless, the collection certainly has some shining moments, especially within its synthesizer elements. For instance, "Take Me" features a ring-modulated synthesizer pattern, and its intensity increases throughout the phrase. Several tracks, including "Unbeliever" and "Sidewinder Phoenix," use effective variations on high-pitched, piercing, horror-movie string stabs. There are also many cool noise and vocal effects, such as the alien speech in "Trench Coat Mafia" that resembles something from *Star Wars*. "Slap Ass 'Til Crimson" had me swaying to its slow, solid, hip-hop groove; slinky synth leads; and a fluttering tremolo delay on the drums.

Sonic Disturbances

The disc's 15 drum loops are more successful than the drums in the construction kits. I enjoyed the solid, 808-style kick drum and hard snare of "Heavy Electro," as well as the strong, driving "Trash Percussion." Similarly, the synth loops offer some good material, such as the crazy arpeggios of "Noise FX Loop 1" and the disturbing, ring-modulated "Stereo SETI Loop."

The Noise Loops section did not fare as well; some of the distorted, pitch-shifted, and otherwise mangled vocal sounds might work well in some settings, but most were too lo-fi and muffled. If I want them that way, I can use my own filters and EQ. Give me bright, full-bandwidth source material, please! Fortunately, the CD ends on a positive note with 70 smooth hip-hop scratch loops.

Helter Skelter is a mixed bag; though it has some gems, finding them means having to sort through a good amount of chaff. I really like the sonic textures of Nine Inch Nails, so I had high hopes for this disc. Too bad it didn't come a little closer to the mark.



Big Fish Audio's *Helter Skelter* sample CD features an interesting amalgam of rock, industrial, goth, hip-hop, techno, and metal.

Overall EM Rating (1 through 5): 2.5
Big Fish Audio; tel. (800) 717-FISH or (818) 768-6115; e-mail info@bigfishaudio.com; Web www.bigfishaudio.com

EAST WEST

Quantum Leap Brass

By David Rubin

East West's *Quantum Leap Brass* fills an important void in most orchestral sample libraries. Desktop musicians who strive for variety and verisimilitude in their jazz and symphonic scores inevitably fall flat when it comes to adding convincing brass parts. In most sample libraries, you can find a modest assortment of solo trumpets, trombones, and saxophones, but few offerings explore the full range of expressive and idiomatic techniques that brass players use. *Quantum Leap Brass,* on the other hand, truly revels in the unique sounds that brass instruments make.

The *Quantum Leap Brass* library ($695) is offered in formats for GigaSampler, Unity, Akai S1000/5000/6000, Kurzweil, Roland, and E-mu. The five-disc collection (six discs for Akai S1000) includes an assortment of solo trumpet, three-part (unison) trumpet section, solo flügelhorn, piccolo trumpet, three-part trombone section, solo tenor and bass trombone, solo tuba, solo saxophone (soprano, alto, tenor, and baritone), and four-part (unison) French horn patches.

Producer Nick Phoenix aptly describes his collection as a "purist, no-compromise library with an emphasis on expression and dynamics." Armed with a variety of high-quality ribbon and condenser mics, Neve and Manley preamps, and Apogee converters, Phoenix recorded all of the samples in large rooms and small to medium halls. Loops were avoided to allow the sounds to evolve naturally over time and to blend realistically in chords. Samples were restricted to the natural instrument ranges (though some of the outer limits do seem a bit strained), and special attention was paid to properly capturing the all-important

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attack portion of each sample. The hard work clearly paid off: this is a stellar collection of samples that mix together beautifully and encourage creative brass writing.

Mod Squad

The GigaSampler version of *Quantum Leap Brass* (used for this review) is especially noteworthy for its many patches that employ the Mod wheel for sample switching, crossfading, and filtering. (For the hardware-sampler formats, many of the Mod-wheel-controlled patches have been split into multiple patches.) The clever use of the Mod wheel offers some exciting real-time manipulation of patches. For example, you can quickly change from doits to shake falls in one solo trumpet patch, from short falls to long falls in another, and from plunger wah to flutter wah in yet another.

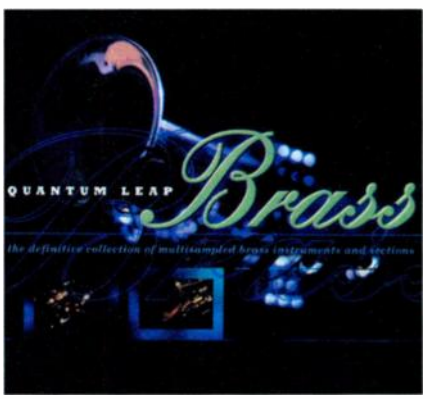
Several patches combine the Mod wheel with Velocity switching to provide an even greater range of expressiveness. In one patch, for example, the Mod wheel in the down position offers three levels of staccato; in the up position it offers three levels of sustain. Of course, these multilevel patches can be quite large, but they offer an exceptional degree of real-time control.

Kickin' Brass

A couple of the trumpet patches do a wonderful job of capturing the cool Miles Davis muted sound, allowing you to switch between long, evolving sustained tones and short staccato notes. Another patch lets you change the filter setting in real time for subtle yet effective expressiveness. If subtlety isn't your thing, though, you'll love "minidamoocha," an over-the-top, funky toilet-plunger patch that is perfect for 1930s band arrangements and cartoon soundtracks. The Mod wheel changes the sound instantly from short wah to sustained wah to a screaming growl that is guaranteed to make you smile.

Flügelhorn is represented by a single mellow-sounding patch with a nice round timbre, and the piccolo trumpet should serve you well the next time you need that *Brandenburg Concerto* sound for a project.

As with the solo trumpet patches, the trumpet section patches include a number of open and muted sounds along with an array of falls, doits, shakes, and sforzandos. The solo and unison trombones offer a comparable set of effects, including flutter crescendos, wahs, falls, slides, and scoops. Best of all, these patches sound like real trombones; you can create a big-band trombone section, and it doesn't come off sounding like a mushy pipe organ. A great-



East West's five-disc *Quantum Leap Brass* collection offers excellent brass samples with a wide array of characteristic performance techniques.

sounding and versatile tuba rounds out the low end.

The sax patches are uniformly high in quality and loaded with playing techniques and effects that add to their expressiveness. As with the trombones, the saxes sound great as lead instruments and when combined into harmonized sections—an essential consideration for jazz charts.

The final half dozen patches offer unison French horn sections with different playing styles. Characteristic effects include sforzandos, rips, falls, and other techniques that are well suited to film scoring. It's too bad there aren't some solo horn patches to complete the set.

Wrap

East West *Quantum Leap Brass* offers an impressive collection of samples. The recording quality is consistently first-rate with clean, well-miked samples and carefully matched natural reverb. The documentation is exemplary with concise yet clear descriptions of the patches and their controller setups. Several patches even include helpful hints for maximizing results. If you're tired of boring, uninspiring brass samples, check out *Quantum Leap Brass*. It's a real blast.

Overall EM Rating (1 through 5): 4.5
East West; tel. (800) 833-8339 or (212) 541-7221; e-mail sales@eastwestsounds.com; Web www.soundsonline.com

BIG BRIAR

Moogerfooger CP-251 Control Processor

By Gino Robair

Thanks to their voltage control-capabilities, the effects of Big Briar's Moogerfooger line blur the distinction between traditional floor-bound stomp boxes and analog synthesizer modules. In an attempt to mold the product line into a full-fledged modular synth system, Big Briar has introduced the Moogerfooger CP-251 Control Processor ($299).

As the name suggests, the CP-251 works with control voltages rather than audio signals, and Big Briar cleverly housed eight independent control processors in one small box. The processors include a mixer, lag processor, voltage-controllable LFO, noise generator, sample and hold, two attenuators, and a four-way multiple. Together, these features provide a useful tool kit for creating complex patches between Moogerfooger products, analog synths, and any other device that uses control voltages.

Big Briar's Moogerfooger CP-251 Control Processor gives you eight different ways to process control voltages. The red-ringed input jacks are active and used to power expression pedals.

Due Process

The CP-251, measuring 5.25 inches tall and 9.5 inches wide, is slightly larger than a VHS video cassette, and you can place it in a rack-mount unit by removing the wooden sides. The CP-251 uses a +9V wall wart and includes two power inputs—one on the side, for use when the CP-251 is on a table, and one on the bottom, for use when the unit is rack-mounted.

The CP-251's ¼-inch jacks accept 2-conductor (TS) or 3-conductor (TRS) plugs. The eight red-ringed input jacks are active and used for powering expression pedals, such as the Moogerfooger EP-1. The four-input mixer, for example, has two active pedal inputs, individual level controls for inputs one and two, and a master level control. In addition, you can add or subtract as much as five volts from the mixed signal by using the Offset control. The mixer has two outputs; the signal of the second output is the inverse of the first. You can create simultaneous melodic patterns in contrary motion by using both outputs with a pair of voltage-controlled oscillators (VCOs).

The Lag Processor offers independent control over a signal's rise and fall time. You can use this to add portamento between notes or to reshape a square-wave input into sawtooth and sine waves.

Two of the CP-251's sections—LFO and Noise—produce audio-rate signals. Noise is available from a single output jack. The LFO features an active input and has dedicated triangle- and square-wave outputs. The LFO frequency range is from 0.2 to 50 Hz, but it can be extended to 0.02 and 100 Hz using a control voltage.

The CP-251's Sample-and-Hold circuit features two active CV inputs. Nevertheless, the Noise module is conveniently normalled to the voltage input, and the LFO square-wave is normalled to the trigger input, which lets you use the Sample and Hold without patch cords. When used in this configuration to control a VCO, Sample and Hold provides a steady stream of randomized pitches, with the tempo controlled by the LFO's frequency.

Like the Mixer, Sample and Hold has two outputs. The first is the direct output of the sampled voltage. Before the signal reaches the second output, however, a lowpass filter smooths the signal's contour. The LFO Rate knob determines the degree of smoothing by controlling the lowpass filter's cutoff frequency.

The remaining processors—two attenuators and a four-way multiple—may seem boring compared with the other processors, but they are just as useful. The attenuators are great for quantifying the level of a control signal, such as an expression pedal's voltage range. Each attenuator has an input, an output, and a control for varying the signal level.

The Four-Way Multiple creates up to three copies of an input signal and works with both voltage and audio signals. The Multiple includes an active input, so you can send the signal from an expression pedal to as many as three destinations.

One with Everything

The CP-251's manual is highly informative. It includes a basic tutorial about the use of voltages for control, an explanation of each of the CP-251's processors, and sample patches for using the CP-251 with other Moogerfooger products.

Although the Big Briar CP-251 is priced in line with most single-process analog-synthesizer modules, it's a true bargain when you consider the amount of processing power it gives you. The CP-251 is an essential purchase for any serious user of voltage-controllable gear.

Overall EM Rating (1 through 5): 5
Big Briar; tel. (800) 948-1990 or (828) 251-0090; e-mail info@bigbriar.com; Web www.bigbriar.com

MIT PRESS

The Csound Book

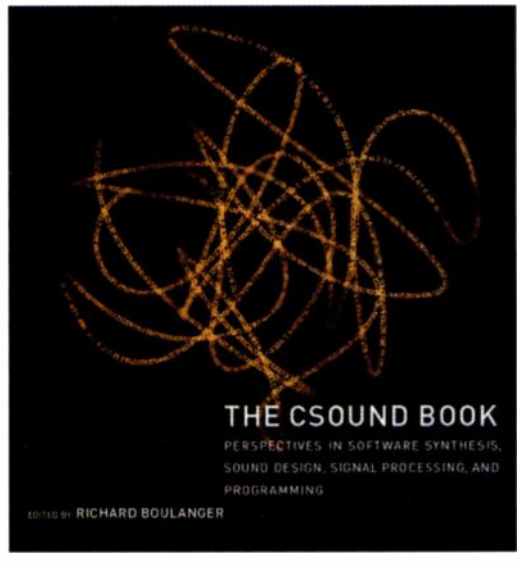
By Lee Ray

Understanding and using a software-synthesis language like Csound is a big undertaking. *The Csound Book*—from MIT Press ($55) and edited by Richard Boulanger—finally offers a comprehensive "cookbook" that can have you up and running pretty quickly.

The Csound Book and its two companion CD-ROMs gather everything necessary to use this powerful sound programming language. Applications for PC, Mac, Linux, BeOS, and other platforms are included along with 77 articles by Csound experts, tutorials from experienced Csound educators, and several hundred examples of Csound instruments. There are scores and orchestras for nearly 100 compositions made entirely with Csound, graphical user interfaces, manuals, and FAQs. If you can't find what you need in the book, surf over to the many Web pages referenced in the text for additional resources.

Sound in C

Csound is a programming language for synthesizing and processing sound. To work with the language, the user creates two text files—an Orchestra and a Score—and runs them through the Csound compiler. It's a powerful approach that requires tremendous attention to detail. *The Csound Book* provides example files in addition to explanatory text, allowing you to try out the theories and techniques covered in the book. The CD-ROMs contain compiled audio files of each article's examples. When you can't stand to read another sentence about HRTF data sets or generating normalized indexes for wavetables, just play the sounds.



***The Csound Book*, edited by Richard Boulanger, is a massive compendium of information about Csound, arguably the world's most powerful sound-synthesis language. The text includes hundreds of examples and is accompanied by two CD-ROMs.**

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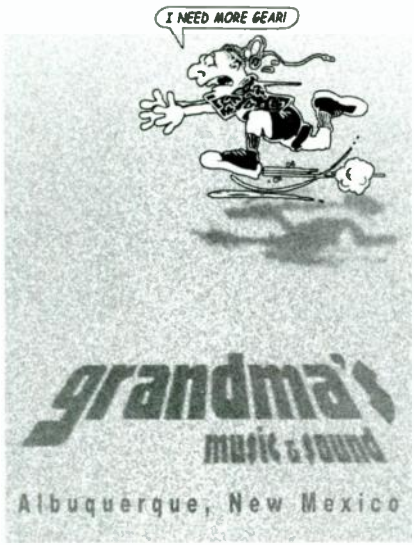

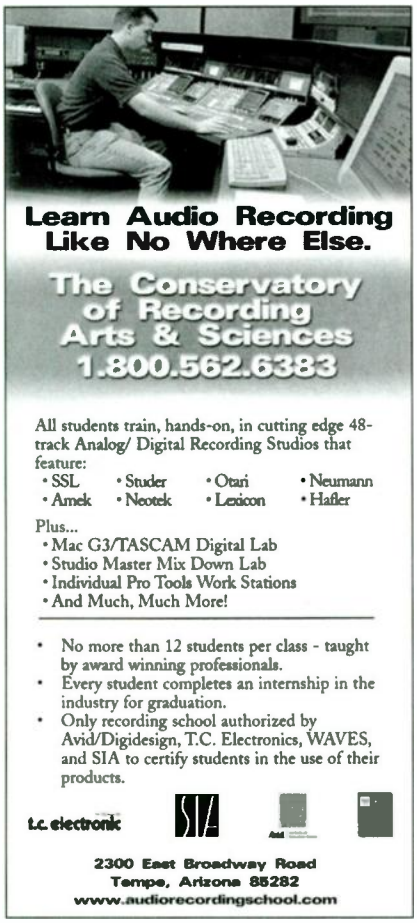

The CD-ROM articles have hyperlinks for all Csound functions. Clicking on one shows the matching entry in the manual, complete with commentary, correct syntax, and examples. That is a real productivity boost that gives users properly functioning code to use in their own creations.

Chapter and Verse

In tone, organization, and language, the texts read like material for college-level courses and are suitable for any motivated individual. There may be room in the world for a *Csound for Dummies* book, but this is not that book. For the most part, the authors took a practical approach and delivered informative, readable text along with actual working instruments. However, chapters vary from the practical to the abstract. Reading and making sense of the book as a whole, the way one would read a novel or even a manual for a single piece of gear, will not work.

For example, the opening article by Boulanger introduces the Csound language in an easy-to-follow progression of theories, études, and exercises. The article seems like a music lesson, perhaps not surprising given the many years Boulanger has taught Csound. Eric Spjut adopts a similar approach. He uses Csound to audibly illustrate fundamental concepts of digital signal processing. A musician may well find such material heavy going, at least until reaching the more useful sound examples such as Spjut's Voice Scrambler-Descrambler.

Articles of specialized interest, such as creating graphical user interfaces, performing sonification or algorithmic composition, and even the inner workings of the Csound language itself, may not be useful to all readers. Similarly, if you like the music of the various composers who have written about their work, you'll no doubt be interested in how they approach the composition process.

Some articles seem too brief for their subjects. For instance, Hans Mikelson's "Modeling 'Classic' Electronic Keyboards in Csound" covers a Hammond organ–Leslie speaker emulation in seven pages, two of which are code listings. Desktop musicians, sound effects creators, producers, and sound designers will all get something out of *The Csound Book*, however.

I would have welcomed a more complete index. The one provided really needs more entries. Worse, the articles on the CD-ROM are not indexed at all. Because *The Csound Book* is encyclopedic in scope, it should offer many different ways to access the information it holds.

The Csound Book is really a kind of hyperbook: you don't so much read it as use it. The book combines executable software, instrument (patch) manuals, white papers, application notes, operation theories, user contributions, and source code to one of the most powerful sound-synthesis programs ever. *The Csound Book* is real value for the money.

Overall EM Rating (1 through 5): 4.5
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DIAMOND CUT PRODUCTIONS

Diamond Cut Live 4.59 (Win)

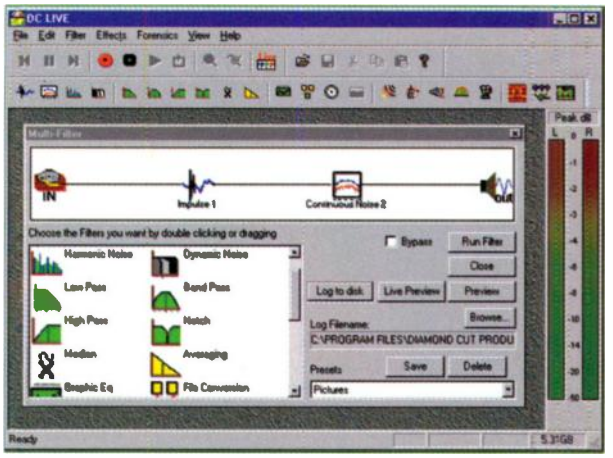
By Brian Smithers

Diamond Cut Live ($995) is the latest incarnation of Diamond Cut Productions' well-respected audio-restoration tool kit. It adds real-time performance to an already formidable bundle of noise-reduction and audio-enhancement tools.

A review of version 3.06 in EM's October 1999 issue praised *Diamond Cut* for the quality and variety of its effects but took issue with its "cumbersome" user interface. For the most part, version 4.59 exhibits the same strengths and weaknesses. The package includes specialized filters for removing everything from clicks and pops on vinyl to electrical hums. It also includes lowpass, highpass, bandpass, and notch filters. After cleaning up your audio, you can sweeten it with basic reverb, dynamics, and spectral-enhancement effects.

Is It Live?

The big news is *Diamond Cut*'s Live mode. Imagine doing a live radio show featuring classic tracks from scratchy old 78s and having the ability to remove every click and pop in real time without ever committing data to your computer's hard drive.



***Diamond Cut Live*'s Multifilter lets you chain a variety of specialized filters and effects for real-time noise reduction and audio enhancement.**

While you're at it, why not warm up the low end with some gentle EQ and add a hint of reverb? All of this and more is possible with *Diamond Cut Live.* Depending on your computer's processing speed, you can string together any number of filters to address whatever ails your audio. Play dirty audio into your computer; after a short processing delay, cleaner audio is issued from your sound card's outputs. Of course, you'll need a full-duplex card to take advantage of Live mode.

I put *Diamond Cut Live* to the test with a 20-year-old cassette of Mussorgsky's *Pictures at an Exhibition* that I copied from vinyl. The original recording has an unusual amount of tape hiss for a classical recording, and my turntable contributed a steady low rumble. Add a constant barrage of clicks, pops, static, and tape hiss from the cassette, and you have a serious audio-restoration challenge.

I opened the software's Multifilter, dragged in the Impulse Filter and the Continuous Noise Filter, dialed up an appropriate preset on each, and heard nothing. The well-written manual explained that I must press the Live Preview button to hear the filters in action. Preview seems like a misnomer for a real-time feature; nonetheless, my concern with taxonomy vanished when I heard the results. *Diamond Cut Live* stripped away the flaws, leaving me with renewed appreciation for a great performance.

Quibbles and Bits

Diamond Cut Live could run several filters at once on my Pentium III/450 MHz, but it choked when I used extreme settings on some filters. I maximized the buffer settings to run the Impulse Filter at the necessary level without dropouts, and that increased processing latency by more than half a second.

Changing settings during Live Preview often caused skips or glitches in the audio output and even produced an occasional crash. Just moving a dialog box caused a dropout of half a second or so. Unfortunately, that effectively reduces *Diamond Cut Live* to a set-it-and-forget-it program, and users might want to tweak settings on the fly.

Whenever you reopen a filter's dialog box, the preset is indicated as Default, even if you change the settings of your selected preset. However, minor annoyances such as these don't outweigh the boatload of high-quality audio-restoration tools that *Diamond Cut Live* offers. At almost $1,000, it's not inexpensive, but the real-time capabilities are certainly an impressive addition to the *Diamond Cut* arsenal.

Overall EM Rating (1 through 5): 3.5
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APOGEE

MasterTools 1.6 (TDM; Mac)

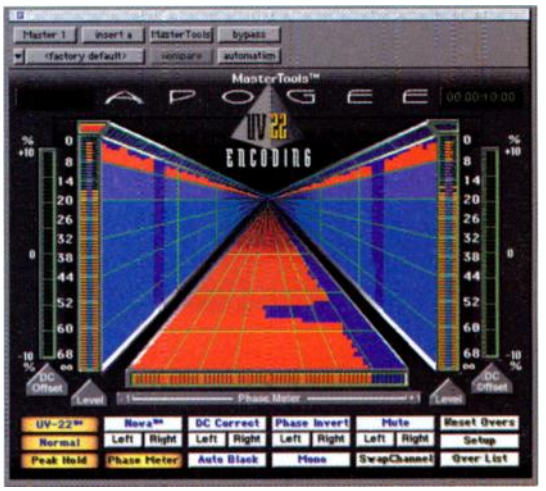
By Jeff Burger

Consumers live in a 16-bit world. However, 24-bit recording in Pro Tools provides both added dynamic range and extra headroom when combining multiple tracks. Other than file size, the only downside is that 24-bit tracks need to be reduced to 16 bits for distribution on CD. Mastering engineers have long favored Apogee's UV22 technology when converting higher bit depths to 16-bit

QUICK PICKS

resolution, and Apogee's *MasterTools* TDM plug-in (Mac; $495) brings UV22 right into the Pro Tools environment.

In lieu of simply lopping off eight bits, most approaches to bit reduction involve dithering and noise shaping, which typically yield some degree of sonic coloration. UV22 adds a clump of high-frequency energy at around 22 kHz—above the theoretic threshold of hearing. That approach smooths out nonlinear elements in the dynamic range in much the same way that bias does with analog tape. The result is that it's hard to hear any difference between the 24-bit and 16-bit versions. Discerning ears will even detect an improvement when running a 16-bit source through *MasterTools*.



Apogee's *MasterTools* TDM plug-in offers the company's proprietary, highly regarded UV22 mastering technology for converting higher bit depths to 16-bit resolution.

Clip Joint

Aside from transparent reduction of bit depth, one of *MasterTools*'s primary functions is to ensure that there is no digital clipping (overs). Users are able to define the number of consecutive full-scale samples that constitute an over. During playback, the software generates a log that displays the time-code locations and duration of any overs that require attention. If users choose not to tweak these points by hand, the software's Nova function will automatically reduce overs to a level that is just below full digital code at 16 bits.

The *MasterTools* interface shows an expansive trench. The front vertical edges of the trench represent the current left and right audio signals with both peak and average levels and holds; the horizontal lip of the trench floor meters phase correlation between the two channels. The trench walls display a five-second history of those three pieces of information. That unique approach conserves screen space relative to traditional horizontal displays.

Pleasure Meter

MasterTools metering is a pleasure. Users can set the headroom scale of the meters as well as the peak hold time. Moreover, the meter's ballistics mirror the strength of the input signal. The program also displays DC offsets between +10 and –10 percent for left and right channels and can automatically compensate for them. Some digital audio converters exhibit a slight DC at their outputs, though Digidesign I/Os are unlikely to have much of a problem in that department.

The remaining controls are straightforward. Auto Black gates any signal that is below a preset threshold. Users can also invert the phase of one or both channels, perform mono checks, and swap channels to ensure proper stereo placement.

Cool Tool

Apogee *MasterTools* is as simple to use because it is sonically transparent. It is hard to refute the credentials and popularity of this mastering technology. The TDM version requires one DSP chip and comes with two authorizations on floppy disk. Alternatively, users can choose challenge-and-response authorization. For anyone serious about mastering directly from Pro Tools, Apogee *MasterTools* does the job at only a fraction of the price of the company's hardware alternatives. ●

Overall EM Rating (1 through 5): 4.5
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digidesign® AVID

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DIGI001 Digital Audio Workstation For Mac And PC

A completely integrated digital recording, mixing and editing environment for the Mac and PC, the DIGI-001 offers a 24-bit multi I/O breakout interface along with Pro Tools LE software— based on Digidesign's world renowned ProTools software. The DIGI-001 interface features 18 simultaneous I/Os made up of 8 analog inputs and outputs— two of the inputs are full featured mic preamps with phantom power, and digital I/O including standard S/PDIF as well as an ADAT optical interface that can also be used as a S/PDIF I/O. ProTools LE supports 24 tracks of 16 or 24-bit audio and 128 MIDI tracks and also features RealTime AudioSuite (RTAS) effects plug-ins. For ease of use, MIDI and audio are editable within the same environment and all mixing parameters including effects processing can be fully automated.

FEATURES–

- 18 simultaneous, 24-bit ins and outs with support for 44.1 and 48 kHz sample rates
- 20Hz - 22kHz freq. response ± 0.5 dB
- 2 channel, XLR mic/1/4" line inputs with -26 dB pad, 48v phantom power, gain knob, and HP Filter at 60Hz
- 6 ch. line inputs (1/4") TRS balanced/ unbalanced w/ software controlled gain
- +4dB balanced 1/4-inch Main outputs
- Balanced 1/4" monitor outs with front panel gain knob
- 1/4-inch unbalanced line outputs channels 3-8
- Headphone output with independent gain control knob
- 2 channel S/PDIF coaxial digital I/O
- 8 channel ADAT optical I/O can also be used as 2 channel optical S/PDIF

Pro Tools LE

- Supports 24 tracks of 16 or 24-bit audio and 128 sequenced MIDI tracks
- Sample-accurate simultaneous editing of audio & MIDI
- Real-time digital mixing capabilities include recall of all mixing parameters, support for edit and mix groups and complete automation of all volume, panning, mutes and plug-ins.
- Route and mix outboard gear in realtime
- MP3 and RealAudio G2 file support (Mac)
- Two plug-in platforms offer multiple options for effects processing— Real-Time AudioSuite (RTAS) is a host-based architecture that allows an effect to change and be dynamically automated in realtime as the audio plays back. —AudioSuite is a file-based format, that renders a new file with the processed sound.
- Bundled RTAS plug-ins include, 1 and 4-band EQ; Dynamics II- compressor, limiter, gate and expander/gate; Mod Delay - short, slap, medium, and long delays with modulation capabilities for chorus or flange effects and dither. AudioSuite plug-ins include Time Compression/Expansion, Pitch Shift, Normalize, Reverse.

MIDI Functions

- MIDI functions include graphic controller editing, piano roll display, up to 128 MIDI tracks and editing options like quantization, transpose, split notes, change velocity and change duration.
- MIDI data can be edited on the fly

TASCAM®

US-428 USB Digital Audio Workstation/ Controller

The US-428 is a 24-bit USB-based audio controller co-designed by TASCAM and Frontier Design Group. The control surface includes plenty of faders, transports and other dedicated controls compatible with the most-used functions in today's DAW applications. The US-428 supports a total of four channels of audio in and two outs simultaneously. The interface plugs right into a USB equipped PC or Mac computer— no opening your computer and no sound card to install. Musicians taking the leap from Portastudios to computer-based DAW programs will feel right at home with the tactile control surface.

FEATURES–

- PC and Mac compatible
- Works with most major DAW programs
- 24 bit D/A and A/D converters
- Bundled with Steinberg's Cubasis VST sequencing software for Windows (MacOS version shipping soon)

I/O–

- Total of four channels of audio in (analog or S/PDIF) and two out simultaneously via USB
- Two XLR mic inputs, two balanced 1/4" TRS inputs, two unbalanced 1/4" inputs (switchable to Hi-Z)
- S/PDIF digital I/O • Two independent MIDI I/O (32 channels)

Controls–

- Unlimited banks of eight faders
- Transport, mute/solo and locate keys
- An EQ module supports control of up to four bands of fully parametric EQ
- Four aux sends and a panpot
- Can be customized to control everything from virtual synths to MIDI lighting panels

MOTU

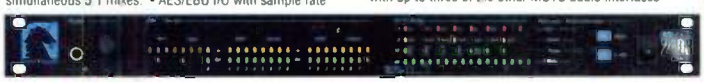
MOTU AUDIO Hard Disk Recording Systems

The MOTU Audio System is a PCI based hard recording solution for the Mac and PC platforms. At the heart of the system is the PCI-324 PCI card that can connect up to three audio interfaces and allows up to 72 channels of simultaneous I/O. Audio interfaces are available with a wide range of I/O configurations including multiple analog I/O with the latest 24-bit A/D/A converters and/or multi channel digital I/O such as ADAT optical and TDIF I/O as well as standard S/PDIF and AES/EBU I/O. Each interface can be purchased separately or with a PCI-324 card allowing you to build a system to suit your needs. Includes drivers for all of today's hottest audio software and AudioDesk, multitrack recording and editing software for the Mac.

THEY ALL FEATURE– • Mac OS and Windows compatible • Includes software drivers for compatibility with all of today's popular audio software plus AudioDesk. MOTU's sample-accurate audio workstation software for Mac OS • Host computer determines the number of tracks that the software can record and play simultaneously, as well as the amount of real-time effects processing it can support • Front panels display metering for all inputs and outputs • **AudioDesk Audio Workstation Software** for Mac OS features 24-bit recording, multi-channel waveform editing, automated virtual mixing, graphic editing of ramp automation, real-time effects plug-ins with 32-bit floating point processing, crossfades, support for third-party audio plug-ins (in the MOTU Audio System and Adobe Premiere formats), background processing of file-based operations, sample-accurate editing and placement of audio, and more

1296 Features–

- 24-bit, enhanced multi-bit 128x oversampling 96kHz converters • A-weighted signal-to-noise ratio of 117 dB
- 12 Balanced XLR inputs and outputs can support two simultaneous 5.1 mixes. • AES/EBU I/O with sample rate conversion both in and out • Compatible with existing PCI-324 cards (requires new PCI-324 driver) • Connect up to 3 1296 interfaces to 1 PCI-324 card for a total of 36 inputs and outputs or mix and match the 1296 interface with up to three of the other MOTU audio interfaces



2408 mkII 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 • • Format conversion between ADAT and DA-88
- 8x 24-bit 1/4" balanced analog I/Os • 24-bit internal data bus for full 24-bit recording via digital inputs • Standard S/PDIF I/O for digital plus an additional S/PDIF I/O for the main mix • Sample-accurate synchronization with ADATs and DA88s via an ADAT SYNC IN and RS422

1224 FEATURES–

- 24-bit analog audio interface • State-of-the-art 24-bit A/D/A • Simultaneously record and play back 8 channels of balanced (TRS), +4 dB audio • 24-bit balanced +4 XLR main outputs • Stereo AES/EBU digital I/O • Word clock in/out • Dynamic range of 116 dB (A-weighted) • Front panel displays six-segment metering for all inputs and outputs • Headphone jack with volume knob

Digital Performer 2.7

Digital Performer is an integrated multitrack digital audio and MIDI sequencing program packed with advanced tools for a wide variety of audio applications. Sample accurate editing, loop based audio capture, realtime DSP effects and the best MIDI timing/resolution available insures unlimited creative potential.

FEATURES-

- 24-bit recording and editing • 32-bit native EQ and effects processing • Includes over 50 real-time MIDI and audio effects plug-ins including 64-bit MasterWorks Limiter and Multiband Compressor • POLAR window provides Interactive audio loop recording • Sample-accurate waveform editing with the tightest sync you can get • 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 video support

NEW FEATURES-

- Full Plug-In FX automation and increased 3rd party support • Drum Editor
- **Adjustable Display Resolution** from 2 to 10,000 PPQ. Tick values up to four decimal places allows 1000 times greater editing resolution.
- **MIDI Time Stamping** (MTS) which exists in MOTU's rack-mountable USB MIDI interfaces, delivers MIDI data from Digital Performer to MIDI devices as accurately as a third of a millisecond for every single MIDI event.

PORTABLE DIGITAL MULTITRACK RECORDING

YAMAHA® AW4416 16-Track 24-Bit Hard Disk Recorder

The AW2416 is an easy to use, high quality 16 track hard disk recorder/digital mixing system designed and built without compromise. Choose between 16- or 24-bit resolution at either 44.1 or 48 kHz recording on a song by song basis. The digital mixing capabilities, based on the acclaimed Yamaha O2R digital mixing console, is expandable to a total of 44 input channels and features 32-bit processing, 4-band parametric EQ, dynamics control, two dedicated effects processors, motorized faders and full automation and library storage capabilities. Eight assignable sample pads allow up to 90 seconds of samples to be placed over two banks for a total of 16 samples. A front panel drive bay accommodates an optional CD-RW drive that allows back-up and storage of songs as well as audio CD burning.

RECORDING/EDITING FEATURES-
- 16 track playback with 8 virtual takes per track
- 16- or 24-bit recording at 44.1 or 48 kHz with 32-bit processing offers as much as 105dB dynamic range
- Manual or Auto Punch In/Out
- Song, Track/Part and Region editing functions
- Jog wheel with Shuttle control
- Up to 99 locate points per song

SAMPLE PADS-
- 8 sampling (trigger) pads, assignable to any channel input of the mixer, x two banks for a total of 16 sounds (90-sec. of sampling time at 16-bit/44.1kHz)

MIXING-
- 4-band parametric EQ and Dynamics per channel
- 2 powerful digital effects processors
- Channel pairing plus 4 fader groups and 4 mute groups
- Recallable scene memories of all settings as well as Channel, EQ, Dynamics and Effects libraries
- Fully automated mixing with 60mm motorized faders

I/O
- 8 analog Mic/Line inputs w/ inserts on channels 1 and 2
- Stereo digital I/O
- Analog outputs include Stereo and Monitor, plus 4 Omni outs to which you can assign up to 4 Group or Aux buss outs, or additional Stereo or Monitor outs.
- World Clock I/O, MIDI In, Out/Thru, MTC Out and a To Host connector for direct connection to a computer
- SCSI connector for external hard drives and other peripheral devices.
- Mouse connector offers point-and-click navigation
- Foot Switch jack for triguering Start/ Stop Play, Recording or Punch I/O by foot.

EXPANDABILITY
- 2 Optional Mini-YGDAI cards provide 8 additional channels of I/O each for a total of 16 inputs and 16 outputs. Available in Alesis ADAT, TASCAM, AES/EBU or additional analog I/O configurations
- Optional internal CD-RW drive

Fostex VF-16 16-Track Digital Multitracker

The Fostex VF-16 is an affordable fully integrated 16 track digital multi-track recorder and digital mixer designed for project and home recording studios. This all-in-one rugged package offers 8 tracks of simultaneous recording, a 16 channel automatable mixer with pan, mute, EQ, compression, effects and 2 aux sends on each channel as well as 8 mic/line inputs— 2 of which are XLR balanced with switchable phantom power and inserts. Also included with the VF-16 is a dual 24-bit effects processor, a SCSI port for data back-up and a full range of non-destructive editing capabilities.

FEATURES-
- 16 tracks plus 8 ghost (virtual) tracks of non-compressed 16-bit, 44.1kHz CD quality recording
- 8 track simultaneous record (16 when using ADAT lightpipe interface)
- 64 x 128 dot-matrix LCD display
- E-IDE 3.5-inch hard drive is used for storage— Up to three hours of recording time per Gigabyte.
- Built-in dual 24-bit stereo A.S.P. effects
- 32-bit processing and mixing

MIXING-
- 16-input channel faders with mute, pan, three-band EQ with parametric mid-range, Compressor, Effects Send and two Aux Sends (selectable pre/post).
- Auto indication and self-illuminating switches allow you to see the status of the entire mixer at a glance.
- 99-mix scene memory
- Totally controllable via MIDI

EDITING-
- Editing functions include COPY/PASTE, MOVE/PASTE, ERASE and non-destructive editing with UNDO/REDO

INPUTS & OUTPUTS-
- 8 x 1/4" Mic/Line inputs
- Channels 7 & 8 feature XLR-balanced mic inputs with phantom power and inserts.
- SPDIF and ADAT I/Os are included as well as external SCSI for backup

ADDITIONAL FEATURES-
- Imports and exports WAV file format.
- ±6% pitch control
- 90 point locate memory

SOUND MODULES

KORG Triton Rack Sound Module

The TRITON-Rack is the long-awaited rack-mount version of the TRITON keyboards. It provides all of the sound and sampling functionality of Version 2.0 of the highly respected synth/workstation/sampler and in addition dramatically expands the possible number of sounds, includes a digital output, allows up to eight EXB-PCM boards to be installed as well as other functions that make it a no-compromise sound source.

FEATURES–
- 60-voice polyphony, 16-part multi-timbral with up to 5 insert and 2 master effects per multi
- 200 user "multi" memory locations.
- Up to 2,057 programs (fully expanded) including 1,664 combinations consisting of up to eight timbres each
- 32MB of ROM with room for up to 8 EXB-PCM 16MB expansion boards
- 102 master and insert effects plus 3-band master EQ
- Dual polyphonic arpeggiator with 328 patterns

OPTIONS–
- ***EXB-DI-*** 6 channel ADAT out with a Word Clock IN
- ***EXB-mLAN-*** supports the upcoming mLAN digital network allowing a single Firewire cable to handle all audio and MIDI I/O signals of the TRITON-Rack
- ***EXB-SCSI-*** interface board for adding an external SCSI device and load AKAI sample data from CD-ROM.
- 6 analog and a 24-bit digital S/PDIF out, 2 mic/line ins
- Built-in 16-bit/48kHz linear sampler with 16MB of RAM expandable to 96MB using (3) 72-pin SIMMS
- Export samples as .WAV and AIFF files
- 240 x 64 dot display- Visual waveform editing
- Time-slicing easily matches rhythmic (looped) samples to a master tempo without changing pitch and creates patterns from the sliced samples
- Song data created on the TRITON can be played back via floppy disk or SCSI device by the TRITON-Rack.
- ***EXB-PCM01-*** Pianos/Classic Keyboards
- ***EXB-PCM02-*** Studio Essentials
- ***EXB-PCM03-*** Future Loop Construction
- ***EXB-PCM04-*** Dance Extreme
- ***EXB-PCM05-*** Vintage Archives
- ***EXB-MOSS-*** DSP 6-voice synthesizer board with 13 synthesis types including analog and physical modeling

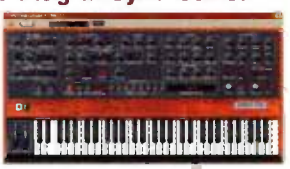
E-mu Systems, Inc. XL-1 Extreme Lead

Xtreme Lead-1 is E-MU's new single rackspace techno/electronica BPM synthesizer. It is an all-in-one dance module with powerful filters and unparalleled rhythmic capabilities including 16 simultaneously synced arpeggiators. The new SuperBEATS Mode allows you to effortlessly trigger, latch and unlatch synced loops and grooves from separate keys on your keyboard. Additional internal ROM expansion capabilities allow you to expand your sound with the many Proteus expansion ROMs available. You can even create your own custom ROMs using E-MU's E4 Ultra samplers.

FEATURES–
- 32 MB Sound ROM w/512 ROM & 512 User Presets
- 64 voice polyphony (128 w/Turbo upgrade)
- SuperBEATS Mode
- 12 assignable real-time front panel controls
- Rhythmic Pattern Generator/Arpeggiator
- 12th order filters
- Advanced synthesis architecture
- 2 outputs (expandable to 6 + S/PDIF w/Turbo upgrade)
- 24-bit dual stereo effects processor
- 1 additional internal ROM expansion slot (expandable to 3 w/Turbo upgrade)
- 16 MIDI channels (expandable to 32 w/Turbo upgrade)
- Plays back Flash ROMs authored on E4 Ultra Samplers

SOFTWARE SYNTHS

NATIVE INSTRUMENTS PRO-52 VST Plug-In Synthesizer



A VST plug-in synthesizer fashioned after the legendary Sequential Circuits Prophet-5. It combines the sonic brilliance, power, warmth and beauty of the original with the practical requirements and advantages available today. Additions to the classic include: no fixed limit to the number of voices, more preset memories, velocity sensitivity, MIDI automation of all available parameters plus the option to run several Pro-Fives in parallel.

FEATURES–
- Operates as VST 2.0 Plug-In
- 2 oscillators per voice
- Choice of pulse, triangle and saw-tooth waveforms
- Detune and synchronization
- 24 dB low-pass filter with resonance and self oscillation
- ADSR envelopes for amplitude and filter
- LFO with numerous modulation options
- Unlimited number of voices (CPU dependent)
- 512 user memories and 512 presets including 50 new sounds by vintage-synth legend John Bowen)
- Velocity sensitivity (optional)
- Real-time MIDI control and automation of all parameters
- Two operating modes: analog warmth or digital accuracy

emagic ES-1 Software Synth

FEATURES–
- Virtual instruments designed for the Logic Series
- Ergonomic user-interface
- Up to 8 simultaneous ES1 instruments, each with up to 16 voices • Sample-accurate timing
- Every parameter can be completely automated
- Instruments can be routed to the effect plug-ins of Logic's internal digital mixer
- With Logic Audio Gold and Platinum, audio recordings can be routed into the ES1 and even used as modulation sources

EXS24 24-Bit Software Sampler

FEATURES–
- Up to 16 instruments can be used simultaneously in Logic Audio with up to 32 voices per unit
- Instruments can be AIFF, WAV, SDII and AKAI S1000/3000 files from 8 to 24 Bit
- Clearly laid out Instrument Editor
- Multi-mode filter • Two full-range envelope generators
- Sample accurate timing
- Full integration with Logic Audio's internal mixer
- Same storage medium for songs and sounds allows total recall of sampler parameters within Logic Audio

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CD RECORDERS & DUPLICATORS

ALESIS Masterlink ML-9600
High-Resolution Master Disk Recorder

The MasterLink ML-9600 combines stereo hard disk recording, CD burning, DSP, and mastering functions to deliver compact discs in the standard "Red Book" 16-bit/44.1kHz format, or high resolution 24-bit/96kHz CDs that utilize Alesis' revolutionary CD24 technology. The ML-9600's amazing sonic quality and powerful built-in tools offers a uniquely versatile and affordable solution for everyone from large commercial facilities to project studios and recording musicians.

FEATURES–
- Reads/writes 16-bit 44.1kHz Red Book Audio CDs as well as files in Alesis' CD24 24-bit/96kHz high-resolution mastering— an AIFF compatible file format that can be read by MacOS, Windows and Unix computer platforms
- 24-bit 128x oversampling A/D/A converters
- Built-in 3.2GB IDE hard drive
- 4x CD burning using standard CD-Rs
- Up to 20-40k Hz frequency response
- 113dB S/N ratio (A-weighted)
- Supports 16- 20- and 24-bit wordlengths and 44.1, 48, 88.2, 96 kHz sample rates
- Built-in sample rate conversion & noise shaping
- Create and store up to 16 playlists containing as many as 99 tracks

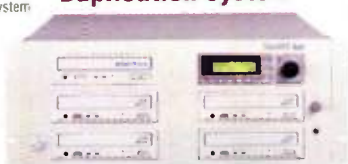
Inputs and Outputs
- **Analog-** XLR-balanced and unbalanced RCA connectors
- **Digital-** AES/EBU (XLR) and coaxial S/PDIF (RCA) I/O
- 1/4" headphone out w/ level control

Editing
- Gain control
- Cropping allows adjusting start and end points
- Join and Split for combining and separating song sections

DSP Finishing Tools
- Equalization, Compression, Normalizing and Peak Limiting

Includes
- Infra red remote control and rackmount brackets

MICROBOARDS Technology, Inc. StartREC
Digital Audio Editing/ CD Duplication System



The Microboards StartREC is the first digital audio editing system combined with a multidrive CD recordable duplication system for professionals. Audio is recorded to the internal 6.2 GB IDE hard drive using analog or digital inputs. Sample rate conversion is automatic. Tracks can be edited and sequenced using the StartRECs user friendly interface and up to 4 CDs can be recorded simultaneously. StartREC is the ideal solution for studio recording, mastering, post production or any pro audio environment requiring digital audio editing and short run CD-R duplication.

FEATURES–
- 2X, 4X, or 8X recording speeds
- 6.2GB IDE hard drive
- Editing functions include move, divide, combine or delete audio tracks, add or drop any index or sub index, and create track fade in or fade out
- Coaxial S/PDIF and AES/EBU digital input plus optical S/PDIF I/O
- XLR balanced and RCA unbalanced inputs and outputs
- Automatic sample rate conversion from 32 and 48kHz
- Automatic CD format detection feature and user friendly interface provide one touch button operation
- Front panel trim pot and LCD display provide accurate input signal and time lapse metering
- SCMS (Serial Copy Management System) is supported, regardless of the source disc copy protection status

StartREC Models Include: ST3000- w/ (3) 8x writers
ST2000- w/ (2) 8x writers ST4000- w/ (4) 8x writers

PROCESSING

JOE MEEK VC1Q Studio Channel
got meek?

The Joe Meek Studio Channel offers four pieces of studio gear in one. It features an excellent transformer coupled mic preamp, an EQ, compressor and enhancer unit all in a 2U rackmount design. Find out why more and more studio owners can't live without one.

FEATURES-
- Fully balanced operation Mic/Line inputs with 48V phantom power and High pass filter switch
- Mono photo-optical compressor
- Front and rear XLR inputs on for easy patching
- Compression In/Out and VU/compression meter switches
- Twin balanced XLR outputs with one DI XLR output for stage use
- EQ section with fixed high and low shelving EQ with approx 18dB lift and cut at 8kHz and 100Hz respectively, sweepable (600Hz to 3.5KHz) mid-band EQ with approx 16dB lift and cut. The 'Q' (bandwidth) value of the mid frequencies increase with the frequency
- Enhancer with Drive, Q, enhance/De-ess control, in/Out switch and enhance indicator
- Internal power supply 115/230V AC

t.c. electronic M3000 Studio Reverb Processor
ULTIMATE SOUND MACHINE

Setting the new industry standard with the VSS-3 technology, the M3000 is the best sounding, most versatile and easiest to use professional reverb today and well into the future. Combining ultimate control of early reflections with a transparent and harmonically magnificent reverb tail, the art of reverberation is brought to a higher level.

FEATURES–
- 250 high-grade factory preset-algorithms: Halls, rooms, plates and springs, Ambience, concert halls, Post small, post large, post outdoor, post FX
- 24-bit A/D conversion, AES/EBU and S/PDIF digital I/O
- Advanced expander and dynamic EQ
- Intuitive user interface with instant preset recall
- 5 band parametric EQ

MICROPHONES

AKG C2000B Condenser Mic

Includes Free H-100 Shockmount

The C 2000 B is an all-purpose cardioid condenser microphone perfectly suited for both recording and live sound situations. The newly developed small-diaphragm transducer capsule is made using a unique manufacturing process that ensures high sensitivity, lowself noise, and excellent bass response.

FEATURES-
- Cardioid polar pattern
- Switchable bass rolloff filter (6 dB/octave @ 500 Hz) and -10dB pad
- Built-in pop screen reduces unwanted noise
- Rugged construction, elegantly styled die-cast metal housing, and silver-gray finish
- **30 Hz to 20 kHz frequency response**

SHURE KSM44/SL
Multipattern Condenser Mic

The KSM44/SL is a multiple pattern dual large diaphragm condenser microphone built without compromise using premium electronic components and gold-plated internal and external connectors. The KSM44/SL is a premium vocal mic and is equally adept for close miking a wide range of acoustic instruments, amplifiers and for ambient room miking.

FEATURES-
- Dual 1-inch gold-layered, Mylar diaphragms
- Class A, discrete, transformerless preamplifier
- Cardioid, omni and bidirectional polar patterns
- Subsonic filter eliminates rumble from mechanical vibration below 17 Hz
- Integrated 3-stage pop grille and shock mount
- 15 dB pad and 3-position switchable low-frequency filter virtually eliminates unwanted background noise and controls proximity effect
- Includes ShureLock elastic-suspension shock mount and swivel mount, protective pouch and locking aluminum carrying case
- **20 Hz - 20 kHz frequency response**

STUDIO MONITORS

EVENT ELECTRONICS PS-5 Bi-Amplified Project Studio Monitors

The PS-5s are small format, full-range, non-fatiguing project studio monitors that give you the same precise, accurate sound as the highly acclaimed 20/20 series studio monitors. The use of custom driver components, complimentary crossover and bi-amplified power design provides a wide dynamic range with excellent transient response and low intermodulation distortion.

FEATURES-
- 5-1/4-inch magnetically shielded mineral-filled polypropylene cone with 1-inch diameter high-temperature voice coil and damped rubber surround LF Driver
- Magnetically shielded 25mm diameter ferrofluid-cooled natural silk dome neodymium HF Driver
- 70 watt continuous LF and 30 watt continuous HF amplification per side
- XLR-balanced and 1/4-inch (balanced or unbalanced) inputs
- 52Hz-19kHz frequency response ±3dB
- 2.6kHz, active second order crossover
- Built-in RF interference, output current limiting, over temperature, turn-on transient, subsonic filter, internal fuse protection
- Combination Power On/Clip LED indicator
- 5/8" vinyl-laminated MDF cabinet

Hafler M-5 Passive 2-Way Studio Monitors

The Hafler M5s are lightweight, portable studio monitors with all the qualities of the TRM6 in a more compact, non-amplified package. They are an ideal monitoring solution for broadcast and project studio environments.

FEATURES-
- 70 – 21k Hz frequency response ±3dB
- 20 - 200 watts power handling
- 5.25" polypropylene/rolled nitrile rubber surround 1" silk dome/waveguide tweeter
- 5-way gold plated binding post inputs
- Shielded woofer magnet
- User selectable front panel 3dB tweeter level control
- 4th order Linkwitz-Riley crossover at 3.2kHz, Zobels, tweeter overload protection
- Dimensions 12.25"H x 6.75"W x 7"D
- Weight 12 lbs. net

VERGENCE M-00 Powered Mini Monitor System

The M-00s are an integrated, self-powered, 2-way acoustic suspension mini monitoring system designed for near/mid-field monitoring. They're portable enough to take anywhere, have balanced and unbalanced inputs with lots of output power (75 watts/ch) and a tough cast metal enclosure.

FEATURES-
- 4.5" treated paper woofer, 1" soft fabric dome tweeter with full magnetic shielding)
- Built-in 75 Watt per channel (continuous) amplification
- 98 - 20k Hz frequency response ±2dB @ 1M
- XLR, TRS & RCA input connectors
- Cast aluminum/zinc alloy body & Glass-filled ABS baffle.
- -10, +4dB input sensitivity & near/mid-field proximity switching
- Power On, Auto-On, Off
- Sensitivity, Power & Standby Display
- Anti-clip circuitry
- 9"h x 5.7"w x 7.3"d / 14 lbs.

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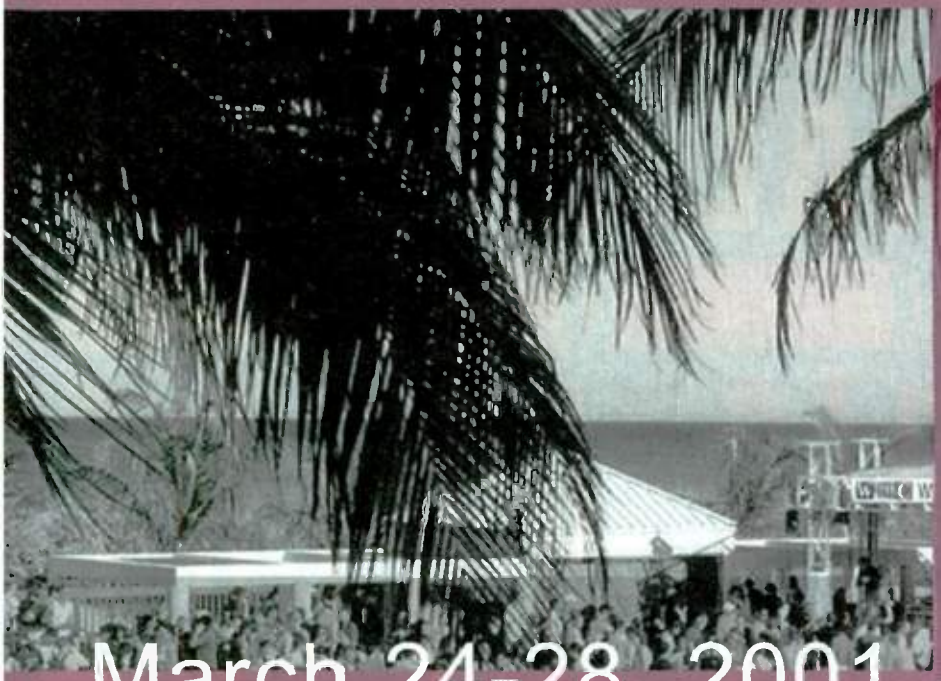

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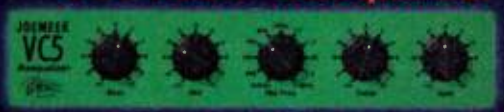

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By Larry the O

When a Divider Is the Uniter

I am often of two minds about things, and sometimes that's on purpose. For example, I lost my ability to enjoy a good movie for a couple of years after I started doing audio post-production for film. I would get so focused on the sound that I'd miss significant chunks of the movie. Determined to reclaim my wayward enjoyment, I cultivated a meditational method of movie watching in which I'd acknowledge observations about the soundtrack and then let them out of my mind.

To provide an outlet for my urge to be the Central Scrutinizer, I would sometimes make an explicit decision before viewing a movie to study its sound and not worry about watching it. (This was generally for action blockbusters that offered little beyond spectacular effects.) Over time, it became natural and unconscious to watch the movie and occasionally make a mental note of some sonic occurrence.

This ability to divide your consciousness is critical in performing situations such as soloing over a sophisticated musical structure—part of your mind has to track the form while another part generates the creative statement to fit with it.

However, some conditions demand a physical separation in time or space, as opposed to a purely mental division. Consider the most double-cursed of performers: the drummer who knows how to set up the P.A. system. Drummer and actor Joe Paulino recently regaled me with war stories of frantically alternating setting up his drums with troubleshooting hums in the sound system right up until the downbeat, and then having to jump up from the drums to stop feedback. I endured similar circumstances many times—until the gig at which the mixer was 40 feet from the stage, and I was totally exhausted from running back and forth by the time downbeat came.

After that experience, I decreed to the band that we would have to shoulder the expense of a sound engineer because I couldn't continue to divide my energy. Even with a sound engineer, though, I still have to really focus sometimes to ignore sound problems that happen while I'm playing.

In fact, some situations can thoroughly challenge your ability to think on more than one plane. I recall the night that same band was onstage with the unflappable Mike "Scoop" Haeffelin mixing our sound, and one of the front-of-house P.A. amplifiers exploded into flame. Did I mention it was a tiny stage?

Now *that* was a circumstance that narrowed my consciousness to one choice centered on a more physical separation in space: whether to keep playing or jump the hell off the drums and either deal with it or just run away. (It also proved to be the only time I ever saw Scoop break into a full run on a gig, so I just kept playing and let him deal.)

Beyond carrying fire extinguishers to gigs, the most successful strategy I've found for managing split-role situations is to draw clear lines that separate and confine the roles. Often this involves setting a threshold: below it, you don't cross the lines between roles; above it, you do. (Let's face it, you can't always have separation.) For example, if a microphone is ringing, I keep playing. If the entire sound system goes into screeching feedback, I dive for the master fader (or, in the worst case, the AC power). It's not easy to decline to do something of which you're capable, but in the long run, your sanity may depend on it.

Unfortunately, not all circumstances yield to this approach. For example, you can't apply this method if you're engineering and producing simultaneously. A powerful unity can come out of doing both on a project, but engineering and producing can be heavy-duty jobs that are inextricably linked yet not the same.

At best, being able to function on more than one level gives me a richer experience and a broader perspective. At worst, it can be downright schizophrenic. Either way, recognizing the beast of duality is the first step toward taming it. ❦

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