JAM
the EM Guide to Gigging

Inside! Free Bonus 'Zine for Gigging Musicians

Electronic Musician

February 1997

Home Improvement

Pro designers help you build the ultimate personal studio

Become a digital audio file expert

Danny Elfman on film composing

Are you violating electronic copyrights?

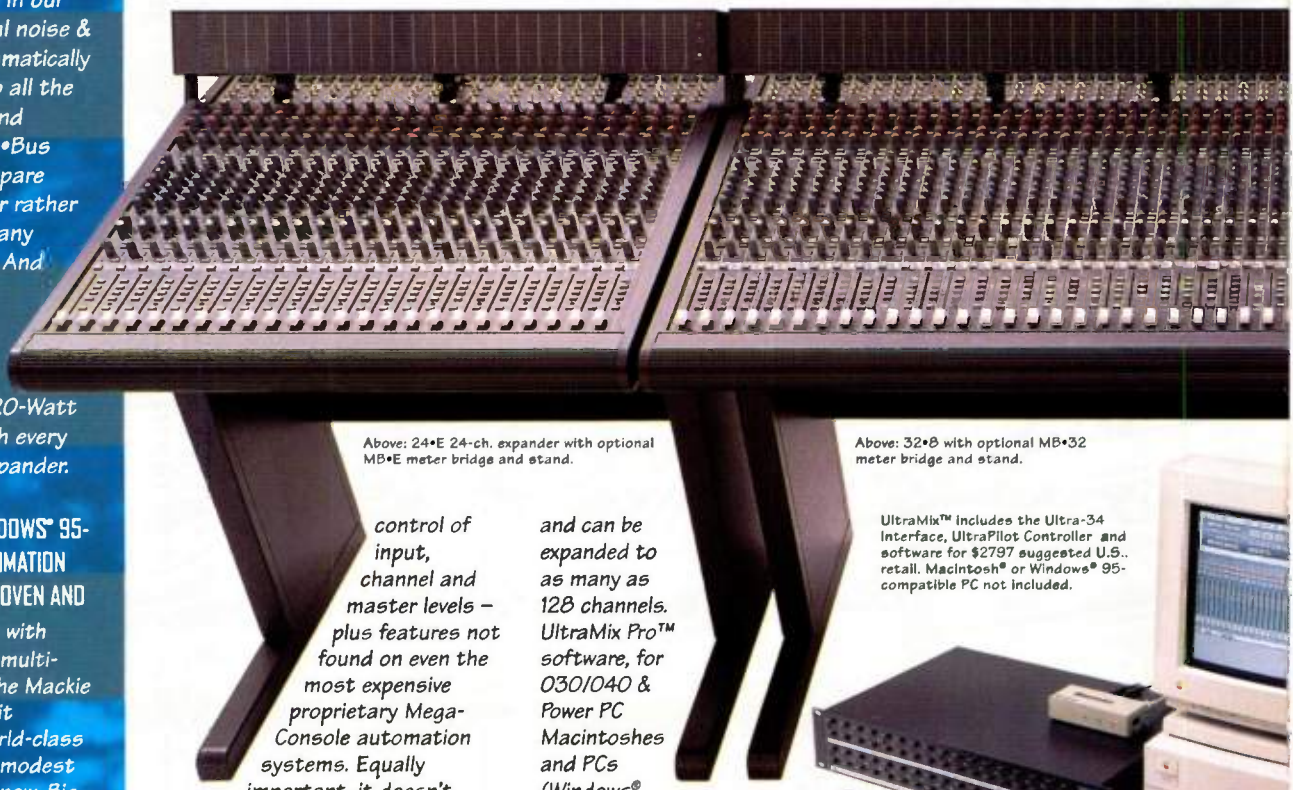
7 REASONS TO BUY OUR MIXERS TO MIX MORE CREATIVELY,

1 VLZ CIRCUITRY FOR ULTRA-LOW NOISE AND CROSSTALK. *A fancy new name for the same old circuitry? Nope. VLZ (Very Low Impedance) is a Mackie innovation based on solid scientific principles. Through the careful deployment of high operating current and low resistor values at critical points in our consoles, thermal noise & crosstalk are dramatically reduced. Open up all the channels, subs and masters on an 8•Bus console and compare what you hear (or rather don't hear) with any Brand X console. And because VLZ circuitry needs loads of high current, we ship a humongous, 220-Watt power supply with every 8•Bus & 24•E expander.*

2 IT EXPANDS ALONG WITH YOUR NEEDS AND BUDGET. *You'd be surprised just how many 8•Bus console setups like the one below are currently in use. But you don't have to start out this way. Start out with a 24•8 or 32•8 and then grow your 8•Bus console 24 channels at a time with our 24•E add-on modules. 1, 2 or even 3 of 'em connect in minutes. They come with their own 220-watt power supply; optional meter bridges are available.*

3 IMPECCABLE MIC PREAMPS. *A console can have motorized dooflammers and an optional MIDI espresso attachment, but if the mic preamps aren't good, you don't have a fully-useful production board. Our discrete preamps with large-emitter-geometry transistors have won a critical acclaim for their exceptional headroom, low noise (–129.5dBm E.I.N.) & freedom from coloration. VLZ circuitry in the preamp section also reduces crosstalk.*

4 THIS CONSOLE JUST PLAIN SOUNDS GOOD. *Sure, you may be able to buy a Brand X console for less. But you end up with a console that sounds like...well...a Brand X console. Granted, we're getting into a pretty subjective area here...but we have tall mounds of 8•Bus warranty cards that rave about our consoles' "clarity," "sonic purity," "sweet sound," "transparency," "lack of coloration" and a lot of other superlatives we wish we'd thought of first.*



Above: 24•E 24-ch. expander with optional MB•E meter bridge and stand.

Above: 32•8 with optional MB•32 meter bridge and stand.

7 MAC® & WINDOWS® 95-BASED AUTOMATION THAT'S RELIABLE, PROVEN AND AFFORDABLE. *Along with affordable digital multi-track recorders, the Mackie 8•Bus has made it possible to do world-class productions on a modest budget. But until now, Big Studios have still had one remaining and unattainable creative "secret weapon"... computerized level automation. That's why we developed the UltraMix™ Universal Automation System. It gives you fully editable and recallable control of input, channel and master levels – plus features not found on even the most expensive proprietary Mega-Console automation systems. Equally important, it doesn't degrade sound quality, introduce zipper noise or cause audible "stepping." UltraMix is currently being used to mix network television music themes and on several major album projects – by seasoned engineers who grew up on Big Automation Systems. Their verdict is that UltraMix is a serious automation solution – stable, reliable and frankly easier to use than more expensive systems. The basic system controls 34 channels and can be expanded to as many as 128 channels. UltraMix Pro™ software, for 030/040 & Power PC Macintoshes and PCs (Windows® 95 required), includes a wealth of features like editable fader curves, built-in level display, up to eight subgroups, SMPTE time code display, event editor with pop-up faders, optional control of outboard effects devices, and the ability to play Standard MIDI files from within the program.*

UltraMix™ includes the Ultra-34 interface, UltraPilot Controller and software for $2797 suggested U.S.. retail. Macintosh® or Windows® 95-compatible PC not included.

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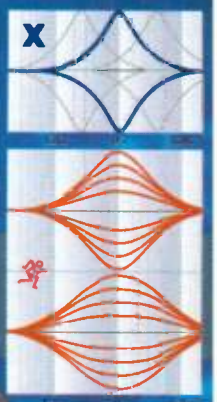
8-BUS CONSOLE... AND 2 TIPS ON HOW EFFICIENTLY AND, WELL, MORE FUNLY.*

5 PROFESSIONALS REALLY USE THEM. *The members of Boyz II Men could have afforded any console they wanted for their studio's second room. They chose an 80-input 8•Bus setup with 102 channels of UltraMix™ automation. In the studios of artists as diverse as k.d. lang[1], Yes, Queensryche, Aerosmith, Lee Roy Parnell, Bryan Adams, Carlos Santana, Whitney Houston, Eric Clapton & U2, our consoles really are used to make great music.*

6 WIDE MID RANGE EQ. *Whether you're tracking or mixing, equalization is one of your most important creative tools. Mackie's 8•Bus consoles feature extremely-wide-bandwidth peaking EQ that can be used to achieve effects that simply aren't possible with narrower EQ. Most Brand X midrange EQs have a fixed bandwidth of about 2 octaves (blue graph at right). You can sweep it up & down the frequency spectrum, but the "sharpness" of the EQ curve is always the same. This kind of EQ is good for some purposes... but if you've worked with it before, you know it's too drastic and localized for gentle changes in overall tonal coloration.*



The 8•Bus' true parametric Hi Mid lets you spread the band-width out to as much as 3 octaves (red curves above). That extra octave of "width" gives you a whole new creative palette.

Above: 24•E 24-ch. expander with optional MB•E meter bridge and stand.

Above: The SideCar, matching 8•Bus equipment rack.

8 WHAT ULTRAMIX AUTOMATION CAN DO FOR YOU:

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- *Clone your best fader moves and use them in other places in the mix*
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- *Make six voice-over versions of a jingle mix – and then easily make the inevitable nitpicky client changes three days later*
- *Step up to big-league automation without breaking the bank!*

9 LEGENDARY RELIABILITY. *One of those factors you probably don't think much about – until your console goes down in the middle of a critical late-night session. Built with pride in Woodinville, WA USA, Mackie 8•Bus consoles have an enviable three-year track record for enduring continuous, round-the-clock use and abuse.*

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1 Mention in this ad denotes usage only, as reported to Mackie Designs and in no way intended to constitute official endorsement by the artists or groups listed.

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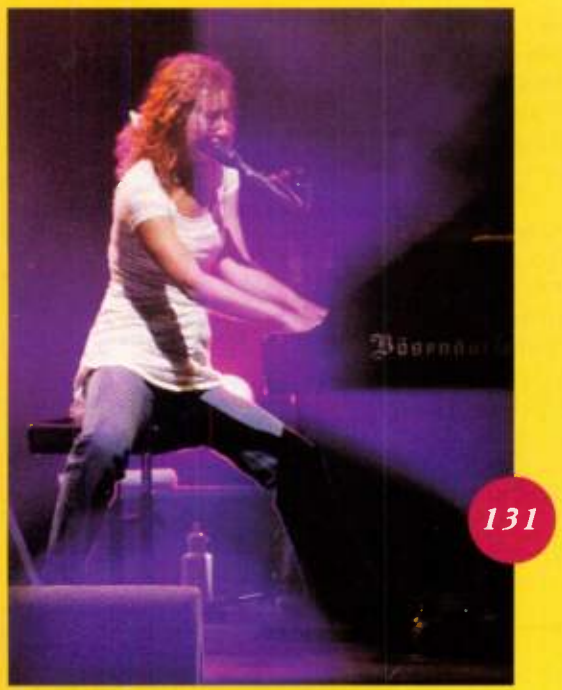
FEBRUARY 1997 VOL. 13, NO. 2

COLUMNS

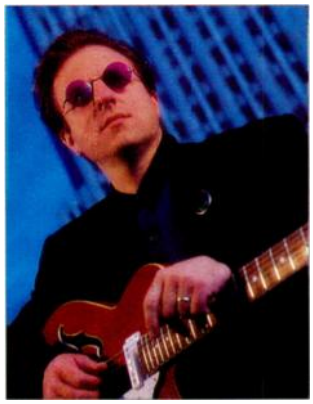


REVIEWS

Cover: Illustration by David Bishop.

No Rest for the Wacky

Why are most artists committed to excellence?



ROBERT PERRY

Way back in the early 1960s, after I had surrendered to the temptations of rock 'n' roll, my parents must have suffered more than a few nightmares about their oldest child becoming insubordinate, drug addled, and hopelessly mad. Sadly, the son of one of their friends had taken that very route, and the media wasn't exactly helping me dispel any of mom and dad's fears. Newspapers (as well as TV and radio news programs) gleefully sensationalized hippies, yippies, free love, campus unrest, and the senseless deaths of Janis, Jimi, and Jim. So, as the family's wannabe rock star, I was under firm but loving suspicion even before I had mastered the chords to "Wild Thing."

To the relief of scores of parents, however, music did not corrupt *all* of the eager souls who picked up instruments (and raised voices) in the wake of the Beatles and other seminal 1960s artists. In fact, the pursuit of technical and aesthetic competence forced musicians to accept some truths that would make even the strictest disciplinarian crack a smile: that it takes crushing amounts of devotion, sweat, and perseverance to do something creative; that true artists take full responsibility for the works they produce; and that inspiration is simply the spark that fuels an intense period of evaluation and revision. And whether musicians communicate through rock, jazz, or classical genres, the passionate ones soon learn that there is no such thing as "good enough."

I've found that this same dedication to excellence is sustained by **EM**'s collaborative ecosystem of readers, editors, authors, and gear manufacturers—and that's one huge reason why I adore working in music publishing. After all, you probably read **EM** because you want to *master* audio production. And I've found that most manufacturers are not business people married to the bottom line; they are music lovers who are truly obsessed with developing products that help musicians enhance their creativity.

For **EM**'s part, readers are well aware how much I gush over the staff's dedication. But the team was awesomely brilliant last year and, consequently, 1996 was tremendously successful. (Bigger issues, more ads, healthy circulation, etc.) Now, it would be easy for us to simply "repeat" the 1996 formula in 1997 and not risk losing much of our hard-won status as the premier mag for personal music production. But we're not going to do that because—remember?—there is no such thing as "good enough."

As a result, 1997 is going to be an exciting year. For one thing, the magazine just keeps getting bigger (the issue you're holding is 212 pages), better looking (thanks to our talented art staff), and more adept at defining the critical "whys" and "hows" of music production (kudos to our editors, including our brand-new editorial assistant, the impossibly tall multi-instrumentalist Rick Weldon). For another, this issue marks the debut of *JAM,* a hip, quarterly supplement for gigging musicians. In addition, 1997 will see the birth (finally!) of an **EM** Web site. We'll also produce the sixth edition of our award-winning *Digital Piano Buyer's Guide* and launch a new buyer's guide designed expressly for the home studio. Whew! No rest for us; big rewards for our readers. We wouldn't have it any other way. Enjoy!

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National Editorial, Advertising, and Business Offices
6400 Hollis Street #12, Emeryville, CA 94608
tel. (510) 653-3307, fax (510) 653-5142

East Coast Advertising Office
tel. (203) 838-9100, fax (203) 838-2550

Subscriptions, Customer Service
PO Box 41525, Nashville, TN 37204
tel. (800) 843-4086 or (615) 377-3322
e-mail sunbeltful@aol.com

Cardinal Business Media, Inc.
1300 Virginia Dr., #400, Fort Washington, PA 19034

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SYNCHING UP

Regarding Senior Editor Steve O.'s advice to Paul Doughty about his computer-ADAT sync question ("Letters: Toslink Info," December 1996): although it is true that the word-clock information will come directly from the Korg SoundLink card into the ADAT, if he is using a MOTU MIDI Timepiece AV, he will have word sync problems. The reason is this: the MTP AV must operate as the master clock to the ADAT, but the MTP AV does not have a word-clock input. This means that two clocks will be driving the ADAT—the word-clock signal from the computer card and the master transport clock coming from the MTP AV.

Paul won't have any problem if he is not trying to sync the transport of the ADAT to the computer (for example, if he is just dumping tracks onto the ADAT but is not trying to sync them to any tracks already on the ADAT). Of course, in this situation he is not using the MTP AV. The problem comes in when the computer and the ADAT need to be locked up together. In fact, he shouldn't have any problem sending tracks to the computer from the ADAT, it's only when he tries to send them back to the ADAT that he will get glitches in his audio.

I currently use the MTP AV with an ADAT, an ADAT XT, and an AI-1 as well as an Audiomedia III card in a Mac (we use *Digital Perfomer*). We ran into this problem trying to fly vocals over to the computer, process them, and send them back. The tracks were clean going into the computer, but while sending them back to the ADAT, we ran into the trouble I described above. I have spent a lot of time on the phone with MOTU and Alesis, and they agreed that the current gear could not do what we wanted.

Paul has a few options. First, he can get a MOTU Digital Timepiece, which has a word-clock input. Second, he could get a BRC for the ADAT, which also has a word-clock input. Both these options mean getting rid of the MTP AV; it's no longer necessary. Third, he could get a different card that has D/A converters and send analog audio back to the ADAT (eliminating the word-clock issue entirely). Fourth, he could get a house-sync generator and send house sync to the computer and the MTP AV. (It does have an input for house sync.) Unfortunately, every option costs money!

Although the MTP AV works great, putting an ADAT sync port on a MIDI patch bay leads people to believe the MTP AV is ideal for transferring digital audio between ADATs and computers. The omission of a word-clock input makes this impossible without additional gear that makes the MTP AV redundant. I hope this information helps.

Larry Clark
lclark1@aol.com

Jim Cooper of MOTU responds: *Larry Clark writes insightfully from what is clearly first-hand experience, and what he writes is true—but only for Audiomedia II and III users. According to my contacts at Korg, SoundLink can successfully transfer digital audio both to and from ADATs while slaving to its ADAT optical input. A MIDI Timepiece AV fits in just fine in this scenario as the word-clock master over the ADAT(s), with the ADAT serving as the word-clock master over the SoundLink card (via the optical connection). Because they are slaving to the same master (the MTP AV), everything stays in sync, and there are no digital audio glitches when transferring audio either way.*

Is the MTP AV required in this situation? As Steve O. pointed out, no—at least not for digital audio sync. But it does provide you with one additional capability: it lets you play, stop, rewind, and cue everything—including your ADATs—from your MIDI Machine Control-compatible sequencer, even when you are doing transfers between ADAT and SoundLink. This makes it easy to keep all the transferred music lined up with the other tracks.

Larry Clark's final observation about the MTP AV is understandable—given his experience with Audiomedia III, which doesn't have the ability to slave to word clock. But the fact is, the MTP AV works great as a master synchronizer for transfers between ADAT and many other hard-disk recording systems out there that can slave to word clock. Even the Korg SoundLink card has word-clock in/out, and we'll probably see other affordable, word-clock capable hard-disk recording systems on the market soon.

DO THE KARAOKE

I applaud two very important articles in your December 1996 issue: your cover story on selling music online ("Going Global") and your article on QuickTime 2.5 ("Desktop Musician: QuickTime Face-Lift"). However, both articles fail to touch base on an increasingly important common denominator to the World Wide Web and QuickTime: MIDI Karaoke files.

MIDI Karaoke files (or .KAR files) are SMFs with additional tracks containing text events. When a .KAR file is viewed through the appropriate software, lyrics and other pertinent information are displayed on the computer screen.

Although I don't expect the worldwide electronic music community to break into singing parties using this very popular Japanese pastime, I would like to see more discussion of the MIDI Karaoke format as a means to advertise music and pursue long-distance collaborations.

For example, songwriters could create SMFs and e-mail them to lyricists, who could in turn add lyrics to them using karaoke-compatible sequencers and send them back to the composer. Composers of works for the stage could add not only lyrics to an SMF, but also narrative and/or stage directions that could help the director better visualize

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the intent of the composer. A producer-songwriter in New York could send a MIDI Karaoke file to a singer in Los Angeles so he or she can learn a song well in advance of a recording session.

The .KAR file format is popular among PC users, and there are several sequencers that support adding karaoke tracks. (Check out the Computer Karaoke Web Pages at www.teleport.com/~labrat/karaoke.shtml.) Unfortunately, to the best of my knowledge, there is only one Macintosh sequencer that supports the format (*MIDIGraphy* shareware). So **EM**, consider taking a closer look at the MIDI Karaoke format in subsequent articles as yet another way for creative musicians to distribute their work to the masses.

Paco Ojeda
Boston, MA

JAZZ IT UP

I use an Iomega Jaz drive with the Jaz Jet PCI SCSI interface, which does double duty as a recording medium and as a mobile drive for my three Kurzweil K2000s. Is it possible to have the Jaz and the K2000 online with the PC simultaneously?

My Sonic Foundry *Sound Forge* 4.0 software refers to SMDI SCSI transfers but only using an Adaptec SCSI card. I have checked all my IDs, and they are fine. I am running the Jaz in auto-termination mode but have tried turning termination off. The Jaz is the flow-through device, although I have tried eliminating it and connecting the K2000 directly to the Jaz Jet SCSI card. The PC still does not recognize the K2000. Both systems operate independently quite well.

If this is possible, why couldn't I use the A/Ds on the K2000 to dump samples to the hard disk in real time? That would free up a slot in my computer.

Hank Lueck
Just Henry Productions
Leicester, NC

Hank—I'll answer the last question first because it is the easiest. With the current operating system, the K2000 can record only to its own RAM, not directly to an external device such as a hard drive. Saving to disk is a separate operation. Once the audio has been sampled to the K2000's memory (RAM), you have three choices.

First, you can transfer the digitized audio directly from the K2000's RAM to the computer's RAM via SCSI using the SMDI transfer protocol. That's the approach you have been attempting without success; let's try to make it work.

You didn't say whether you were using Windows 3.1 or Windows 95, which makes a big difference. According to sources at Sonic Foundry, if you are using Windows 3.1, the problem probably is that you don't have an ASPI driver for your Jaz Jet SCSI controller card. You'll have to ask Iomega whether such a driver exists. If Iomega hasn't written an ASPI driver for the Jaz Jet, you're screwed. With Windows 95, ASPI is not an issue; instead, you probably need an updated SCSI (MPD) driver for your card. Again, check with Iomega. Sonic Foundry recommends the Adaptec card because it is a de facto standard, and up-to-date drivers for it are widely available.

The second choice, which is much less desirable, is to transfer the sample to the computer via the MIDI Sample Dump Standard. This only requires a MIDI connection between computer and K2000 rather than a SCSI connection. But MIDI SDS transfers are terribly slow and are only recommended when you are desperate. You don't need this.

The third choice is relatively easy for Windows users, and it avoids your SCSI problem. The K2000 can read DOS-format disks and can save files in WAV and AIFF format as well as in its native KRZ format. First, format a Jaz cartridge as a DOS volume using your PC. Next, sample your sound to the K2000 and save your sample as an AIFF or WAV file to the DOS-format Jaz cartridge. I recommend using AIFF format because it supports sample loops; the K2000 supports the original WAV file specification, which does not support loops.

You can then mount the Jaz cartridge on your Windows machine and open the AIFF file directly into Sound Forge. *That gets around your SCSI problem because you aren't transferring the digitized audio from the K2000 to the computer's RAM. Your PC never has to recognize the K2000 as long as both the PC and K2000 can mount the DOS-format Jaz cartridge.—Steve O.*

SOUNDS OF SILENCE

I am trying to soundproof my garage from sound coming out/in. Although I live in a quiet neighborhood, there are a lot of airplanes flying in the background. The garage is separate from my house and has just a skeleton for a wall (outer wall and 2 × 4 support), so I have plenty of room to work with. When I asked for advice from insulation companies, they all said cellulose would do the trick. Is this my best option? I don't have a big budget to work with.

Billy Theriot
Sloth75683@prodigy.com

Billy—A thick layer of cellulose might be helpful as an adjunctive barrier but alone will not stop sound transmission from a bass amp or drum set, much less airplanes flying overhead.

The fact is, constructing a soundproof space is neither easy nor cheap. Your best approach is to build a complete, sealed room within your garage. By "complete" I mean that it must have not only four (or more) walls, but its own ceiling, door(s), and possibly floor. Furthermore, this room-within-a-room should be at least three or four inches away from the existing structure at all points. (The resulting air space between the two rooms is one of the "secrets" to reducing sound transmission.)

I highly recommend that you consult a good book on the subject. An excellent one is How to Build a Small Budget Recording Studio *by F. Alton Everest and Mike Shea, available from Mix Bookshelf at (800) 233-9604 or (908) 417-9575; fax (908) 225-1562. Good luck.—Brian K.*

ERROR LOG

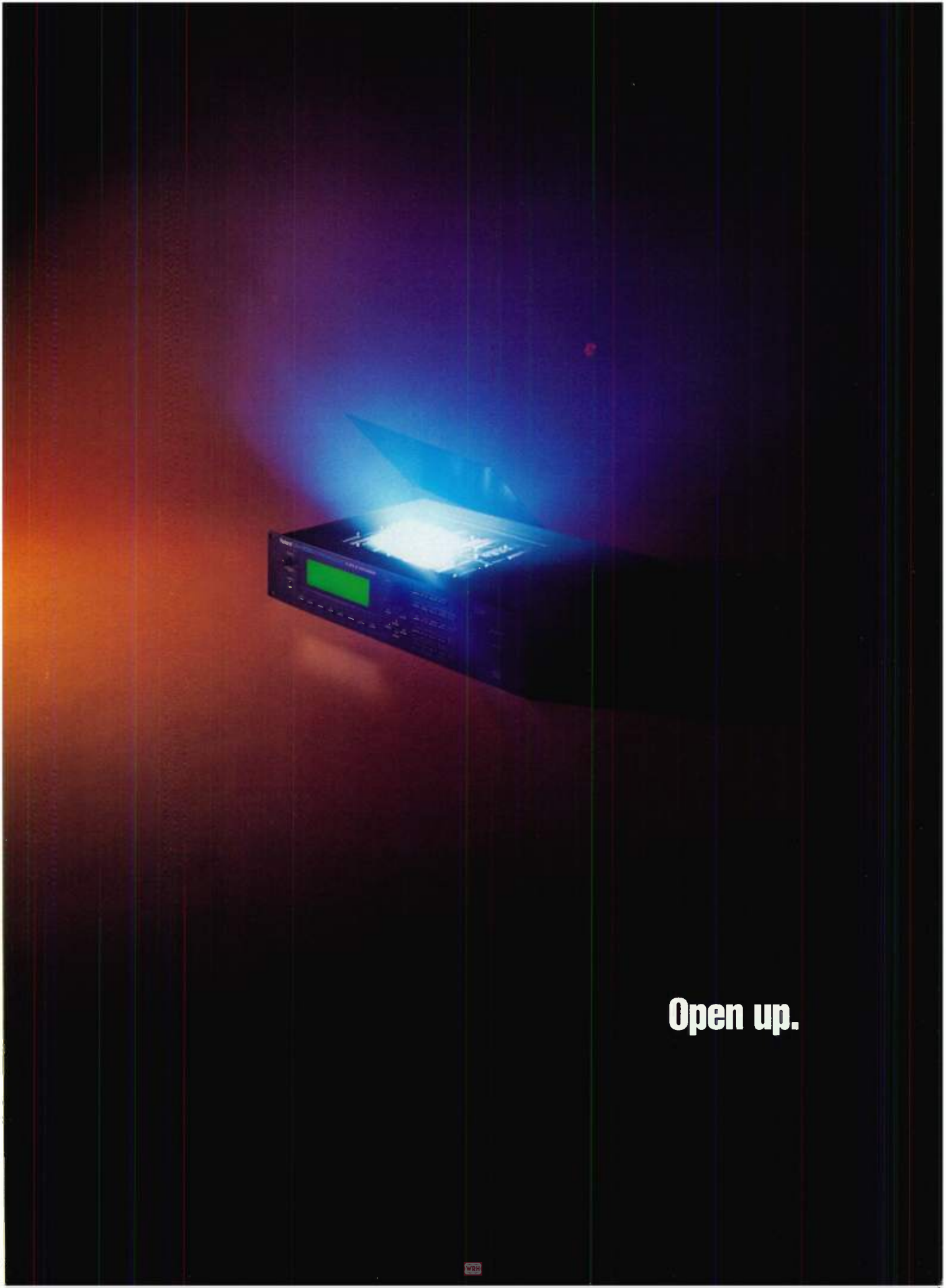
December 1996, "Tech Toys," p. 33: We inadvertantly gave the wrong phone and fax numbers for Z Right Stuff. The correct numbers are tel. (800) 520-4380 or (847) 520-4380; fax (847) 520-4212.

December 1996, "Desktop Musician: QuickTime Face-Lift," p. 94: We stated that "the internal synth kicks in when the external synth's polyphony is exceeded, which QuickTime determines from OMS." This is, of course, impossible, as OMS has no way of knowing anything about an external synth's polyphony.

December 1996, "Letters," p. 11: Although Emagic announced the intention of supporting SDII-format DSP plug-ins in *Logic Audio,* the company did not, in fact, add this feature.

WE WELCOME YOUR FEEDBACK.

Address correspondence and e-mail to "Letters," Electronic Musician, *6400 Hollis St., Suite 12, Emeryville, CA 94608 or emeditorial@pan.com. Published letters may be edited for space and clarity.*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Then say hello to Roland's new JV-2080 Synthesizer Module. It's the latest evolution in the world's most popular line of sound modules.

With 8 expansion slots, the JV-2080 sets a new standard for user-expandability, giving you access to more than 2,900 patches when fully expanded. To help you take advantage of all these classic Roland sounds, we've added the Patch Finder so you can hear and audition only strings when you need strings, separate your basses from your brasses, and quickly locate any of your favorite internal or expansion patches.

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WHAT'S NEW

By the EM staff

▲ POWER TECHNOLOGY FX•PACK

Now you can get the 32-bit, floating-point effects processing; intuitive graphic interface; and expandability of Power Technology's DSP•FX even if you don't have a Windows PC with available card slots. The FX•Pack ($1,299 with one card; $799 per additional card) is a 2-rackspace system that can hold up to four DSP•FX processing cards. It connects to a PC or Macintosh via serial port or ethernet. As it requires no card slots, it can even be used with a laptop computer. All DSP•FX effects plug-ins and hardware options, including the DSP•FX controller, can be used with the FX•Pack.

Power Technology also announced the release of the *ParaEQ* plug-in ($199) for DSP•FX. It features stereo equalization with high and low shelving sections and four parametric bands with center-frequency ranges of 20 Hz to 20 kHz, Q range of 0.1 to 100, and up to 30 dB of boost/cut. The graphic interface is modeled on familiar, knob-based EQ panels, and a display gives you the left and right channel composite frequency responses. Power Technology; tel. (415) 467-7886; fax (415) 467-7386; e-mail dspfx@dspfx.com; Web www.dspfx.com.

Circle #401 on Reader Service Card

▼ LEXICON MPX 1

If you thought fully configurable, true-stereo (not just discrete-channel) multi-effects processing was a luxury beyond the reach of personal studios, think again. Thanks to the MPX 1 ($1,299), home-studio owners can now process their tracks with effects that won't compromise the temporal and spectral differences inherent in opposite channels of stereo source material.

The 1U rack-mount MPX 1 features 56 effects distributed among six effect blocks. There's a mono, dual, or stereo pitch shifter; a chorus block with flanging, phasing, and comb filtering; and a modulation block that offers auto-panning and tremolo. In addition, you get mono, stereo, or dual delay and ambience, chamber, hall, plate, or gated reverb. The sixth block is an EQ that can be 4-band mono or 1- or 2-band stereo. Each effect block is independent, with its own mix, level, and bypass controls.

You can use the effects individually or simultaneously chain up to five effects. A routing function lets you distribute the effects blocks across two stereo paths that can be split and/or merged at multiple points. These operations use a simple drag-and-drop procedure on a visual map in the LED display window.

The MPX 1's front-panel interface is designed to be user-friendly. For example, each effects block can be enabled/disabled with a dedicated button that is equipped with a status LED. A multilevel help system provides guidance through any operation. In addition, a numeric display shows program and patch numbers and an alphanumeric display shows parameters, values, and effects routings.

The MPX 1 offers 200 presets and 50 user registers. Programs can be sorted by user-definable name or number and searched for by source type (such as drums, vocals, and live P.A.), effect type, or both. Fine tuning of presets is accomplished through a definable "soft row" of the most useful parameter combinations for each program, and Edit mode gives access to a more extensive pool of parameters.

Other front-panel features include input and output controls, dual 6-segment input-level meters with clip light, a tap-tempo button, an A/B button (assignable as a continuous controller) for morphing between effects or parameters, a value knob, and bypass. Dedicated Mix and Patch buttons give access to mix, level, and patch settings of all effects. An Options button offers still more features.

You can MIDI control all effects parameters, the A/B function, bypass, levels, and more. All internal control sources (audio levels, LFOs, ADRs, pedal, and so on) can be transmitted as MIDI Control Change messages. Tempo-based parameters can sync to incoming MIDI Clock, or the MPX 1 can transmit MIDI Clock based on the front-panel tap-tempo feature.

The unit's rear panel offers balanced XLR and ¼-inch TRS analog I/O, S/PDIF digital I/O (at 44.1 kHz), footswitch and footpedal jacks, and MIDI In, Out, and Thru. The A/D converters are 18-bit and DACs are 20-bit. Operating levels range from -14 dBu to +20 dBu. The internal, universal power supply automatically switches to the correct voltage range. Dynamic range is rated at 95 dB. Lexicon; tel. (617) 280-0300; fax (617) 280-0490; e-mail info@lexicon.com.

Circle #402 on Reader Service Card

The Power Without the Pain

DARWIN 2.0

DIGITAL AUDIO DISK RECORDER

SOUND ON SOUND — *"Friendly, familiar tape-like operation and superb audio quality."*

ELECTRONIC MUSICIAN — *"E-MU's modular hard disk recorder offers great sound and a graphic LCD that makes editing and track management easy."*

EQ — *"Makes 8 tracks feel like a whole lot more – The Darwin/ADAT combination delivers the goods."*

RECORDING — *"Darwin appears as another ADAT to a BRC"*

DARWIN OPTIONS:

Seamless interface with ADATs and BRC. Adds 800 virtual tracks to an ADAT

DSP option card provides time compression, state of the art Poly-Gender™ pitch change, and gain control

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Jaz drive: records to inexpensive, removable media, eliminates the need for back-up between projects

SEE US AT NAMM BOOTH #3340

FEATURES:

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- 16 layers of undo
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REV UP! ▲ ▲ ▲ ▲

▼ OPCODE

Opcode's *Studio Vision Pro* 3.5 for Macintosh ($995; upgrade price tba) features Power Mac native code, new tools for multimedia authors, new DSP effects, and assorted other goodies. The new version supports Adobe Premiere audio plug-ins. It has an integrated QuickTime movie window and reads and writes QuickTime audio and MIDI files, Red Book audio, and WAV, AU, AIFF, and SDII files.

Support for TDM systems has been expanded, including support for Pro Tools III PCI, SampleCell II, and TDM busing. (Eight buses with line inserts are available from the program's consoles.) The number of Audio Instruments available has been increased from 16 to 64, and up to 64 channels of I/O are available in Pro Tools III PCI systems. For users of Sound Manager, an improved interface allows twice as many audio tracks as earlier versions—at least eight on a Power Mac—with tighter sync.

Studio Vision Pro's pitch shifting now incorporates formant-based shifting for changing pitch independently of the timbral characteristics of vocals and works on polyphonic audio. Timbres can also be altered without changing pitch. A new resynthesis effect called Audiomorf is also included.

Other improvements include integrated MIDI Machine Control, OMS 2.0 Timer support (which lets other OMS-compatible applications control the program with frame accuracy), real-time fader grouping in the Console window, and vertical zoom in the Track Overview, with four preset zoom sizes. Opcode Systems; tel. (415) 856-3333; fax (415) 856-0777; e-mail info@opcode.com; Web www.opcode.com.

Circle #403 on Reader Service Card

▼ BIAS

You might think the folks at BIAS would take a break after introducing their debut product to much fanfare, but the release of *Peak* 1.5 ($499; upgrade $129) for Macintosh shows they've been hard at work. The new version lets you create RealAudio 3.0 and 2.0 files for use on the Internet. It also includes a Batch File Processing feature that will apply almost any operation that *Peak* can perform to multiple files.

A new Playlist function supports external DSP plug-ins and allows customizable crossfades and synchronization with the program's editing windows.

The main editing window now includes an optional Overview window for navigating large files, and it displays SMPTE time-code units (sync to time code is planned for a later release).

Peak 1.5 supports standard SD II audio regions. The quality of the Change Pitch and Change Duration functions has been improved. Some operations have been simplified, and you can now open multiple files from the Open dialog or by dragging and dropping entire disks or folders onto the *Peak* icon. The interface has also been enhanced with a sporty, 3D look. BIAS; tel. and fax (415) 331-2446; e-mail sales@bias-inc.com; Web www.bias-inc.com.

Circle #404 on Reader Service Card

▲ STEINBERG

Steinberg released *WaveLab* 1.5 for Windows 95 and NT ($499; upgrade $99), adding support for 24-bit and 20-bit digital audio and incorporating a new Realtime Engine that lets you run the signal through up to six real-time processes. Although the program will run on an 80486, a Pentium 90 is required for the Realtime Engine, and some plug-ins require faster CPUs. *WaveLab* 1.5 supports Microsoft's ActiveMovie API, and plug-ins for this format will run within the Realtime Engine.

A number of plug-ins for *WaveLab* 1.5 are also available from Steinberg, some of which were previously released as TDM plug-ins. The *Externalizer* plug-in ($399) is for use when monitoring with headphones; it simulates a listening environment with a defined set of speakers so you can hear the location of instruments in the mix. *DeNoiser* ($399) is a noise-reduction plug-in that automatically detects the noise floor of single tracks or complete recordings.

Steinberg's *DeClicker* plug-in ($399; TDM version $799) not only removes clicks from vinyl records but also removes digital clicks and static discharge cracks from DAT masters. The *Loudness Maximizer* plug-in ($399; also available for *Cubase VST*) is designed to dynamically optimize your signal level, pushing it consistently up to 0 dB. Steinberg North America; tel. (818) 993-4091; fax (818) 701-7452; e-mail steinberg@aol.com; Web www.steinberg-us.com.

Circle #405 on Reader Service Card

How to buy your next reverb without listening to it.

The first criteria in choosing a reverb is always sound quality. Everyone claims they sound the best but ONLY the Yamaha REV500 has 32-bit internal processing and 20 bit A/D and D/A conversion. Everyone else has 24-bit or less processing and 16 or 18-bit conversion.

The difference between 32-bit and 24-bit is audible: 32-bit processing gives you significantly smoother decay so your reverb fades imperceptibly into "silence." The difference between 20-bit and 16/18-bit conversion is also dramatic; 20-bit A/D D/A means the REV500 is exceptionally quiet, eliminating noise in recordings and adding headroom in live applications.

Stereo professional +4dB/-10dB balanced inputs and outputs.

The second criteria is the actual effects—what are they and how do they sound? The REV500 includes 100 top quality halls, rooms, plates and special reverbs. These simulations of actual environments place your signal in "real" locations for unparalleled realism. And the REV500 also gives you 100 spaces to store your own settings.

The third criteria is the ease of accessing the effects. REV500 true stereo halls, rooms, plates and special reverbs are instantly recalled from the front panel. The REV500 has front panel controls for reverb time, pre-delay, high-frequency decay and early reflection level. If you want to change these settings, just turn that control and the LCD instantly shows a picture of what's happening. You can **see** what you're hearing.

If you want to hear it, too, only Yamaha has built-in drum samples so you can hear the effects without an external sound source. Just hit the "audition" button as you select and customize your effect.

The fourth criteria is price. Once again, the REV500 delivers like no one else: $499 for a professional reverb. But if you'd rather pay more, you can buy the $1299 Yamaha **ProR3** which has the same 32-bit engine and 20-bit A/D D/A with a keypad for calling up effects, two types of EQ and a gate which can be placed around any reverb.

Now that you know enough to buy the REV500 sight unseen, we recommend you go listen to it. The sound quality will blow you away. Call (800) 937-7171 ext. 680 to find the Yamaha REV500 dealer nearest you.

Large LCD display shows effect type, name and current settings.

Turn the reverb time control, see the reverb time displayed graphically.

Turn the HI-RATIO control, the REV500 instantly shows high frequency decay.

EXT. 680
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REV500

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Audiowerk8, Emagic's new PCI based digital audio recording card, makes hard disk recording as easy as tape. With 8 discrete outputs, stereo analog ins and digital I/O, solutions are solutions without exceptions. Shipping with VMR, the "Virtual Multitrack Recorder", software so transparent, the manual is included in this ad. And for your growing studio needs, Audiowerk8 is fully compatible with the Logic Audio production system. The choice is simple, with a list price of $ 799.– creative expression with professional results is now affordable to all.

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- Up to 20 Tracks with Logic Audio Software
- PCI Busmaster Technology for maximum System Performance

Mac OS

MacOS and Windows '95 compatible.

1. Tracks
Play back eight tracks while recording two additional tracks. You can name each track and access a virtually unlimited number of alternative takes.

2. High Definition Level Meters
Allows you to accurately monitor the levels of your recordings on each track.

3. Left and Right Locator Displays
Shows you the currently selected in/out points. Values can be easily edited with click/drag mouse operation. Values can be readily selected and dragged into any of the 20 positions in the Position Memory bank and vice versa.

4. Cycle Button
Enables cycle playback and record between the left and right locators.

5. Track Switches
Besides the standard switches such as Solo, Mute and Record, the Stereo Buttons allow you to group 2 tracks together as one track for easy handling.

6. Wave Display
The positive waveforms of your recorded tracks smoothly scroll from right to left during playback and recording. This allows you to easily navigate through your recordings.

7. Set-Locator Button
These buttons allow you to write the current "tape" position into the locator displays on the fly. Controllable either with the mouse or with a keystroke, these buttons allow you to quickly generate new left and right locators.

8. Tape Indicator
Indicates how much space or "tape length" you have left for your recording. It also indicates your current position within the recording.

9. Auto Drop Button
With this button enabled, the VMR will automatically switch to record mode using the currently displayed left locator as record IN and the right locator as record OUT.

10. Tape Button
You'll need to choose a "tape" before making your first recording. Pressing this button gives you a variety of tape length options depending on the size of your hard drive.

11. Copy Button
This button allows you to copy, move or mixdown the data of "soloed" source tracks between the left and right locators to the tape position on a record enabled destination track.

12. Input Selector
Press "A" to choose the Analog Input, or "D" to choose the Digital Input on your Audiowerk8 Card.

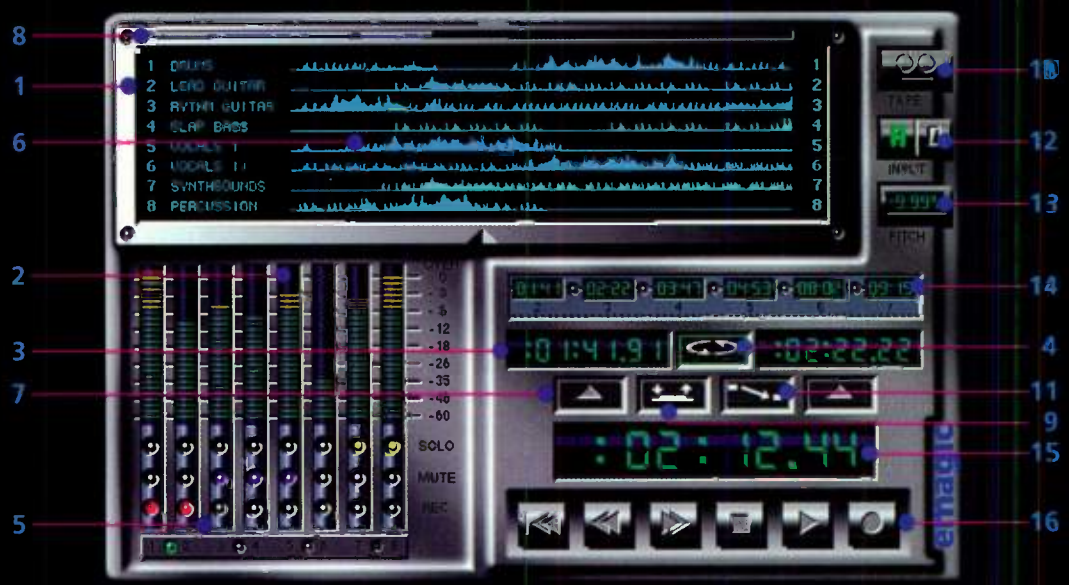
13. Pitch Variation
Clicking on this button opens a display where you can simply enter the desired pitch value by clicking and dragging. Pitch ranges are from -9.99% to +9.99% in steps of 0.01%.

14. Position Memory Bank
Up to 24 positions can be stored with each tape. 6 can be displayed simultaneously. To view others, simply click on the slider and drag left or right. Clicking on the "pair" button between 2 position memories gives you a "pair selection" which can be dragged into both locator displays simultaneously. Position memories can also be set and recalled via keystrokes.

15. Position Display
Optimized for legibility, the position display shows you the current tape position. A maximum tape length of 1 hour, 59 minutes and 59.59 seconds can be displayed.

16. Tape Controls
As simple to use as the controls of any conventional multitrack tape recorder.

VMR "Virtual Multitrack Recorder" Software, included with Audiowerk8, Emagic's new 8 Channel Digital Audio Recording PCI Card.



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The Audiowerk8 will be available in early March at leading music retailers. The suggested retail price is **$799.–** including the Virtual Multitrack Recorder Software. For more information about Audiowerk8, system requirements and all available upgrade options please visit our webpage or give us a call.

circle #535 on reader service card

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Fax +916. 477 1052

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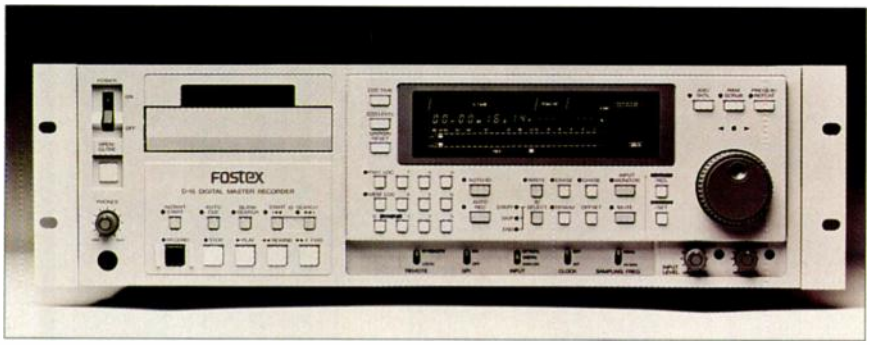


▲ FOSTEX D-15

Until Fostex introduced the D-15 DAT recorder ($3,295), only very expensive DAT decks had the ability to sync to time code. The D-15 offers sync to time code and chase-lock capabilities via the optional 8335 T/C external sync card ($595). Fully loaded with the 8335 sync card and the optional 8336 RS-422 9-pin machine control card ($195), the D-15 provides maximum DAT-machine capability for under $4,100. Compared to the cost of its predecessors, that's a bargain.

The unit offers several improvements over the Fostex D-10, including multiple reference levels (-12, -18, and -20 dB), cue levels (-20, -30, -40, and -55 dB), and cue times (1500, 1200, 900, 600, and 300 ms). The D-15 features 18-bit, 64x oversampling inputs and 20-bit, 8x oversampling outputs, I/O trim pots, a Reference Play mode, a panel-lockout switch, and an adjustable Offset command. The transport has been improved and can shuttle from one end of a 120-minute tape to the other in less than 60 seconds. In addition, the Fostex D-15 sports a 37-pin D-sub parallel remote port (for remote-control operation) and front-panel calibration controls.

But what really sets the D-15 apart from the DAT pack is its ability to incorporate time-code chase-lock and 9-pin remote machine control via the expansion cards. The fully configured D-15 can read externally generated time-code data from a number of sources (including video time-code black burst and color bars) and can record time code simultaneously with audio. The 9-pin remote machine-control protocol allows remote access to the D-15 from video-editing systems for all location functions except varispeed. (The 8336 card must be used with the 8335 card.)

The unit also provides balanced XLR and unbalanced ¼-inch connectors that operate at +4 dBm and -10 dBV, respectively. Both 48 kHz and 44.1 kHz sampling rates are supported. Fostex rates the D-15's THD at <0.05% (1 kHz @ +4 dBu) and signal-to-noise ratio at >92 dB. Fostex Corporation of America; tel. (310) 921-1112; fax (310) 802-1964; Web www.fostex.com.

Circle #406 on Reader Service Card

▶ FOCUSRITE GREEN RANGE

Although the Flintstone-like contours of their cast-aluminum cases may cause them to resemble artifacts from the Stone Age, the new Focusrite Green Range processors are anything but primitive. The series currently comprises the Green 1 Dual Microphone Preamp ($1,099), Green 2 Focus EQ ($1,249), and Green 3 Voicebox ($1,349). Each module is internally powered and 1U rack-mountable.

The Green 1 Dual Mic Preamp features a variable gain of +10 dB to +60 dB, 48V phantom power, phase-reverse switches, a highpass filter with a 12 dB/octave rolloff at 75 Hz, a clip light, and a mute switch that can be operated manually from the front panel or triggered externally via a jack socket. The unit offers balanced XLR I/O. Frequency response is rated at 10 Hz to 150 kHz at +10 dB gain and 10 Hz to 50 kHz at +60 dB. Equivalent input noise is rated at -128 dBu and distortion at <0.001% @ 1 kHz.

The Green 2 Focus EQ is a multi-input, single-channel unit offering six stages of equalization: low-mid and high-mid parametric, low- and highpass filters, and low and high shelving. The two parametric mid bands offer frequency, cut/boost, and Q (filter bandwidth) controls. The low-mid filter can range from 40 to 400 Hz or 120 to 1,200 Hz (switchable); the high-mid band can be 600 to 6,000 Hz or 1.8 to 18 kHz. The lowpass filter ranges from 4.7 kHz to 30 kHz and the highpass filter from 10 Hz to 330 Hz, with fixed 12 dB/octave slopes.

The low and high shelving filters provide ±18 dB cut/boost with variable rolloff frequencies: 30 to 500 Hz for the low shelf and 3 kHz to 18 kHz for the high shelf. These filters can be switched from shelving operation to bell curve.

The input stage features a balanced XLR mic input as well as XLR and ¼-inch jacks for line (0 dB gain) or instrument (+20 dB gain) inputs with ±12 dB of trim. A 10-LED bargraph provides VU metering of input levels.

The Green 3 Voicebox packs a microphone preamp, EQ section, and dynamics processor into a single box. The mic preamp is the same as the one on the Green 1 Dual Mic Preamp. The EQ section features a mid semiparametric band with a gentle bell curve that can be switched to a tight notch. High and low shelving filters provide both frequency and gain control. The dynamics section offers compression, de-essing, and a noise-reducing expander. All three sections can be switched in and out of the signal path, and gain reduction can be metered via the dual-function 10-LED input-level bargraph. Group One Ltd. (distributor); tel. (516) 249-1399; fax (516) 753-1020.

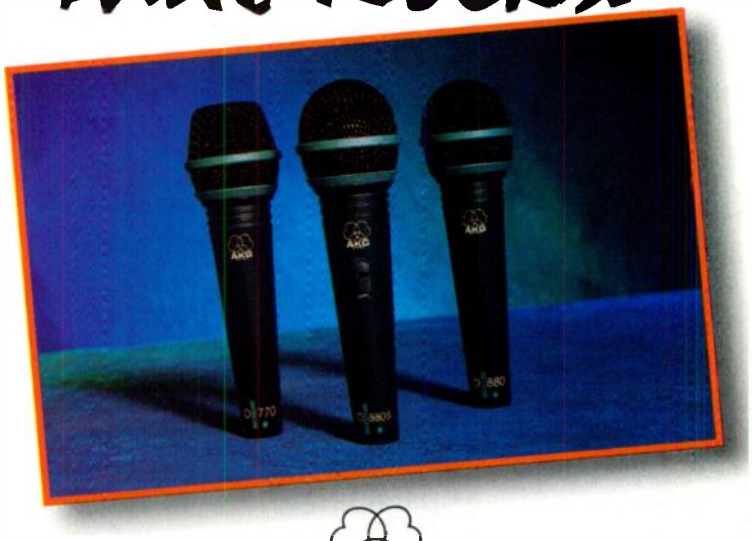
Circle #407 on Reader Service Card

Emotion.
The Sound That Rocks.

The accepted standard for music microphones has suddenly been challenged with the introduction of AKG's Emotion microphone series. AKG's new design and manufacturing approach, known as Tiefzieh Varimotion Technology™ (patent-pending), optimizes a microphone's response while delivering killer sound at a price that will blow you away.

Neodymium magnet assemblies also mean that these mics produce some serious output, ensuring that your vocals soar over even the heaviest rhythm section and your instruments retain their crisp presence without getting stuck in the mud. With AKG's Doubleflex™ antivibration system, handling noise is all but eliminated, while superior gain-before-feedback prevents your PA from howling even when the guitars crank up to eleven.

Experience the new Emotion series at your local AKG retailer and add an entirely new dimension to your performance, as Emotion delivers "The Sound That Rocks" for a lot less money than you would expect



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101ST AUDIO ENGINEERING SOCIETY CONVENTION ▲ ▲ ▲ ▲

After an absence of six years—not nearly long enough, according to some attendees—the Audio Engineering Society convention returned to Los Angeles. Fortunately, the L.A. Convention Center proved a welcome oasis in the middle of the grim wasteland that lies at the heart of that wounded city. Showgoers didn't seem the least fazed by the necessity of going around armed to the teeth—with bags of product brochures, that is.

Outside the doorway of the main hall, we were greeted by a large sign announcing that, among other things, the use of recording devices was banned. I waited with morbid curiosity for security to eject TASCAM, Alesis, Akai, Fostex, and their ilk. But the security staff must have been too busy browsing the booths to give any of these blatant violators of hall policy the old heave-ho; at any rate, we didn't spot any DAT decks or hard-disk recorders in the street. (Darn!)

As with last year's AES convention, this year's show was less about new technology—though there was some of that—than about further development of recently established technology. There were lots of new products, but the most immediately interesting for the **EM** community were affordable digital consoles; small, self-powered, near-field reference monitors; and DSP plug-in software.

There was also a lot of excitement over the arrival of Liquid Audio, the real-time audio streaming technology for the Web (discussed in the January 1997 "Tech Page"). In fact, there were so many goodies of interest that you'll be reading about them in these pages for months to come. So send in those subscription renewals right away, and let's get to it!

SPEAKING OF WHICH...

Powered monitors really are on a roll. Event Electronics and Genelec led the way with miniature near-field speakers that combine with matching subwoofers to deliver full-spectrum, pro-quality sound. Event's Tria system includes the subwoofer; the 1091A low-frequency component for Genelec's 1029A speakers is a separate purchase.

Spirit and Mackie also joined the self-powered speaker parade (see on p. 28 "Active Duty"). So did HHB Dynaudio Acoustics, and Audix, which showed improved versions of its PH-15 and PH-25. Bag End continued to plumb the sonic depths, introducing the Infrasub-18, a 400W, powered subwoofer that uses Bag End's ELF (Extended Low Frequency) technology to get all the way down to 8 Hz.

CONSOLE YOURSELF

After warming up with the ProMix 01 in 1995, Yamaha made a big splash in 1996 with its fine-sounding, under-$10,000 02R digital console (reviewed in the July 1996 **EM**). Recognizing that many project studios are on a tight budget, Yamaha unveiled the 03D. With a price tag under $4,000, this new board could turn out to be the digital console of choice for the budget project studio.

Like the 02R, the 03D is packed with features. It offers 24 mono input channels, which are accessed by sixteen sets of 60 mm motorized faders and Select and On/Off buttons. The first eight sets of faders and buttons can be switched between channels 1 to 8 and channels 17 to 24; the second eight control either channels 9 to 16 or four of the six aux sends and the unit's four subgroups. The other two aux sends are dedicated to the 03D's two independent, stereo effects processors.

All 24 mono channels have balanced ¼-inch line inputs, but the first eight channels also have XLR inputs, microphone preamps, and individually switched phantom power. The first two channels have insert points, and the first sixteen channels have direct outputs. Almost every input and output on the unit—not just the channels—has an independent, 4-band parametric EQ and a dynamics processor. There also is a stereo input channel without all the fancy features, which goes to the main L/R bus for use as a 2-track return or extra mixdown inputs. All analog I/O connections use 20-bit digital converters. (Internal processing is 32-bit.)

The 03D has one YGDAI card slot, which accepts the same digital I/O cards as the slots in the 02R. That means you can have eight channels of AES/EBU, S/PDIF, Yamaha Y2, ADAT Lightpipe, or TASCAM TDIF (DA-88/DA-38) digital I/O.

Everything on the 03D is programmable with onboard Scene and dynamic (real-time) automation, which is referenced to MTC. Most parameters also can be controlled in real time via MIDI Control Changes or SysEx. The unit includes an onboard Mac/PC serial connector that acts as a direct MIDI interface, and it can send MMC.

That's not all; the mixer can output LCRS surround sound, which is great for audio post-production. You view your settings with a backlit, 320 x 240-pixel fluorescent display that features graphical icons, and a mouse port lets you navigate and edit parameters with any PC-compatible mouse. You can double your pleasure by cascading two 03Ds or an 03D and 02R. It appears we will see 03Ds hopping off the store shelves and into studios everywhere right around Easter Bunny time.

Although the 03D was the coolest affordable (by **EM** standards) digital mixer

Yamaha 03D

that was in working order on the show floor, it wasn't the only digital console in this class. As expected, Korg showed their RC168 digital mixer (described in "NAMM's Greatest Hits" in the May 1996 **EM**), which is now shipping. But TASCAM caught everyone by surprise with its attractive, if nonfunctional, prototype digital console, which the company showed in order to get industry feedback.

The prototype has two rows of faders in an over/under configuration (24 above, 24 below). In the top row, sixteen faders control the XLR mic/line analog inputs, which can be switched to TDIF digital format. Channels 9 to 16 can be switched to AES/EBU, and channels 9 to 12 can also use S/PDIF. Eight faders provide control for two multipurpose stereo returns and six stereo aux returns (five analog, one switchable analog or AES/EBU). The lower 24 faders provide control for three TDIF-format tape returns.

The console will offer six aux sends, two of which can be analog or AES/EBU. You'll be able to automate the beast via MIDI or using dedicated computer software. The mixer's integrated transport control area supports TASCAM sync, MMC, Sony 9-pin, and GPI. The user interface features a backlit LCD and twenty rotary encoders. The console is expected to ship this summer, and the price will be "well under $10,000."

TURN ON, PLUG IN, DROP JAW

Plug-ins continue to be the largest single new-product category for the electronic musician. Having just announced a slew of TDM plug-ins in our January issue, we'll be back next month with a passel of major new ones shown at AES.

In addition to the many TDM introductions, conventioneers witnessed the birth of Digidesign's new (albeit long expected) AudioSuite plug-in architecture, which ships as part of *Pro Tools* 4.0. AudioSuite looks like an improved version of the Sound Designer II plug-in architecture, improved mostly because it runs on a Power Mac with Sound Manager and has a new graphic interface for crossfades that includes custom fade curves.

Mark of the Unicorn is among the first third-party developers to have an AudioSuite plug-in ready to rock. MOTU showed its *PureDSP* plug-in, which features time-scaling and both conventional and formant-based pitch-shifting. Waves also has announced support for AudioSuite.

But quietly bubbling beneath all the surface din surrounding the gigantic Digidesign Development Partners complex of booths was a fascinating undercurrent. There was plenty of TDM support, of course. But the big buzz regarding non-TDM plug-ins was not about AudioSuite. No, the buzz was about Steinberg's Virtual Studio Technology (VST) architecture for the Mac. Why? Because unlike AudioSuite, Sound Designer II, or Premiere plug-ins, *Cubase VST* plug-ins can process audio in real time on the Power Mac with Sound Manager, and no DSP hardware is required. That's what desktop musicians want, and plug-in developers know it.

Similarly, I heard a lot of talk at the various Windows software-developer booths about plug-ins for the Microsoft ActiveMovie API, which *Cakewalk* and *Sound Forge* (among others) will support. Waves has already announced it will develop ActiveMovie-format versions of its plug-ins.

SHIFTING SAND

When Mark of the Unicorn added formant-based pitch shifting to *Digital Performer,* it rather soundly boxed its competitors' ears. (Formant-based pitch shifting allows you to transpose an audio region without affecting its basic tonal content, thus avoiding the "munchkinization" effect.) MOTU also applied this technology in its new *PureDSP* plug-in for AudioSuite.

Now, the rest of the industry is scrambling to catch up. Not only has MOTU's arch-rival, Opcode Systems, added formant-based pitch-shifting to *Studio Vision Pro* 3.5 (see on p. 20 "Rev Up!"), but E-mu has also added this capability to its Darwin modular hard-disk recorder via an optional DSP card, Analog Devices has implemented it in its SHARC chip (see "Tech Page" on p. 210), and Wave Mechanics offers it in its *PurePitch* TDM plug-in (see the January 1997 "What's New"). Several other companies also declared themselves advocates of this pitch-shifting approach. In almost all cases, you can still opt for traditional pitch-shifting if you prefer.

—Steve Oppenheimer

TASCAM Digital Mixer

ACTIVE DUTY ▲ ▲ ▲ ▲

▶ EVENT ELECTRONICS TRIA

Designed for computer users and others who work in tight spaces, Event's Tria Monitor System ($799) includes two magnetically shielded, satellite speaker cabinets and a floor-mounted subwoofer station housing five power amplifiers and a floor-loaded 8-inch bass driver. The front-ported satellites are constructed of medium-density fiberboard and feature 5.25-inch, polypropylene-cone woofers; one-inch, neodymium, soft-dome tweeters; and power-on/clip indicator lights. The subwoofer module features trim and level controls and balanced, gold, combination ¼-inch/XLR input connectors. The outputs and multipin amplifier-interconnect cables (provided) also use gold connectors.

Each driver in the Tria system has its own active, asymmetrical crossover and power amp: 37W for tweeters, 75W for midrange woofers, and 100W for the sub. Continuously variable trim controls for each amplifier are calibrated in 1 dB increments and allow the user to compensate for speaker placement and personal tonal preferences. Frequency response is rated at 45 Hz to 20 kHz. Event Electronics, Inc.; tel. (805) 566-7777; fax (805) 566-7771; Web www.event1.com.

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GENELEC 1029A/1091A

Genelec's 1029A/1091A close-field monitoring system (less than $1,800/pair) includes a pair of 1029A biamplified speakers as well as the 1091A matching subwoofer. The 1029As are housed in die-cast aluminum enclosures that provide heat dissipation and RF-interference shielding. Each monitor utilizes a pair of 40-watt amps powering a 5-inch woofer and ¾-inch, hard-domed tweeter. A volume control is mounted on the front, and bass and treble room-response controls are on the rear. The speakers are also shielded against electro-

magnetic interference.

The matching 1091A subwoofer features an 8-inch driver enclosed in a dual-vented, medium-density fiberboard cabinet. The input levels are handled by the gain controls on the 1029As. A bass-attenuator switch on the 1091A's 70W amplifier module lets the user balance the subwoofer output to complement the room response. The 1091A extends the system's low-frequency response to 38 Hz. Genelec Inc.; tel. (508) 440-7520; fax (508) 440-7521; Web www.genelec.com.

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▼ MACKIE DESIGNS HR824

Mackie's HR824 biamplified near-field monitors ($1,498/pair) use an 8.75-inch woofer with a mineral-filled polypropylene cone powered by a 150W amp and a 1-inch, aluminum-dome tweeter powered by a 100W amp. The drivers are housed in a high-resin, wood-composite enclosure. Free-field frequency response is rated at 39 Hz to 22 kHz, and maximum peak sound pressure level is rated at 121 dB.

The HR824 utilizes a Linkwitz-Riley, 24 dB/octave electronic crossover and

an electronic limiter section with independent low- and high-frequency limiting. The monitor's back panel features an input-sensitivity control; an acoustic-space switch with settings for quarter, half, and whole space (to compensate for different speaker positions); an 80 Hz rolloff switch that lets the monitor simulate a home bookshelf speaker; a high-frequency switch that provides ±2 dB cut/boost; and a power-mode switch that can be set to off, on, or auto-on.

Balanced XLR and ¼-inch TRS input connectors are provided. These are positioned vertically (pointing downward) so the monitors can be mounted flush to a wall without causing damage to cables or connectors. Mackie Designs; tel. (800) 898-3211; fax (206) 487-4337; e-mail sales@mackie.com.

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SPIRIT ABSOLUTE 4P

Close on the heels of the Absolute 2 passive close-field monitors comes Spirit's Absolute 4P biamplified monitor system (approximately $1,000/pair), which utilizes a 6.5-inch woofer and a 1-inch, ferrofluid-cooled, soft-dome tweeter. Each driver is powered by its own 100W, thermal-protected amplifier. The vented, medium-density fiberboard cabinet features a large, flared port for extended low-frequency response.

The Absolute 4P features a constant-voltage active crossover with time compensation to assist driver alignment. The monitor also features a long-throw low-frequency driver, magnetic shielding, limiting protection for both drivers and amps, a switchable 40 Hz highpass filter, and a stepped input-level control for accurate setup. Frequency response is rated at 35 Hz to 23 kHz and maximum peak SPL at 109.5 dB. Spirit By Soundcraft, Inc.; tel. (916) 888-0488; fax (916) 888-0480; Web www.spirit-by-soundcraft.co.uk. ●

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Gimme a...

Compressor

The compressor in the 286A is a patented hybrid feed-forward/feed-back design that was engineered by a guy named Bob Orban. (His stuff is used on almost every radio station on the face of the planet). Using the classic dbx VCA (the heart and soul of a compressor) the 286A compressor is very easy to use with its drive and density controls. Drive is the amount of signal sent into the compressor and the Density is a combination of controls that allow you to achieve anything from a nice transparent gain reduction all the way to a fat squashy compression so popular on heavy rock vocals. The 8 stage LED meter gives you a great visual indication of how much the compressor is working on your vocals or whatever else you may run through your 286A. Now's actually a good time to tell you that you're not stuck just using the 286A as a mic pre, it's got a line input so you can use the 286A's 5 processors on any of your audio: guitars, keyboards, drums - anything!

Expander/Gate

Background noise in the room, sound leakage from headphones, vocalist breathing etc. are all problems that can crop up while recording your vocals. These situations are all easily taken care of with the 286A's Expander/Gate. This multi-use design allows you to gate out any noise during breaks in the intended signal and when used as an expander, will push down any unwanted noise in the signal such as headphone bleed.

Enhancer

Another very cool device in the 286A is the Enhancer. The low frequency detail is a very trick circuit that not only adds warm low end but fattens up the signal by cutting out some of the mud in the lower mid-band at the same time! The HF (high frequency) detail adds sparkle to your signal. Now if you're thinking that the sparkle that you add with the Enhancer is going to put back the high frequency sibilance the de-esser is taking out, think again. The two circuits are tied together in an ingenious (and patented) way such that they work in tandem to do both jobs beautifully!

De-esser

One of the biggest problems with processing vocals are those pesky "ssssssss" sounds. Our dbx de-essers are in use in virtually every major recording studio in the world. The 286A's de-esser (yet another patented dbx circuit) gives you a frequency control so you can pick out exactly the range where the de-esser will do its thing and a threshold control so you can control the amount of de-essing that will occur. Because of the unique design of this circuit the de-esser monitors the amount of signal coming in and adjusts itself you don't have to constantly change the settings for different volume levels.

This unit is fully balanced for clean connections, has an internal power supply for reliable power without the wall wart, and knobs that click at each setting for accurately reproducible settings.

Oh...

and a GREAT Mic Pre

The 286A utilizes a precision laser trimmed ultra low noise (0.95 nV /√Hz) circuit that translates to an Equivalent Input Noise spec of 128dBu and an extremely wide bandwidth of >200kHz. All this engineering jargon translates to a gorgeous mic pre-amp circuit that you can count on to sound warm, transparent and accurate. With a precise 48 volt phantom power supply built in, you can count on the 286A to work with all your professional standard microphones.

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And if I want, let me use the processors for my other gear and I want it all in one box... You got it. The dbx 286A gives you the flexibility and control you need to make your vocal tracks sizzle and your other tracks shine. Visit your local dbx dealer today for a test drive of all the processors in the dbx 286A or call us for more information.

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PRO FILE

Laying Down the Crunch

Guitar guru James Murphy shreds the beat.

By Diane Lowery

James Murphy has clocked a lot of professional career time as a guitarist for the heavy metal group Testament. He is known for speed-demon guitar riffs, and that style of playing is what dominates Murphy's new solo album, *Convergence*. But to record the album tracks the way he wanted to hear them, he had to split time between his ADAT-based home studio and a pro facility.

The reason for this cross-studio traffic was that Murphy wanted to take advantage of the benefits of a large studio—its great mics and great tracking rooms—as well as make the session musicians he hired for the project feel comfortable. (It's a little tough expecting a studio pro to record tracks in your bedroom!) In the process, he ended up with fourteen drum and bass tracks recorded on a Studer 24-track, 2-inch analog machine. He then took the pro rhythm tracks home to overdub at his leisure. But to make this happen, he had to be extremely careful when transferring the analog tracks to ADAT to avoid sonic compromises.

At the beginning of the project, Murphy booked two days at Prairie Sun studios in Sebastopol, California, to record with top-flight bassist Brad Russell and session drummer Deen Castronovo. Prairie Sun's assortment of Neumann U 67s and U 47s, AKG C 414s and C 112s, and Sennheiser MD 421s allowed him to do some heavy-duty drum miking. Ultimately, up to twelve mics were used to record the drums, which allowed Murphy the luxury of tracking many elements, such as the toms and hi-hat, individually to produce more sonic impact.

"I thought it would be an interesting experiment to record to analog and then transfer the tracks to ADAT," says Murphy. "I also wanted to see if the analog sound quality could be maintained when the tracks were transferred to digital. And anyway, Deen is more comfortable with the sound of his drums on analog."

For the most part, the transfers came together smoothly. However, there *was* one problem.

"It's crucial to pay attention to all of your signal levels during the transfer," explains Murphy. "I was so busy making sure the kick drum, snare, and toms were going to tape at a high enough level—and that the signals were not distorted—that I simply forgot to check the hi-hat track. On the 2-inch, the hat was recorded with a nice, robust level. But after the transfers, it appeared on the ADAT at a low level and it sounded buzzy. Of course, I didn't notice this until I got home."

Murphy initially thought he was going to have to do the transfer session all over again, but he discovered—to his relief—that the hi-hat signal was loud enough in the overhead tracks. Although everything turned out fine, Murphy walked away from the session much wiser.

"I didn't check the transfers and listen back to them before I left Prairie Sun, because I was in a hurry," he says. "But I was fortunate that it worked out okay. My advice is 'don't be in a hurry.' Listen back to the transfers, and make sure everything is well recorded before you strike the setup and leave the studio. It's time consuming, expensive, and a real pain if you have to hire the studio to do the transfer again."

For more information contact Shrapnel Records, PO Box P, Novato, CA 94948; tel. (415) 898-5047; e-mail sndtemple@aol.com.

James Murphy

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Dynamics processing is a critical part of any recording. You need the proper tools to help capture the performance and bring it to life. A tube-equipped compressor/limiter and de-esser, the VCS-1 will help add punch to instrument tracks or help your vocal tracks jump out in a crowded mix. Using the adjustable Tube Gain control, you can add tube warmth to any source signal, especially one that is digitally recorded. Let the VCS-1 from DigiTech be your tool of choice for dynamics processing.

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If you think the thundering sound of John Bonham's drums on Led Zeppelin's legendary fourth album is the result of slick signal processing, think again. Three elements are responsible for that sound: the way Bonzo tuned his drums (he used, for example, a double-headed 26-inch kick drum with no muffling); the way he played (slammin'!); and the acoustic properties of the room he was recorded in. In this case, the drum room was a stone hallway at Headley Grange, the country house in Hampshire where most of the album's tracks were cut. Had Jimmy Page and engineer Andy Johns not thought to put the drums in that hallway, the awesome intros to "Rock and Roll" and "When the Levee Breaks" may not have exploded their way into rock and roll history.

That kind of experimentation is what makes recording as much of an art as a science. The only problem is that we tend to be creatures of habit—and habit is the nemesis of art. For example, it is easy, once you've found a reverb that works nicely on vocals, to get in the rut of relying on it for all your vocal tracks. Ditto for miking—after discovering the "best" way to record a cello, we tend to do it the same way forevermore. That's a shame, though, because a different mic or mic position might yield a tonal quality more suitable to a particular piece of music.

But that is not the only reason to break old recording habits. Even if you're no Luddite, there's a healthy pleasure to be had in sidestepping digital effects processors and other technological contrivances in favor of seeking out and documenting new, organic soundscapes. After all, where's the sense of adventure in turning knobs and pushing buttons? Frankly, once I've blown through the bells and whistles on a black box, the thrill is gone.

By Brian Knave

The New Mini-Mixer Revolution

At Spirit we are committed to providing users with unfussy, straightforward information. After all, making a product choice is difficult enough without being confused by an excess of unhelpful jargon. Some manufacturers hype up their designs with fancy names and unrealistic capabilities. In many cases, they claim supposedly "new" features and circuit designs that have been common to Spirit and Soundcraft consoles for as long as 25 years.

Below, you can find out just a few of Folio F1's many attributes. If you like what you read, contact us for more information or visit your Spirit dealer for a demonstration.

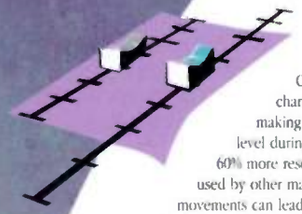
At *Spirit, we have a simple mission; to design and build the highest quality professional audio products at the lowest possible prices. In January 1993, this mission led to the introduction of Spirit Folio. Offering digital sound quality, portability and a host of features at a fraction of the price of existing consoles, Folio revolutionized small mixer design. More than 200,000 users around the world have chosen to buy Folio.*

Following 3 years of listening to Folio users, relentless research, and the development of 21st century surface-mount manufacturing techniques, Folio F1 has now improved upon this heritage.

Visit your local dealer and take F1 for a spin today.

MORE CONTROL OVER YOUR MIX

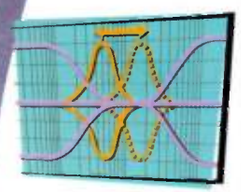
100mm faders



Of all the small mixers on the market *only Folio F1* offers the extra control and creativity of 100mm faders, the type used on professional consoles. Clearly, the longer the fader, the smaller the change in level between each movement, making it is easier to make fine changes of level during complex mixes. F1's faders give over 60% more resolution than standard 60mm designs used by other manufacturers*, where even the smallest movements can lead to large leaps in level, especially near the bottom of the fader's travel.

* We used to fit them just like everyone else, too. But as a result of feedback from users, Spirit has stopped using 60mm faders on all of its consoles in favor of 100mm controls.

Musical British EQ



While some manufacturers claim to have the "British Sound" Folio F1's EQ really is formulated exclusively in the UK by Graham Blyth. Graham, designer of Soundcraft mixers for over 25 years (and a man with over 2 million channels of EQ to his name), designs musical EQ's on (in his words) a basis of 10% inspiration and 90% perspiration. He believes that many imitators lack the experience or the time to obtain the right, natural-sounding combination that he creates. Here are just a couple of factors essential to Graham's designs:

Careful positioning of the fixed shelving filters

Mid range frequencies will be affected if a mixer's High Frequency shelving control is set too low and with too gentle a slope. The same happens if the Low Frequency control is set too high. Careful positioning of these controls gives F1's EQ its warmth and naturalness. Inaccurate positioning by "British EQ" imitators means that any extra top-end sizzle or added low-end punch is masked by unwanted mid frequency mushiness.

The sweep frequency Q factor

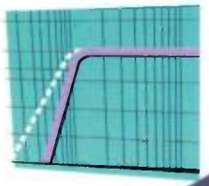
The Q of a sweep EQ refers to the bandwidth of the "bell curve" around the frequency being treated. Setting the Q too wide affects a whole series of unwanted additional higher and lower frequencies.

A Q that is too narrow, however, alters only a small number of frequencies so that any boost will sound too strident.

Severe High Pass Filters

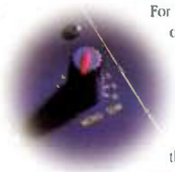
High Pass Filters are essential for Live performances, reducing the low frequency stage rumble that can cause "muffled" mixes. However, most small mixers either do not have them or use ineffective designs.

With a heritage of 25 years of Soundcraft design experience in Front of House consoles for large tours, Spirit fully appreciates that over-subtle High Pass Filters don't give the corrective control needed to reduce stage rumble. Often standard 12dB filter designs may not become effective until well below the frequencies where mushiness and rumble occur.



As the diagram shows, the rate of attenuation of Folio F1's 18dB per octave slope is faster than standard 12dB per octave filters (shown in white), guaranteeing extra control over unwanted low frequencies, and extra clarity for your live mixes.

Custom Designed Controls



For cost effectiveness, many manufacturers use off-the-shelf rotary controls. These are not designed for high performance audio, so that a small move often causes a huge leap in level at one point and hardly any change at another. In contrast, all Spirit controls are custom-specified by Graham Blyth especially for their task. This ensures that they provide a consistent, accurate response, even during the most complex mixes.

PROFESSIONAL SOUND QUALITY

UltraMic Preamps

In common with all Folio desks, F1 uses Spirit's patented UltraMic™ Preamps. These represent a revolution in low cost mixer electronics, with the ability to handle +22dBu of input level, high CMRR and an EIN figure that's as close to the theoretical noise floor as its possible to get. In other words, they're virtually silent! In addition, the mic and line inputs offer up to 60dB of gain range, meaning you can plug in anything from the feeblest of keyboard inputs to the hottest of mics without any worries.

Balanced Connections

All Folio F1's main inputs are balanced, reducing potential hum and keeping the noise-floor to a minimum. In addition, all F1's outputs are impedence-balanced so that long cable runs are possible without the worry of interference from electrical equipment.

GROUND BREAKING BUILD QUALITY & RELIABILITY

Surface Mount Manufacturing

Spirit's huge investment in 21st century surface-mount manufacturing technology provides F1 with considerable advantages over conventionally designed mixers.

Because F1's surface-mount components are soldered direct to its PCB, they are less prone to vibration fatigue, improving the product's reliability.

The accuracy of surface-mount insertion also means that Spirit can fit more components and hence more features into a smaller space. F1's array of professional features is made possible by this.

EXTRA CREATIVITY

Up to 18 inputs as standard

With 6 or 8 mono channels, 4 stereo inputs and a tape return, Folio F1 gives you plenty of input capacity in a very small footprint.

Inserts on Every Mono Input

Unusual for a console of its size, F1 is equipped with inserts on every mono input. This gives you the flexibility to connect signal processors such as compressors and limiters to individual mic or line sound sources during a performance or recordings.

Flexible Auxiliaries

The requirements of an auxiliary are different for effects sends and musicians' monitor mixes. For effects, the aux should be post-fader so that the amount of the effect is linked to the channel level. However, for monitoring, the musician needs a pre-fade aux to be independent of the main outputs. Many small mixers are not flexible enough to deal with both of these requirements. In contrast, one of F1's three aux's is pre/post switchable, ensuring that up to 2 Aux's are available for either effects-heavy or 'monitor'-heavy applications.

Mono Output

Most mixers in F1's price bracket only offer simple stereo outputs, which can often limit your scope in performances. F1's additional Mono Output is a creative tool that can be used to feed separate speaker clusters, or a bass bin to reinforce bass frequencies.

In worship or conferencing applications, it can be used to create a simple mono mix with ease, or to feed induction loops in installations.

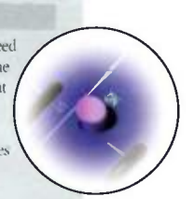
Additional Stereo Inputs

Folio F1's four stereo inputs provide enough scope for you to connect pre-show tapes, keyboards and CD players, as well as effects units. And if even 4 stereo inputs are not enough, then F1's tape return can be routed to mix, allowing a fifth stereo sound source to be utilized.

THE ADDED EXTRAS

Channnel On

Many small mixers do not offer mute switches, so when you need to take a sound source out of the mix, you are forced to pull the fader down to zero. Even then, poor design can often mean that these signals are not fully attenuated and spill over into your mix. F1's Channel On switch avoids the need for any fader movements when sounds are no longer required, and guarantees complete silence from unused signals.



Rackmount Option

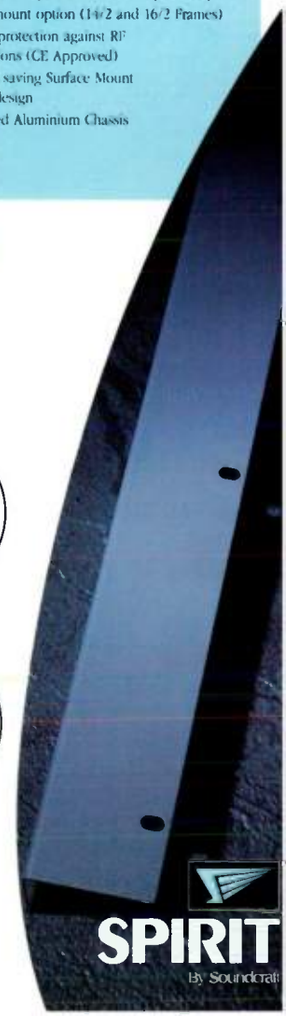
F1's optional rackmount (shown at right) is carefully designed for ease of fitting with the minimum of tools. Available for both mixer sizes.

Phantom Power

Allows you to use condenser microphones without the need for batteries.

PROFESSIONAL FEATURES

- 2 versions - **14/2 and 16/2** frame sizes
- **8 Outputs** including Aux's
- **Digital Sound Quality**
- **6 or 8 Mono Channels** with high quality **UltraMic™** mic preamps
- **4 Stereo Inputs**
- **100mm Faders** for extra fine control over mixdown
- **Inserts** on every mono channel and mix outputs for signal processors
- Authentic **British 3-band EQ** with creative **swept mid** control
- Steep 100Hz **18dB per Octave High Pass Filters** to eliminate low frequency rumble
- **3 Aux's**, with up to 2 pre- or post-fader for foldback or effects, plus Aux Master controls
- **48V Phantom Power**
- **Channel On** switches
- Separate **Mono Output** with level control for drum fills or mono PAs
- Professional **Impedance-Balanced XLR type Outputs**
- Built-in Carry Handle for extra portability
- Rackmount option (14/2 and 16/2 Frames)
- High protection against RF emissions (CE Approved)
- Space saving Surface Mount PCB design
- Rugged Aluminium Chassis



FOLIO F1 FADER 100 High Performance Audio

SPIRIT
By Soundcraft

FOLIO

F1

FADER 100

MONO INPUTS

Connectors

A Neutrik XLR allows you to connect mics. Phantom Power is available globally from the master section if condenser mics are being used. The 1/4 inch jack is for connecting line level instruments such as guitars or keyboards. Both mic and line inputs are balanced to provide extra protection from noise. An insert point allows you to plug in signal processors such as compressors or noise gates using a Y-cable.

Gain Control

Spirit's UltraMic™ padless preamp provides an exceptionally large 60dB of gain range, allowing anything from line level instruments to the most sensitive of condenser mics to be level matched, using a single control. A maximum of 22dBu of headroom is available to avoid clipping of strong signals.

Equaliser Section

Spirit's steep 18dB per Octave 100Hz High Pass Filter cuts out unwanted low frequency "rumble" from your mixes. Its use is especially recommended for any vocal performance. A genuine 3-band British EQ provides extra clarity and control over your sound with the fixed frequencies chosen carefully to add extra "punch" or "sizzle". In addition, Spirit's swept mid EQ allows you to choose the frequencies that need to be altered, giving you extra power and flexibility compared to small mixers with only fixed frequency mid EQ's.

HF EQ at 12kHz with 15dB of cut or boost.
LF EQ at 60Hz with 15dB of cut or boost.
Mid EQ sweeps between 250-6kHz with a Q of 1.5. 15dB of cut or boost available.

Auxiliary Section

There are 3 flexible auxiliary controls, allowing F1 to be equally useful in live or recording situations. Aux 1 is pre-fader/post EQ for foldback or stage monitor mixes, with Aux 3 set post-fader/post EQ as an effects send. Aux 2 is normally post-fader/post EQ but is switchable pre-fader/ post EQ from the master section for a maximum of two effects sends or monitor mixes at once.

Fader section - 100mm

A 100mm Alps fader with 10dB of gain above its zero mark provides more resolution and control over your mix than small mixers with standard 60mm sliders. The channel is muted until the channel ON switch is pressed. The PFL switch allows you to solo the channel pre-fader to check your level settings.

STEREO INPUTS

Folio F1 has 4 stereo inputs housed in two channel strips. Inputs 9-10 and 13-14 are intended to control sound sources such as CD players, DAT players and Hi-Fi equipment. Alternatively they can be used as simple instrument inputs or effects returns. Inputs 11-12 and 15-16 are more sophisticated and intended mainly for keyboards, and effects returns or playback machines where more control is required.

Inputs 9-10 and 13-14*

Two unbalanced phono connectors allow easy connection to to hi-fi equipment or DAT players without the need for adaptors. The LEVEL control governs how much of the input signal is routed to mix.

Inputs 11-12 and 15-16*

Connectors

Two balanced line jacks allow you to connect the left and right signal from a balanced or unbalanced stereo source. You can use these inputs as mono channels by connecting a signal to the left jack only. A Gain control allows you to match any line level signal.

EQ Section

A fixed high and low frequency control allow you to shape the tone of your stereo instruments.
High Frequency at 12kHz with 15dB of cut or boost.
Low frequency at 60Hz with 15dB of cut or boost.

Balance

This sets the amount of signal feeding the left and right mix outputs.

All other functions on the stereo inputs are identical to the mono channels.

* Channel numbering for 16/2 version only

MASTER SECTION

Phantom Power

When pressed, this supplies +48V globally to power condenser mics without the need for external power supplies or batteries. An LED indicates that +48V is active.

Mono Output

An impedance balanced 1/4 inch jack provides an extra output taken as a mono sum from the mix. It is invaluable for simple mono PAs in small venues. The MONO SUM rotary control allows you to alter output levels.

Meters

Two 3-color, 10 segment bargraph meters show mix output levels. The meters are peak reading (PPM) allowing you to see possible sources of distortion from instruments with high transient levels. These transients would not register on standard VU meters. When any PFL or AFL solo is pressed, both meters switch to show the solo level.

Phones

The phones connector allows you to plug in headphones of 200 Ohms or greater. The MONITOR + PHONES rotary allows you to control headphone volume or monitor output.

2 Track Return

Two unbalanced phono connectors allow you to connect a DAT machine or cassette deck to play pre-show music at live events, or playback your final masters when recording. The 2TK LEVEL rotary controls playback levels. The two track return can also be routed to Mix, by pressing the 2TK TO MIX switch. This provides an extra stereo input for instruments or allows you to jam along with previously recorded music.

Monitor Outputs

Two impedance balanced 1/4 inch jacks marked MONITOR L&R allow you to connect F1 to an amplifier and monitor speakers. Monitor out levels are controlled by the MONITOR AND PHONES rotary.

Auxiliary Masters

Three rotary controls govern the output levels from the impedance-balanced auxiliary outputs. Each master can be solo'd using the AFL switch, so you can monitor foldback or effects send levels. Pressing the Pre/Post switch allows you to change all Aux 2 signals between post- and pre-fader.

Mix Outputs

The MIX L & R outputs are taken from two impedance balanced XLR connectors, with two insert points available to connect signal processors such as graphic EQs across the mix. The mix level is governed by two 100mm faders with 0dB at the top of their travel for extra resolution.

GENERAL

Integral Carry Handle

Folio F1 has an integral carrying handle which doubles as a tilt for the control surface, giving a comfortable working angle.

Latched Power Connector

This ensures that the AC power lead remains secure.

Rackmount Option

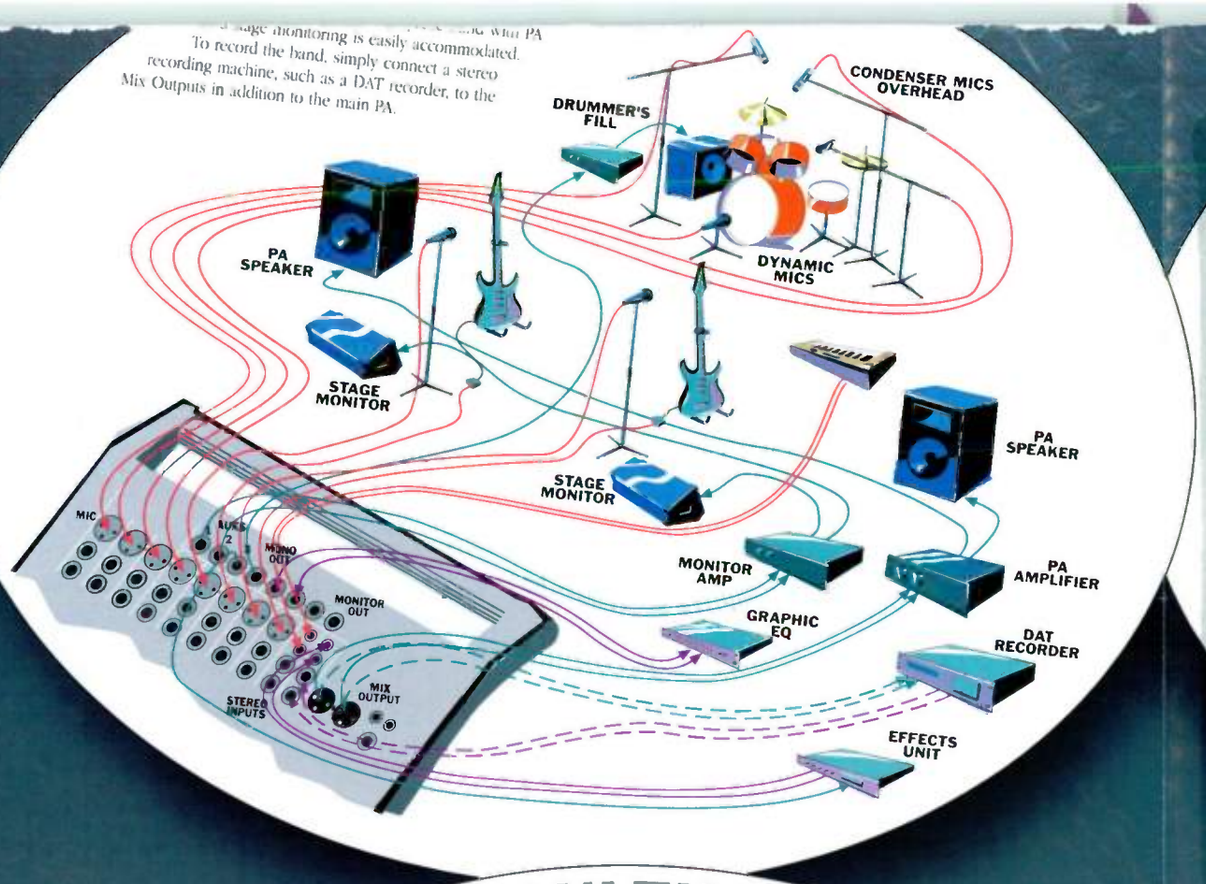
Additional rackmount ears are available to convert Folio F1 into a 19 inch rackmount unit. The unit occupies 10U of rack space and an additional 1U is advised for routing cables and leads out of sight.

LIVE MIXING

In this example, a four piece band

MULTITRACK

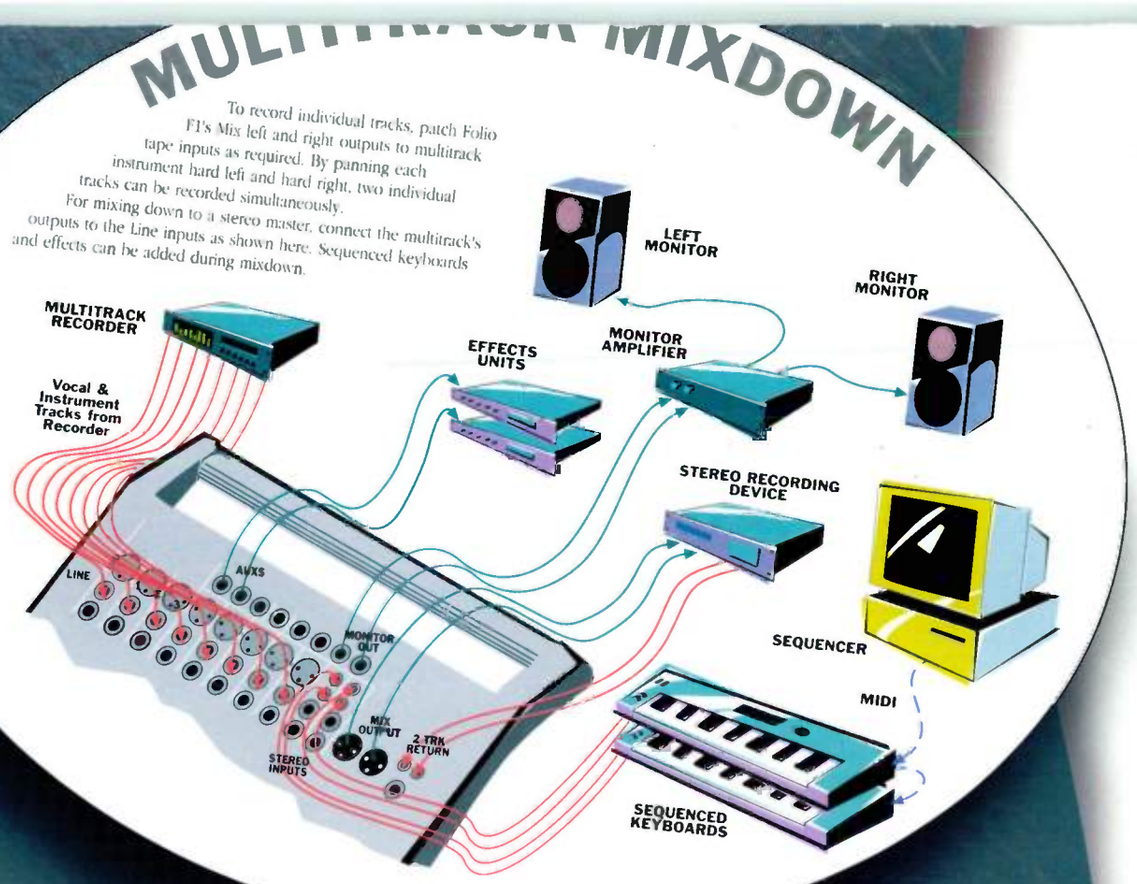
...stage monitoring is easily accommodated. To record the band, simply connect a stereo recording machine, such as a DAT recorder, to the Mix Outputs in addition to the main PA.



MULTITRACK MIXDOWN

To record individual tracks, patch Folio F1's Mix left and right outputs to multitrack tape inputs as required. By panning each instrument hard left and hard right, two individual tracks can be recorded simultaneously.

For mixing down to a stereo master, connect the multitrack's outputs to the Line inputs as shown here. Sequenced keyboards and effects can be added during mixdown.

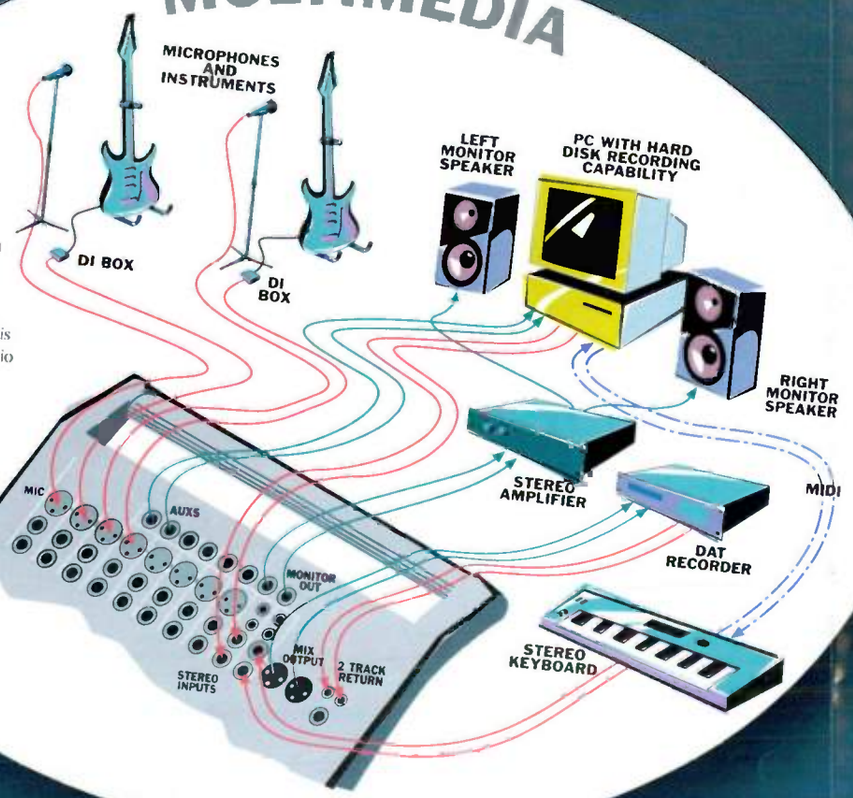


MULTIMEDIA

For simple multimedia applications, vocals and non-MIDI instruments are recorded on the computer via Aux 1 and Aux 2, with keyboards controlled via MIDI from sequencer software.

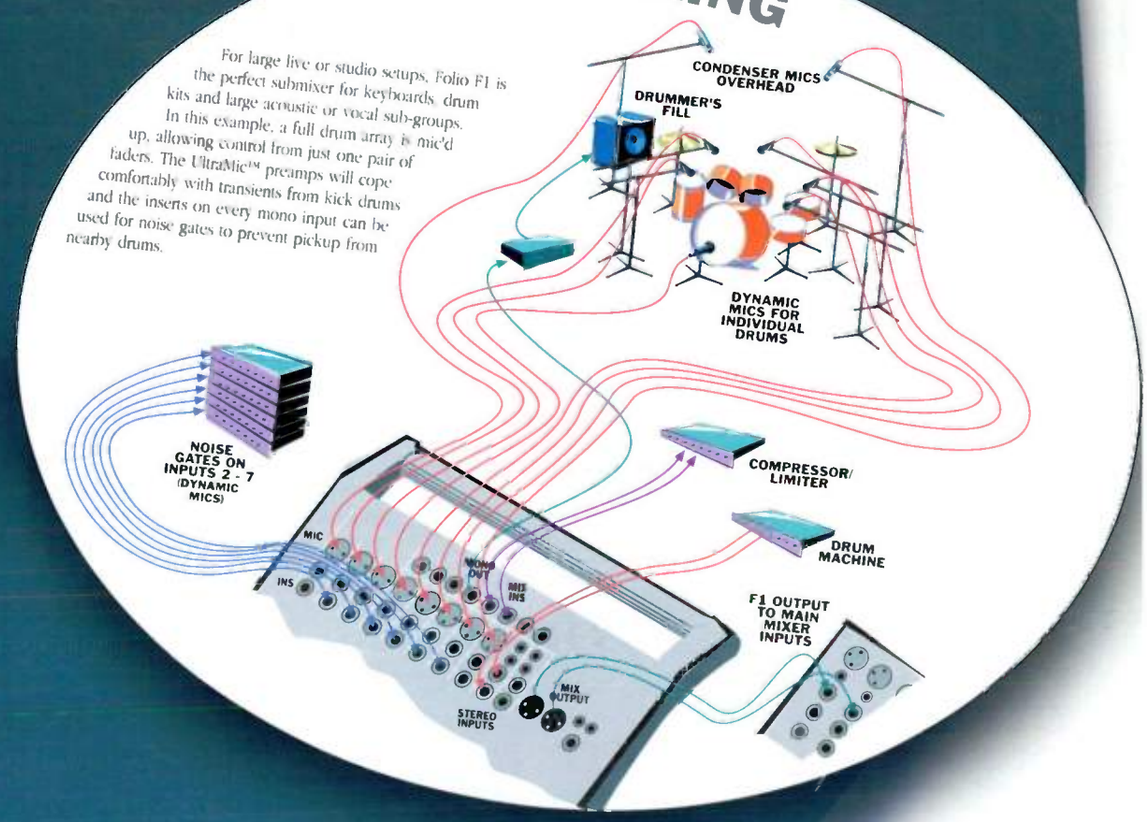
The output from the computer is connected to a stereo input on Folio F1, allowing you to listen to your recordings.

Recordings can be mastered to DAT from the Mix Outputs.



SUBMIXING

For large live or studio setups, Folio F1 is the perfect submixer for keyboards, drum kits and large acoustic or vocal sub-groups. In this example, a full drum array is mic'd up, allowing control from just one pair of faders. The UltraMic™ preamps will cope comfortably with transients from kick drums and the inserts on every mono input can be used for noise gates to prevent pickup from nearby drums.



SPECIFICATIONS

NOISE

Mic EIN @ max. gain, 20Hz - 20kHz bandwidth.	
150Ω source impedance	-129dBu
Aux, Mix & masters at max., 10 inputs routed with faders/pots down	<85dBu

CROSSTALK (@1kHz)

Channel Mute	<96dB
Fader Cutoff (rel 0 mark)	<96dB
Aux Sends Pots Offness	<89dB

FREQUENCY RESPONSE

Mic/Line Input to any Output, 20Hz to 30kHz	<1dB

THD+N

Mic sens. -30dBu, +20dBu at all outputs @ 1kHz <0.006%

INPUT AND OUTPUT IMPEDANCES

Mic Input	1.8kΩ
Line Input	10kΩ
Stereo Input (Unbalanced RCA Phono)	12kΩ
Stereo Input (Balanced Jack)	10kΩ
Mix, Aux & Inserts	75Ω

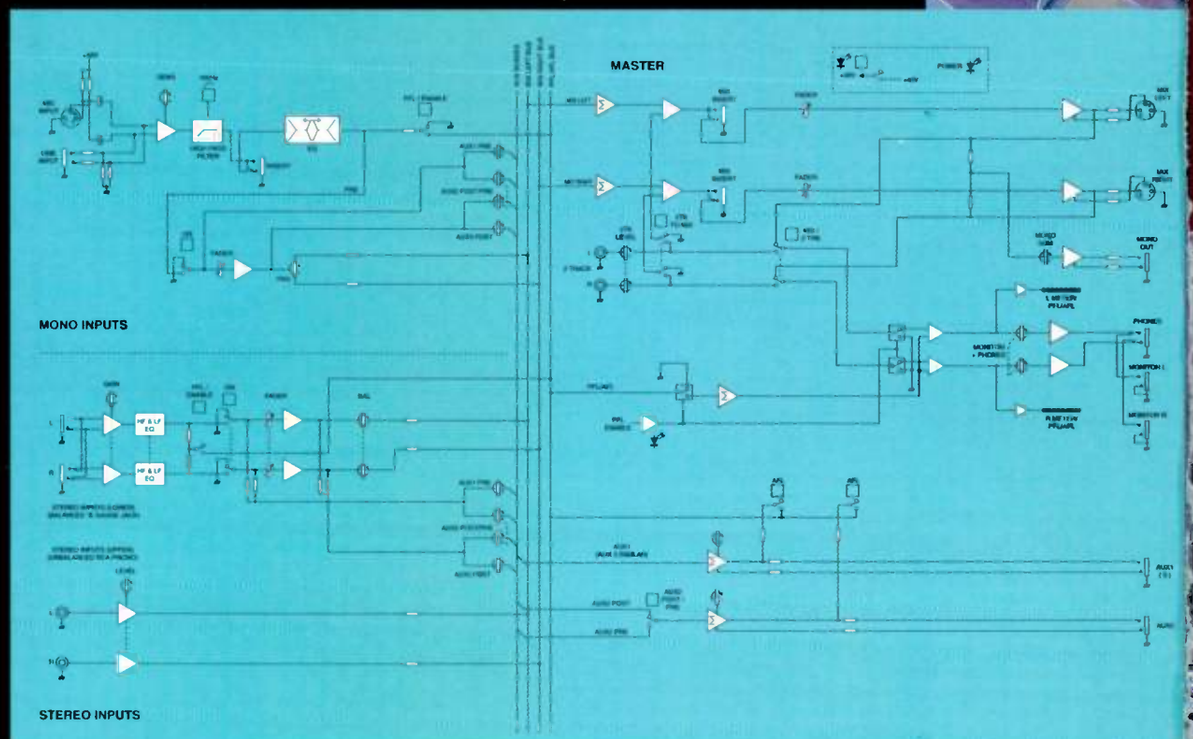
INPUT AND OUTPUT LEVELS

Mic Input max. level	+22dBu
Line Input max. level	>30dBu
Stereo Input max. level	>30dBu
Headphones (@ 200Ω)	150mW

DIMENSIONS INC HANDLE

16/2 model	70H x 399W x 512D mm (2.8H x 15.9W x 20.4D ")
14/2 model	70H x 349W x 512D mm (2.8H x 13.7W x 20.4D ")
Rackmounted	440H (10U') x 480W mm (17.5H x 19W ")

BLOCK DIAGRAM



SPIRIT
By Soundcraft
A Harman International Company

Spirit by Soundcraft™,
Harman International Industries Ltd.,
Cranborne House,
Cranborne Industrial Estate,
Cranborne Road, Potters Bar,
Herts EN6 3JN, England.
Tel: +44 (0)1707 665000.
Fax: +44 (0)1707 665461.

Spirit by Soundcraft™ Inc.,
11820 Kemper Road.
Auburn, CA. 95603
Tel: (916) 888 0488.
Fax: (916) 888 0480.

Part # Z10384

CE This equipment complies with the EMC Directive 89/336/EEC

http://www.spirit-by-soundcraft.co.uk

PROFESSIONAL PORTABLE MIXING CONSOLE WITH 100mm FADERS

FOLIO F1 FADER 100

SPIRIT

APPLICATIONS

Live Sound - Bands, Small Venues and Installations, Conferences, Schools, Places of Worship

Recording - Project Studios, Home Studios, MIDI Programming Suites, Video Post Production Suites

Submixing

Location Sound

FOLIO F1 FADER 100

SPIRIT

High Performance Audio

100mm FADERS

Of all the small mixers on the market, ONLY Folio F1 offers the extra control and creativity of 100mm faders used on large professional consoles.

As you can see, the longer the fader, the smaller the change in level between each movement - making it easier to make fine level changes during complex mixes. With standard 60mm designs even the smallest movements can lead to large leaps in level, especially near the bottom of the fader's travel.

• UltraMic™ preamps

Copied but never equaled, our padless UltraMicPre Amp design with up to 22dB of headroom and noise-free transparent professional performance is the best MicPre you can buy at even 10 times the price. There's alot of marketing hype about MicPre's out there but simply put, Spirit's UltraMic Pre is as good as anything we've made for 25 years - comparable to Mic Pres in $50,000 consoles. We've had customers call and say "I've never heard a mixer sound so open and clean". Give us a listen... You'll be amazed!

• British EQ

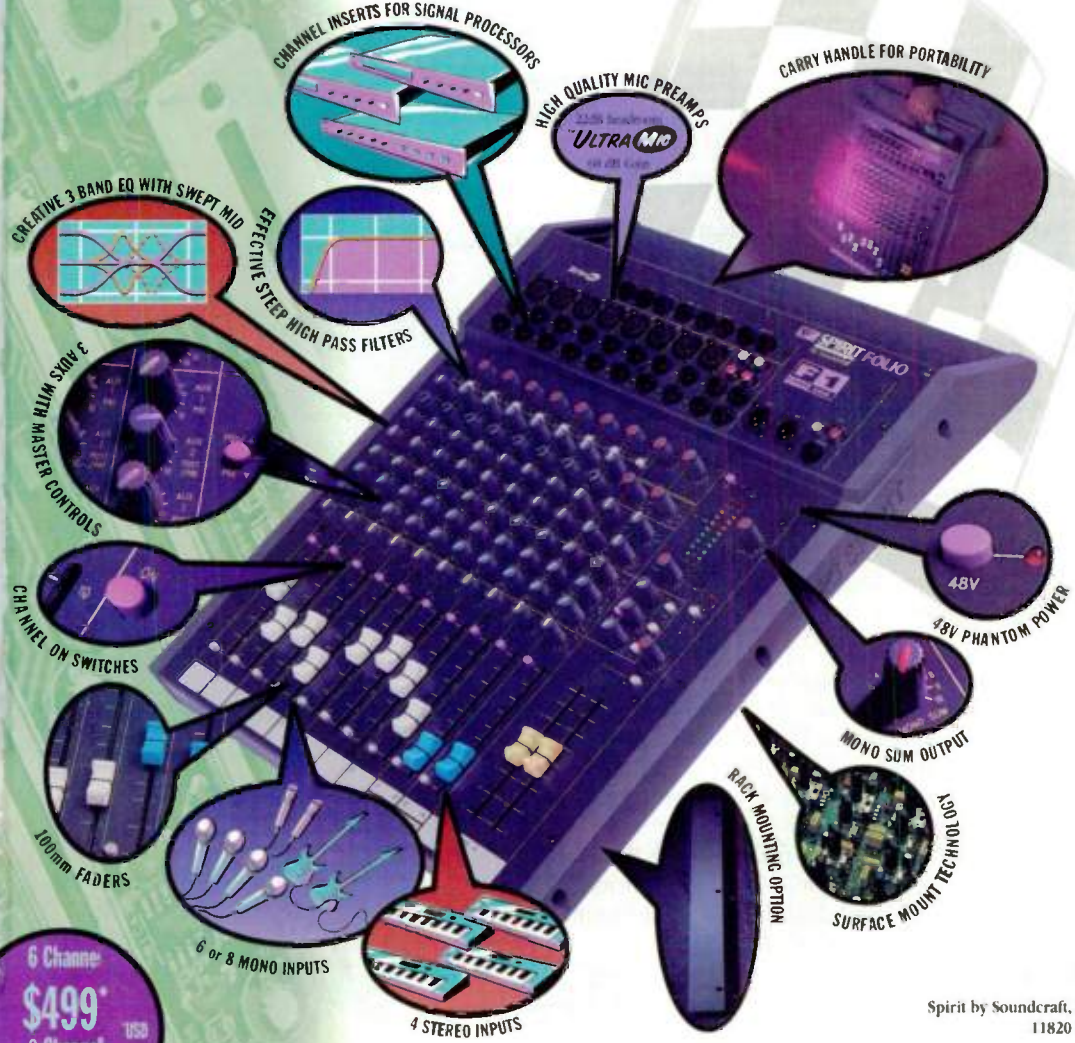
Ask any major engineer why the top recording consoles in the business are British designs and you will soon discover that the SOUND of the EQ is at the top of their list. It's a knack they have for designing warm musical turnover filter points by paying attention to the selection of good sounding quality components - like polyester capacitors in the critical audio path. Spirit is no stranger to these specifications and we refuse to compromise (as others have) simply to reduce our cost.

• Total control

We believe that all of the elements in the signal path of a mixer should allow the engineer the widest range of control—afterall, YOU are the engineer and subtle changes can make or break the quality of a final mix. Unlike other small mixer manufactures that use off-the -shelf components, at Spirit we've paid meticulous attention to the "Law" of the rotary controls so you can use the entire 360 degrees while mixing without the worry of small movements creating large jumps in level or clipping the summing amps. We've also added 100mm faders (not 60mm) allowing 60% more fader resolution - putting YOU back in control.

• Surface Mount Technology

As one of the largest mixing console manufactures, Soundcraft has invested in the most advanced production systems available. Our new leading edge circuit construction using Surface Mount double-sided glass epoxy PCBs means lower manufacturing cost without any downside. The result: much more mixer for your dollar.



6 Channel $499* USB
8 Channel $599*

APPLICATIONS

Live Sound - Bands, Small Venues and Installations, Conferences, Schools, Places of Worship

Recording - Project Studios
Home Studios
MIDI programming Suites
Video Post Production Suites

Submixing

Location Sound

PROFESSIONAL FEATURES

- 2 versions - 16 AND 18 TOTAL INPUT frame sizes (including Tape Return to Mix)
- 6 OR 8 MONO MIC/LINE CHANNELS with high quality ULTRAMIC™ mic preamps
- 8 OUTPUTS Including Aux's
- DIGITAL SOUND QUALITY
- 4 STEREO INPUTS
- 100MM FADERS for extra fine control over mixdown
- INSERTS on every mono channel and mix outputs for signal processors
- Authentic BRITISH 3-BAND EQ with creative SWEPT MID control
- Steep 100Hz 18DB PER OCTAVE HIGH PASS FILTERS to eliminate low frequency rumble
- 3 AUX'S, with up to 2 pre- or post-fader for foldback or effects, plus Aux Master controls
- 48V PHANTOM POWER
- CHANNEL ON switches
- Separate MONO OUTPUT with level control
- Professional IMPEDANCE-BALANCED XLR OUTPUTS
- Rackmount option
- CE Approved for protection against RF emissions
- Ultra reliable spaceage SURFACE MOUNT PCB design

SPIRIT
By Soundcraft

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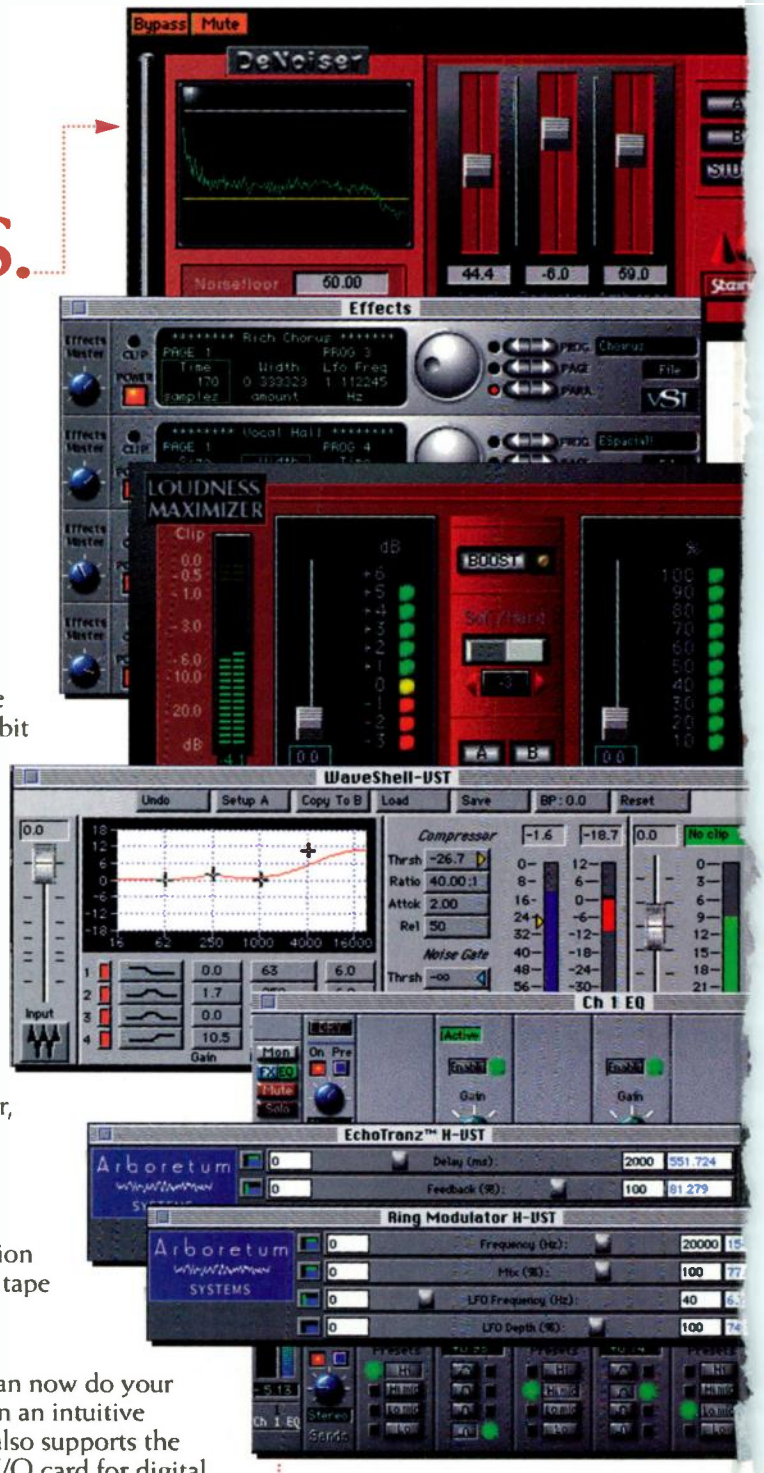
Spirit by Soundcraft, Inc., Harman International Industries Ltd.,
11820 Kemper Road • Auburn , California 95603
Tel (916) 888-0188, Fax (916) 888-0180

http://www.spirit-by-soundcraft.co.uk

IT'S MULTIPLE CHOICE AND THERE ARE NO WRONG ANSWERS.



When it comes to raising your audio standards, the days of having few options are finally over. Steinberg's Cubase VST® now takes you to the next level with open-ended plug-in architecture and an expanding range of professional audio plug-ins that turn it into an absolute production powerhouse. Whether you want to edit your mix, master your album or just make some great music, there's a plug-in that suits your needs.

☑Steinberg DeNoiser

Removes broadband noise from any audio source material in a musically intelligent manner. (Adaptive processing with automatic noisefloor detection, 24-bit processing, realtime and offline, mono/stereo)

☑Steinberg Loudness Maximizer

Increases the subjective loudness of your recording, boosting "punchiness" with zero distortion. (Adaptive processing, 24-bit processing, punchy and brilliant sound, no "pumping and breathing," realtime)

☑Audio Track by Waves®

Combines equalization, compression/expansion and gating into one very easy-to-use program. (4-band paragraphic equalizer, compressor/expander, gate, A/B comparison, mono/stereo)

☑Hyperprism VST by Arboretum®

A package of 24 different realtime dynamic sound effects. (Pitch changing, reverb, echo, ring modulation and the unique EchoTranz™ which simulates analog tape delay by adding dynamic transposition)

The processing power of Cubase VST means you can now do your projects entirely in the digital domain, in realtime, in an intuitive manner and with outstanding results. Cubase VST also supports the Digidesign® AudioMedia III™ card and Korg® 1212 I/O card for digital input and output. And with so many different tools to choose from, it also means that upgrading the power and capability of your system is no longer the ultimate test. It's just simple addition. For more information, call your local dealer or contact Steinberg at (818) 993-4091.

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Mac OS

MacUser AWARDS Editors' Choice 1996 Finalist
Best New Music/Sound Product

CUBASE VST
Virtual Studio Technology

SEE US AT NAMM BOOTH #3609

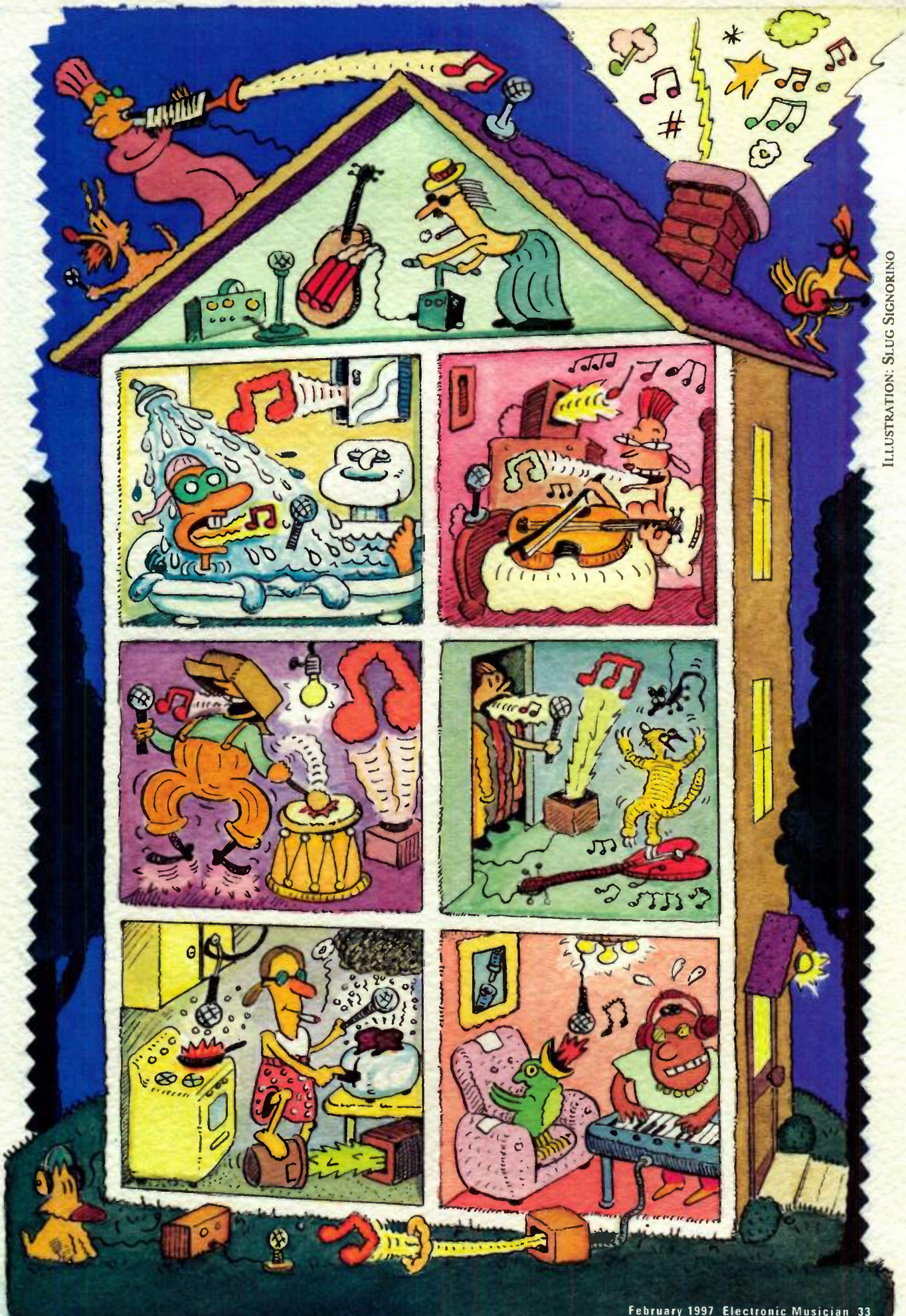


ILLUSTRATION: SLUG SIGNORINO

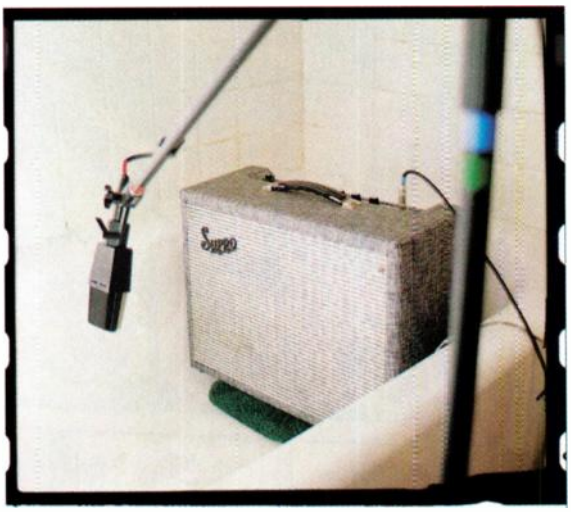
A House Divided

The fact is, even the finest effects processor on the planet is still a wannabe. That's because it's filled not with real spaces but with algorithms attempting to replicate the sound of real spaces. Of course, having a gazillion acoustic environments at your fingertips is a great convenience. But why settle for an approximation when the real thing is as close as your kitchen? You may not have a concert hall in the back of your house, but you do have several other rooms and sonically distinctive enclosures. Making use of those spaces not only will expand your sonic palette but may get your creative juices flowing, as well. With a little experimentation and luck, you'll end up with some sounds not available in that rack full of gear.

GETTING STARTED

It's simple to utilize your home's acoustic spaces in your recordings. Aside from the usual gear, all you need are cables long enough to reach from the source to the recording console. (Due to potential signal degradation, it's better to buy 50 or 100-foot cables than to connect a bunch of shorter ones.) First, play the instrument you intend to record in different parts of the room or space. The goal is to determine the position that maximizes the particular quality of sound that attracted you to the room in the first place. Then you simply position a microphone or two to capture the sound. You can monitor the results with headphones. (Of course, it always helps to have a friend acting as engineer.)

Don't think that because you live in a small apartment there are no spaces large enough to be useful. A spacious delay, after all, is not the only effect in the book. There are many usable enclosures and "subspaces" in the average home, each with a unique sonic signature. Recording within these spaces can alter the sound of a voice or other instrument in strange and sometimes wonderful ways. And of course, if you still desire a big double delay, you can always add it digitally afterward. To get you started, I've explored some acoustical nooks and crannies in my own domicile. Here are some of my findings.



BRIAN KNAVE

FIG. 2: The sound of a cranked-up Supro tube amp is given a dose of attitude and reverb by being recorded in a bathtub.

TILE STYLE

If you're like me, the last time you sang in the shower was this morning. What is it about the daily ablution that coaxes a song even from the throats of folks who couldn't carry a tune in a bucket? The hot water and negative ions perhaps help with inspiration, but it's the tile that is responsible for the great tone. Ceramic tile is highly reflective to sound waves (as are many of the other surfaces in a typical bathroom). A shower stall with three walls of tile creates a chamber in which the voice, bouncing wildly between the surfaces, engulfs the listener. Generally, because the walls are so close together, the reflections are heard not as several discrete echoes but as a field of reverberation. The result is a rich, focused tone replete with detail and nicely boosted low mids.

You can take advantage of this sound simply by setting up a microphone in the shower stall (see **Fig. 1**). Although any mic will capture the distinctive sound of singing in the shower, a large-diaphragm condenser mic set to the omnidirectional polar pattern will best document the effect. (Just make sure no one turns on the water or you could be kissing that costly condenser mic good-bye!)

Of course, vocals aren't the only thing to record in the shower stall. Any of the wind instruments can benefit from the acoustical reflections of a tiled-in space. Try flute, saxophone, trumpet, harmonica, or recorder. I've even heard of someone recording an accordion in a shower stall. Finally, if you really love what the tile is doing for your sound but you don't always have the time or inclination to set up in the bathroom, you can bring a similar sound into your studio by building some tile gobos. Simply lay tile on a 4 × 4-foot (or larger) sheet of plywood and fit it with sturdy legs. With three or four such gobos, you can quickly erect a reflective tile booth wherever it's most convenient.

BRIAN KNAVE

FIG. 1: Singer-songwriter Enna Deer takes advantage of the rich, focused tone most of us enjoy every morning while singing in the shower.

TUB TONE

Want to add some ambient attitude to an electric guitar sound? Try positioning the guitar amp in your bathtub (see **Fig. 2**). The smooth, sloping, porcelain contours of a bathtub are even more reflective than walls of tile—and if your bathtub and shower are built together, you'll get the reflective benefits of both. This trick can do wonders to bolster the already searing tone of a cranked-up tube amp.

A House Divided

Use an omnidirectional polar pattern for this application so as to capture the whole gamut of reflections. Also, position the mic far enough back to further emphasize the ambient sound. It also helps to place a towel or some other kind of padding under the amplifier to prevent annoying amp vibrations that can soil the sound with rattles, screeches, and hums.

PORCELAIN PERKS

If you want to record vocals with the bright, reflective sound that comes with porcelain but happen to live in an older-style house or apartment that has a bathtub but no shower stall, you can get similar results using a good-sized kitchen sink. Simply position the mic so that it is down inside the sink, facing up toward the ceiling (see **Fig. 3**). The resulting sound is less bassy and reverberant than what you get in the tub or shower, but it's still pretty dramatic.

Obviously, this application requires that you sing down into the sink. Audition the cardioid, omnidirectional, and figure-8 polar patterns to determine which sound is most appropriate to the song. Oh yeah—be sure to push the faucet away from the sink so you don't accidentally drip water onto the mic.

BRIAN KNAVE

FIG. 3: To create naturally bright backup vocal tracks, try recording in the kitchen sink.

COOK AND CLEAN

The kitchen sink is not the only distinctive-sounding acoustical enclosure in your dwelling. You can cook up an exceedingly bright, metallic, resonant tone in the oven (see **Fig. 4**). Practically any mic will work for this application because the glassy coloration imparted by the oven's metal interior is anything but subtle. Simply position the mic so that it's approximately in the middle of the oven with the capsule facing out. Naturally, to capture the full effect, this requires that the musician sing or play into the oven—a feat most easily accomplished with the performer sitting on a low stool or chair.

BRIAN KNAVE

FIG. 4: It's only a clambake if you play a wrong note. Otherwise, recording from inside an oven will provide a glassy, reverberant tone sharp enough to cut through the toughest sonic grime.

Because the enclosure exaggerates the high end yet imparts some reverberation, it is especially well suited to recording bassoon, oboe, and other low midrange wind instruments—that is, if you can persuade a bassoonist to perform while bending over an oven. But I also got very cool results tracking vocals, harmonica, kazoo, shaker, and claves this way.

For a warmer, surprisingly intimate sound, position the mic the same way inside a dishwasher (see **Fig. 5**). You can remove the rubber-coated dish trays to effect a brighter, more reverberant sound, or you can leave them in place for some diffusion. Also, experiment with how far back you position the source sound. For example, you could record a trumpet (with the bell aimed at the mic) from ten or fifteen feet back and still retain the coloration imparted by the dishwasher.

KITCHEN KUDOS

If you need a more spacious sound but still want plenty of ambient reflections, stay in the kitchen. The combination of linoleum floor, wood cabinets, smooth walls, and glass windows (as well as glass or metal doors on the oven, dishwasher, and so on) makes a perfect antidote to the dampened acoustics you struggled to achieve in your studio proper.

I documented an authentic Chicago-style blues-harp sound simply by putting the harp amp on my kitchen table (see **Fig. 6**) and miking it with two mics: a Shure SM57 up close, angled off-axis, and an AKG C 414 (in the omni mode) positioned as a room mic about fifteen feet back. Of course, it helped that I played a Shure 520D "Green Bullet" mic through a small, vintage tube amp (an Airline model by Montgomery Ward) with the volume cranked nearly to ten. The reflective fury of the kitchen perfectly complemented the gutsy distortion of the diminutive, overdriven amp. Talk about a shredding tone!

DOWN THE HALL

A long hallway—preferably with a wood floor—can make a great reverb chamber. In fact, this is just the trick producer Tony Visconti employed to create the multiple-reverb effect on David Bowie's song "Heroes." Visconti set up three mics for the vocal track: one directly in front of Bowie, another halfway down the hall, and a third at the very end of the hall. The second and third mics were gated with settings that required a medium-loud note to open the gate on the second mic and a full-volume note to open the gate on the farthest mic.

The recorded effect is extremely cool. As Bowie sings quietly during the verses, we hear his voice—warm and dry—solely from the perspective of the close mic. As he increases volume during the initial choruses, the gate on the second mic opens up, creating a bigger, mildly delayed vocal sound. By the end of the song, as Bowie belts "We can be heroes just for one day," the gate on the far

A House Divided

FIG. 6: The numerous reflective surfaces in a kitchen lend glorious reverb to the sputtering chaos of an overdriven tube amp. Here a Shure SM57, angled off-axis, documents the rattling of the tiny Jensen speaker in a vintage Airline amp while an AKG C 414 in omni mode (not shown) captures the room tone from fifteen feet back.

mic opens as well, resulting in a spacious, triple-miked vocal sound that is truly wondrous.

THE REAL McCOY

With the present-day glut of digital effects processors on the market, it's easy to forget that, originally, all recording studios had separate rooms called *echo chambers* that were used to generate natural reverb and echo. But just because the echo chamber is antiquated doesn't mean it is an inferior technology. Remember, the best digital reverbs and delays attempt to replicate the complex sonic attributes of real spaces. However, they never entirely succeed, a fact that becomes evident when you record a musical performance in an acoustically first-rate concert hall. (For more proof, check out the natural reverb sound on the popular CD *Chant*, a recording of Benedictine monks chanting in an ancient stone cathedral.)

But even without a concert hall or stone cathedral, you can devise a true echo chamber in your own home if you have a room that's large and reflective enough. A big bathroom—or even one that is long and narrow—can work splendidly. Other good candidates are large basements, garages, washrooms, attics, or even a deep stairwell. Simply go from room to room alternately singing and clapping your hands in each space. What you'll be listening for is at least one discrete echo and/or a rich, reverberant quality to the sound. (Reverb is the result of thousands upon thousands of tiny echoes, each with slightly different delay times, that intermingle in a complex wash of sound.)

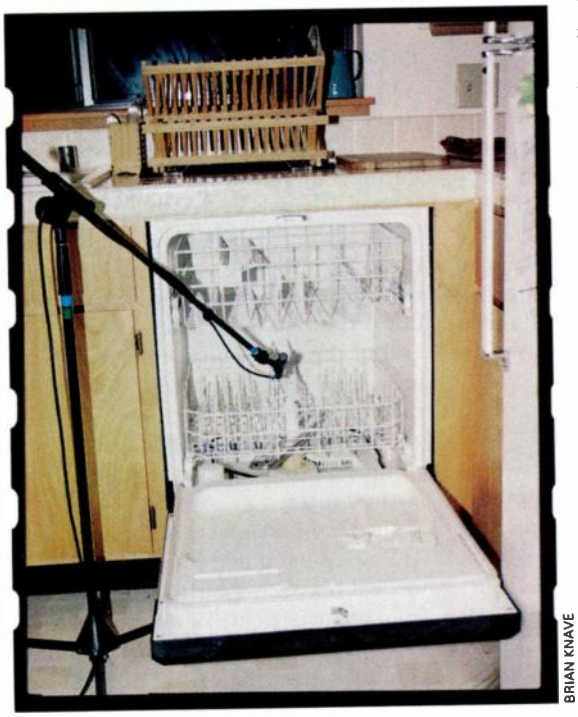


FIG. 5: The plastic walls and rubber-coated dish trays inside a dishwasher add a surprisingly warm and intimate sound to a vocal track.

If you're lucky enough to have a workable space, you'll need an extra speaker and enough cable to run it from the console to the echo chamber. Set the speaker up at one end of the room facing the farthest wall, and set two mics (or just one if a mono signal is sufficient for your needs) at the other. Aim the two mics at opposite corners of the ceiling from two or three feet below. Depending on the shape and acoustical qualities of the room, you may need to move the mics around a bit to find the positions where they best pick up the sound of the echo and/or reverb.

For optimum results, send a dry signal to the speaker by using an aux send on your console. (You may also need some type of adapter to accommodate the connectors on the board and the speaker.) If you can't get enough gain from your board to drive the speaker, you can try using a combo guitar amp as the "monitor." Just remember that anything but the lowest, cleanest setting on the amp may compromise the quality of the source signal. Return the signal (or signals, if you're miking for stereo) on a console input channel so you can mix in and EQ the chamber ambience to your liking. Creating your own reverb and/or delay in this manner can be very gratifying. And if you have a good-sounding chamber to start with, chances are you'll end up with a sound that, although not nearly as versatile as a digital "room" reverb, sounds decidedly more natural.

IN THE CLOSET

Creating effects like reverb and echo isn't the only recording application your house is good for. After all, what's sometimes needed is an *absence* of reverb and delay. Specifically, any signal that you intend to process heavily in the mix is usually best recorded dry so that ambient artifacts aren't processed along with the source signal. The only problem is that not every home recordist has a padded isolation booth at his or her disposal.

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But every home does have a closet or two, and in a pinch a closet can serve double duty as an iso booth. Just be sure to pick a closet that's well stuffed with clothes or other sound-absorbing material (blankets, towels, etc.). For maximum absorption, it should have a carpeted floor, as well.

A classic use for the closet is recording a bass amp (see **Fig. 7**). For the best results, elevate the amplifier at least six inches off the ground (a foot or two is even better) to help dampen the signal. Then put down a layer of blankets or towels on the floor directly in front of the speaker to provide more absorption. It's also helpful to surround the speaker enclosure a few feet back with baffles or a "tent" made of blankets. The goal is to reduce reflections so as to increase the presence of the bass signal.

Close-mike the bass cabinet with a dynamic mic or condenser set to the cardioid polar pattern. Position the mic three or four inches from the cabinet, facing the speaker at an angle of about 30 degrees off center. A good place to aim the mic is halfway between the center of the cone and the outer edge of the speaker. It is not necessary to play the bass amp at a loud volume; in fact, a medium volume is generally best because it's less likely to cause sympathetic vibrations in the speaker cabinet or elsewhere in the room.

SURROUND SOUND

The purpose of this article is not just to suggest alternative, organic means of creating new sounds for your recordings, it is also to cultivate your creativity. The key is enhanced listening skills. Everyone agrees that keen ears are essential in the studio, but too often we make the mistake of turning off our ears as soon as we exit the double doors.

To stretch your recording chops, it's helpful to listen no matter where you are. For example, though people may think me crazy, I often clap my hands or sing aloud in different environments, just to check out the acoustics of the space. I'm always looking and listening, trying to correlate a space's shape, dimensions, and material construction to its sound. Even if I don't discover exciting new places to record (and often I do), this awareness ultimately translates into a better understanding of the many parameters on sophisticated digital effects processors.

I hope this little guided tour of house sounds inspires you to investigate the sonic potentials available under your own roof. An acoustically neutral studio may be preferable for many applications, but it's not the only way to go. In fact, there's an impressive tradition of great records that have been made using the various rooms in normal houses, including treasures from Motown, Men at Work, and the Red Hot Chili Peppers. Besides, even the most content studio lizard in the world will eventually go berserk if he doesn't crawl out of the iso booth from time to time.


BRIAN KNAVE

FIG. 7: A closet crammed with loads of clothing provides considerable sound absorption for capturing a dry, very present bass-guitar signal. This Harmony 525 bass amp is miked with an AKG D 112 dynamic mic.

Assistant Editor **Brian Knave***'s next home will have a concert hall. Special thanks to Univibe of Berkeley for loaning the vintage Harmony 525 and Airline amplifiers.*

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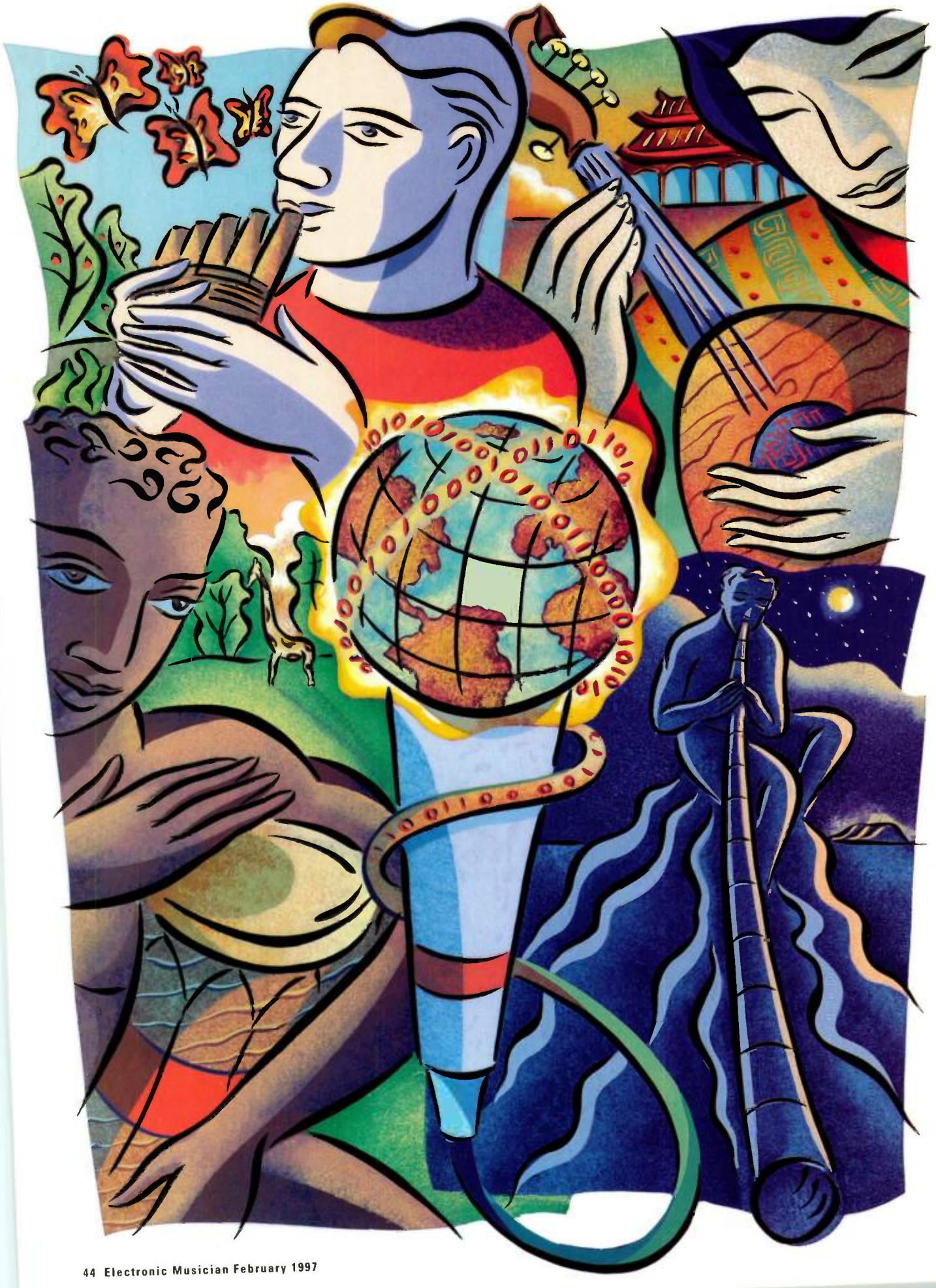


Illustration by Nicholas Vitacco

SAMPLING THE WORLD

By Jim Miller

In the 1960s, along with long hair, mind-altering drugs, and other assorted psychedelia, an entire generation of musicians and music lovers were introduced to a sound that was totally new to them, although its true origins lay in the mysterious past of a distant country. The sound of the Indian sitar, as used by the Beatles in such songs as "Love You To" and "Norwegian Wood," was quite startling at the time. It was the first introduction of an ethnic instrument into mainstream pop music.

Expand your creative frontiers with ethnic-music samples.

Since then, as our world has gotten smaller and our global village has become more cross-cultural, recordings of ethnic instruments—or, more accurately, *world instruments*—have become fairly widespread. Even sampled ethnic instruments are becoming relatively common. Certainly no self-respecting keyboard player will ever forget the famous (or infamous, depending upon your perspective) shakuhachi sample that emerged around 1985 in E-mu's Emulator II library and was used to great effect by Peter Gabriel on his hit "Sledgehammer." In fact, that particular sample, probably more than any other factor since the Beatles first picked up a sitar, caused many musicians (myself included) to wonder what other strange instruments might await discovery in remote countries, thousands of miles from the nearest Fender Stratocaster.

I have been involved with sampling almost since its beginnings in the early 1980s, and although I have recorded countless traditional instruments, I have long been interested in accurately sampling the exquisite sounds made by some of the most intriguing musical instruments on the planet.

That quest continues to this day. In this article, I'll give you an overview of some of these wonderful instruments, based upon my first-hand experience, along with a few tips on sampling them. If you have the opportunity to sample some of these instruments, I should warn you right up front that you may be somewhat disappointed at first by the results. Repeat after me: "World instruments are not synthesizers."

Why do I bring up this point? Simple—it's because many of the instruments we will be discussing are somewhat primitive in design, so a real pan pipe, for example, will not sound as consistent up and down the scale as a synthesized pan pipe. Cane is not electronic; it's organic and thus subject to imperfections. You can sample a note, say a G, that sounds great, but the A above it may be slightly brighter or somewhat breathier. This is typical of how these instruments behave, and once you get used to the differences in timbre, you will in all likelihood come to love each sound (as I have) for its individuality. With this caveat, we're ready to begin our world tour.

CLOSE TO HOME

There are a number of wonderful-sounding instruments that have been made for several hundreds of years right here in the United States, one of which is the stringed instrument known as the dulcimer. There are actually two quite different types of instruments that bear this name: the hammered dulcimer and the Appalachian dulcimer.

The modern hammered dulcimer probably evolved from an instrument of the fifteenth century; it can be seen in Italian paintings of the 1430s. However, its lineage probably goes back even farther, to the Middle Eastern santur (or santir). There is some evidence that the hammered dulcimer inspired Italian inventor Bartolomeo Cristofori to build his first piano (which is, in effect, a giant hammered dulcimer).

The hammered dulcimer is basically a trapezoidal soundbox with metal strings (usually three or four for each note to increase the amplitude) that run across one of two wooden bridges. The strings are struck with wood, cane, or (rarely) metal hammers about ten inches long. This gives the instrument a fairly wide dynamic range, depending upon the force of each strike. It is, however, still relatively quiet when compared to modern instruments like the guitar or piano, which are louder due to their size and design.

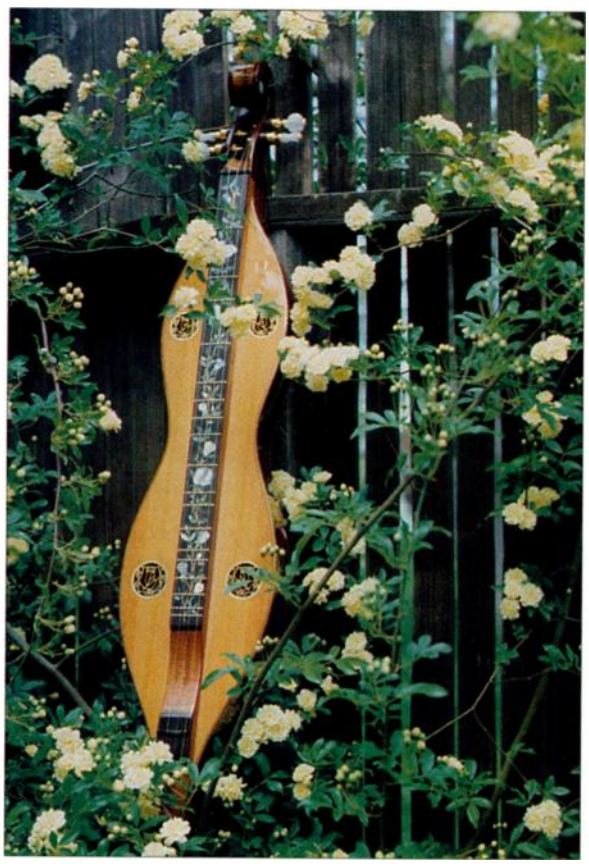
The hammered dulcimer was the first instrument I ever sampled. As with many of the other world instruments we will discuss, a great deal of care is required to adequately sample this instrument. First of all, because it is not particularly loud, you have to move your mic (or mics, if you're working in stereo) fairly close in. So be careful that the sound does not become overly boomy; the finished sample should be crisp and bright, with lots of overtones and relatively modest amounts of bass. Try placing a high-quality condenser mic, such as an AKG C 414, Audio-Technica AT4050, or Neumann TLM 193, set to a cardioid pattern, about one and a half to two feet away at a slight angle to the soundboard.

To get a good sample, it is critical to have access to a good instrument and an accomplished player who can accurately tune the dulcimer to bring out all the shimmering overtones. An improperly tuned dulcimer will sound rather lifeless. With three to four strings per note, you'll expect to hear some "beating," as you would with a well-tuned 12-string guitar.

The Appalachian dulcimer is a different critter altogether. It looks rather like a small, thin guitar with three or four steel strings and a fretted fingerboard that is played across the lap with the tuning pegs over the left knee. The left hand fingers the notes while the right hand plucks the strings with a turkey- or goose-quill plectrum. Traditionally, a melody is played on the first string while the other two or three act as drone strings.

This instrument produces a guitar-like sound, though it is much quieter because of its size. Oddly enough, unlike many of the other instruments we'll be discussing, the Appalachian dulcimer (which is, obviously, indigenous to the Appalachian mountains) seems to have no counterparts in other countries, its closest relative being either the guitar or lute.

A good Appalachian dulcimer sample will be hard to come by because there aren't too many of these instruments around, nor are there many players available. Should you be fortunate enough to have access to an Appalachian dulcimer, you will probably have to move your mic in quite close to get a decent level because this instrument is even quieter than its hammered relative. Again, a good-quality condenser mic is the best choice for capturing the soft tones produced by the instrument.



The Appalachian dulcimer has three or four strings. It is traditionally played with a turkey- or goose-quill plectrum. (Courtesy Blue Lion)

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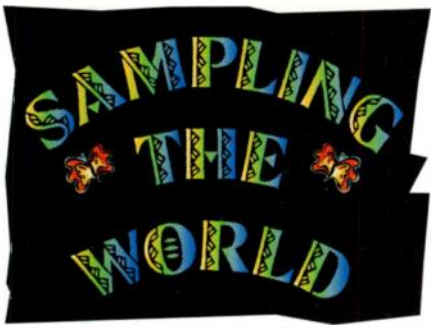

AMERICAN INDIAN SOUNDS

Staying with our "at home" theme, we should take a look at some beautifully expressive instruments that have been played for many hundreds of years by Native Americans. Although traditional American Indian music is often considered to have a relatively simple style—the flutes the players used, for instance, have very limited ranges—it is this music, more than any other, that has led to the birth of the so-called ethnomusicologist in the U.S.

American Indian tribes such as the Comanche, Navajo, Sioux, and Pawnee (to name just a few) had relatively few instruments, mostly various flutes which were played along with simple drums and rattles. The majority of Native American flutes are end-blown instruments made of various woods, such as cedar, although side-blown flutes as well as bone flutes and wooden whistles have been played for centuries. I can tell you from first-hand experience that, although these flutes may appear to be quite ordinary, many are capable of producing a very evocative sound.

Indian flutes in general don't produce much amplitude, and therefore require close miking. Care must be taken to place the mic in a position where the player's breath will not be overemphasized. This can usually be accomplished by placing the mic (or mics) slightly off to the side of the flute, out of the path of the direct breath noise. I've also had success placing a mic right above the player's head, which seems to produce a nice fundamental tone with warm overtones. As with all the instruments we'll be discussing, a little experimentation with mic placement will go a long way. Fortunately, most of the drums and rattles used in American Indian music are somewhat louder and therefore much easier to sample.

One other instrument of note is the bull-roarer, or thunderstick, which is a thin piece of wood six to twelve inches long and one to three inches wide with a hole at one end. A piece of string is tied through the hole, and the wood is swung in circles over the player's head, resulting in an outlandish roaring or screaming sound, depending on the size of the wood. This was mainly used by the Plains Indians, and those to the west of them, in religious rites.

Properly recorded in stereo, with the mics positioned four to five feet from the performer, the thunderstick can produce a very compelling sound. Condenser mics should capture all the swoosh and roar, but you may have to activate the bass rolloff on the mics to minimize rumbles and pops. A windscreen may be employed as well. (Of course, the aggressive rushes of air may *add* to the instrument's effect.) If the "ambient storm" is too intense, try using dynamic mics, as they may be less sensitive to the wind factor.

CHRISTI RAYBURN

Barry Stramp and Michael Graham Allen of Coyote Oldman make and play a wide variety of American Indian flutes and Latin American pan pipes. (Courtesy Hearts of Space)

LATIN AMERICAN FLUTES

Compared to the relatively simple flutes of North America, the instruments used by the Indians of Central and South America are quite diverse. Most of these are concentrated in the Andean region of Bolivia and Peru.

The quena (sometimes also seen spelled "kena") is a notched flute (meaning it is end-blown through a notch, like the Japanese shakuhachi) that dates back to at least 200 B.C., when it was sometimes made from a human bone. Modern quenas are usually made of cane or various soft woods and come in several sizes, including a bass quena (or quenacho) that can be up to two feet long. I love the sound of this instrument and am the proud owner of two fine handmade quenas, but I have yet to find anyone that can play them well throughout their full range. I recommend the same miking techniques for quena as for Native American flutes.

Pan pipes (or pan flutes) have a long history in both the old and new world. Variations are common throughout the Andean region. Pan pipes typically have a single row of cane pipes tied together. These are cut to different lengths to produce a fundamental pitch when the player blows at a slight angle across the open top end (the lower end is sealed).

Everyone has heard the classic Korg M1 pan flute, but the real things come in all sorts of sizes and shapes and are usually tuned in whole steps or thirds. Heavy tonguing attacks (chiffs), flutter-tonguing, and even humming in tune with each note are common performance practices that add variety to what is typically a very limited scale. The sicu (sometimes spelled siku) is a type of Bolivian pan flute that has two rows of cane pipes and thus a wider range. These instruments come in a wide range of sizes up to the huge bass sicu whose pipes may be up to two feet long in the lowest range.

Because pan pipes can be played by a skilled player at reasonable levels, miking can be done from two to three feet away and sometimes even more, depending upon how quiet the recording environment is. Because the sound

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is produced by blowing at an angle to the tops of the pipes, some care must be taken in close-miking situations to keep your mic out of the direct path of the breath as it passes across the pipe. Otherwise, you can place your mic wherever you feel the sound is best. And make sure your player produces the various attacks and playing styles typically used in performance. (For more on recording quena and pan pipes, see "Recording Musician: South American Session" in the September 1996 **EM**).

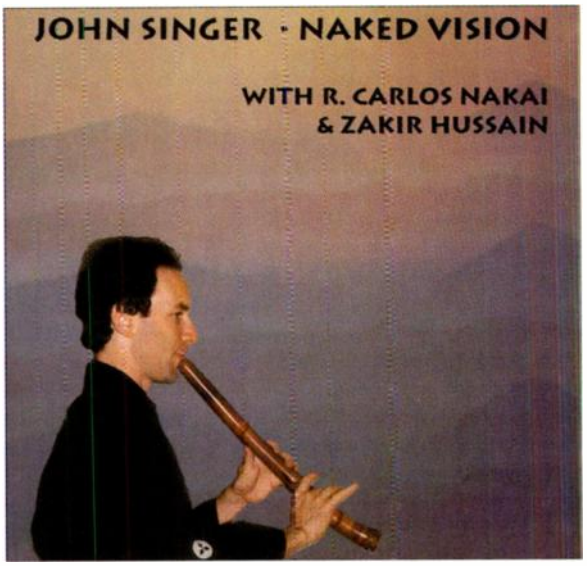
Other South American wind instruments include idioglot (Argentina and Bolivia) and heteroglot (Venezuela and Brazil) clarinets as well as clay, bark, and bamboo trumpets; simple ocarinas (usually made of clay); and even conch shells. It's doubtful that you will run across any of these instruments (except for the conch shell), though modern ceramic ocarinas can be purchased from many music stores. These instruments produce a very pure, sine wave–like sound and were originally used to imitate bird songs. The ocarina's major claim to fame is that it played the solo in The Troggs' 1960s hit, "Wild Thing."

There are also many different types of shakers and rattles found throughout Central and South America. A common example is the maraca, but there are literally dozens of gourd-type and wooden shakers as well as many large seed pods (usually from trees in the rain forest) that make exceptionally interesting sounds. You can treat most of these just as you would any percussion instrument.

THE FAR EAST

Here is where things begin to get really interesting (and a bit confusing) because of the profusion of instruments, many of questionable lineage. As mentioned earlier, most musicians are familiar with the Japanese bamboo flute, the shakuhachi, thanks to E-mu's classic EII sample. The shakuhachi is the descendant of an even older Chinese instrument, the dongxiao, and it has been around in its present form since about the sixth century. It comes in several varying sizes from 1.4 to 3.3 feet with 1.8 feet being the standard. The full history of the shakuhachi is fascinating and could easily fill the rest of this article.

What most people do not realize is that a traditional shakuhachi performance is not a series or collection of notes as we westerners think of them (as in a solo), but a complex combination of intricate "gestures" or what might be thought of as specific musical phrases. Sudden swells in dynamics and a deep tremolo are parts of this style and help add individuality to each player's performance. Most of us, however, cannot comprehend this musical vocabulary, so it's best to sample your shakuhachi player performing individual notes. Be warned, though, that a quality player will have a hard time comprehending the reason why you want them to play in this manner.



On the album *Naked Vision*, John Singer plays shakuhachi, accompanied by tabla, American Indian flute, violin, guitar, bass, and synthesizers.

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SAMPLING THE WORLD

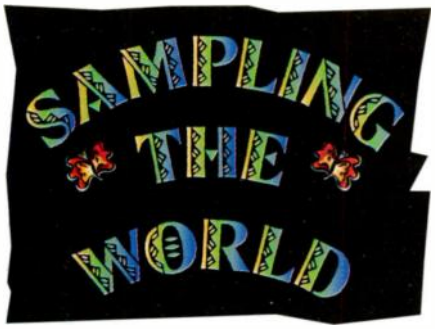
Recording a shakuhachi is relatively easy because it produces ample volume when played by a skillful performer, though some of the subtle tones can be on the quiet side. Try a condenser microphone (or a stereo pair) slightly above and off to the side of the instrument about three feet away, and move in the microphone (or mics) only as necessary to capture the quieter timbres. I'd suggest getting several of the basic gestures as well, because this adds

to the authenticity of any performance or recording you might do with these samples.

Another classic Japanese instrument is the 13-string koto, which is also a descendent of a Chinese instrument, the zheng. Essentially, this is a nylon-stringed (traditionally silk) harp constructed from two pieces of wood, each about six feet long. The thicker piece is hollowed out to form a resonator; the other piece covers the bottom of the instrument and has a sound hole located at each end. Thirteen movable wood bridges allow the instrument to be tuned, usually with five notes per octave.

The koto sits flat on the floor and is plucked with plectrums on each of the first three fingers of the right hand. A wonderful vibrato can be produced by depressing the strings to the left of the bridges, and this technique also produces the distinct sharpening of notes that we associate with eastern music.

Another classic Japanese instrument is the shamisen, a 3-string spiked lute that has a square, skin-covered resonator (somewhat like a banjo) and a long, unfretted neck with exceptionally



large tuning pegs. The instrument is played with a huge plectrum (traditionally wood or ivory) that strikes the skin resonator as the strings are plucked, resulting in a very percussive sound.

The yueh ch'in is an interesting Chinese instrument from the family known as moon lutes. These have a large round wooden body, a short neck with very large tuning pegs, and usually two pairs of silk or nylon strings. The strings can be played individually or with a distinct tremolo (fast, repeated picking) technique.

Recording the yueh ch'in, koto, or shamisen will require a fairly quiet room because they are not particularly loud and their notes tend to die off rather quickly, thus allowing unwanted background sounds to creep in (an air-conditioning system can sound like a machine shop at times like this). I recently had great success using the Earthworks TC40K omnidirectional mics on these instruments because they let you get within an inch or two of the strings without the proximity effect causing the sound to get too boomy. In fact, these are great mics for most world instruments because of this capability.

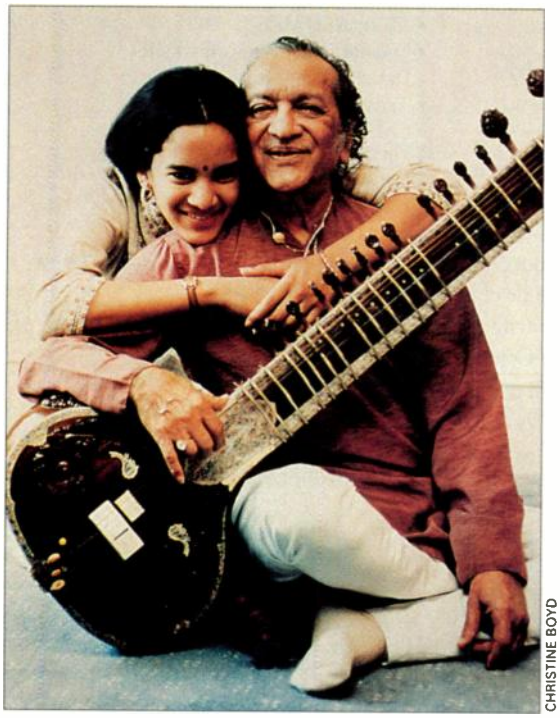
There are tons of other eastern instruments, including ceremonial bells, gongs, a wide variety of drums (such as the huge, well-known taiko drum). Each is worthy of mention, but we must move on.

SEAN MURPHY

Stephen Kent, shown here with his ornate didgeridu, performs in Trance Mission and Beasts of Paradise.

INDIA

Instruments from this region of the world could fill a book (in fact, several have been published). Of course, here we find the sitar, which you've probably seen many times in photos or on TV, but actually hearing or even seeing one in person is quite an experience. The sitar itself is a rather large instrument, with a resonator made of half a gourd, on which there is a wooden face and a large, rectangular, table-like bridge. The long neck is hollow, and the movable frets allow a wide variety of intonations. A second, smaller gourd sits at the end of the neck, behind the tuning pegs. There are five melody and two drone strings that are played with a plectrum as well as a dozen or more additional strings (sometimes called chakari or, more accurately, tarab) that vibrate sympathetically when the melody and drone strings are played.



CHRISTINE BOYD

Ravi Shankar is an acknowledged master of the sitar. He is shown here with his fifteen-year-old daughter and disciple on the sitar, Anoushka.

Very similar to the sitar is the vina, which is usually smaller but in some cases (as in the rudra vina of northern India) is much larger than the sitar. It is also a more ancient instrument, with fixed frets, and is played with the fingers rather than a pick.

In Indian ensemble music, either the sitar or vina may be accompanied by the tambura (sometimes spelled tamboura), which creates a fierce background drone from four strummed strings usually tuned G1, C2, C2 (again), and C1. Sometimes these may be tuned up a whole step. A small piece of thread inserted between each of the strings and the bridge creates the otherworldly buzzing resonance associated with this instrument.

Rounding out the ensemble is the tabla, a pair of single-headed upright drums that have tuned heads weighted with black tuning paste. Grab your *Revolver* album and listen to "Love You To," or pick up any Ravi Shankar recording to get a feel for how these instruments all work together. From personal experience, I can tell you that no recording can prepare you for the overwhelming sound of the real thing.

Different sitars can vary in volume, depending upon the materials and craftsmanship involved. Most are fairly loud (about the volume of a good acoustic guitar), making them fairly easy to sample in even a moderately quiet room. The tambura is usually quite loud and therefore subtlety must be used by players in an ensemble, but this works to our advantage: let that tambura player cut loose a little (tell him or her it's a solo). Tablas are fairly loud, about the volume of a set of congas, again making them easy to sample.

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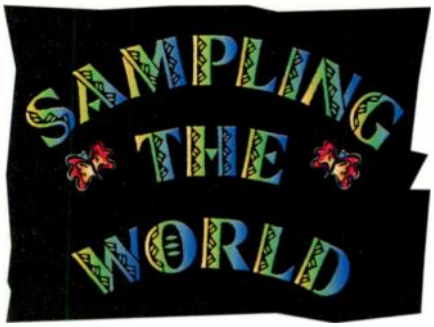


Space prohibits me from discussing some of the other exquisite Indian instruments I have had the good fortune to record, such as the saranji (a bowed instrument, almost a cross between a small sitar and a viola, that is played upright like a cello), bansuri (a beautiful transverse bamboo flute), and the shahnai (a reed instrument sometimes stereotypically associated with Indian cobra handlers).

OTHER COUNTRIES

I can't finish this article without mentioning two very distinctive instruments from the Pacific region. The first is actually not an instrument but a family of instruments from Indonesia known as gamelans. Claude Debussy heard these instruments at the Paris World Exhibition of 1889 and was the first in a long line of composers to acknowledge their influence on his own music.

"Gamelan" is the word used to encompass instruments such as the gong ageng (a large hanging gong), kenong (a set of horizontally mounted gongs), bonang (a double row of gong chimes), and gender (a metallophone with bronze keys suspended over resonating tubes). A thorough discussion of all the many instruments associated with gamelan (there are also many drums; the celempung, a zither-like instrument; and rebab, a two-string spike fiddle) is impossible here. Suffice it to say that there are many unique sounds to sample here. Most are fairly loud (some say too loud) and thus very easy to record.

A number of major universities have gamelan ensembles, so that would be a good place to start in your hunt for these instruments. Be sure to keep in mind that most of these instruments are played in microtonal scales, making them difficult to adapt to western music (although, of course, a sampler can pitch shift notes from any microtuned instrument to their nearest root note).

Finally, from Australia comes the didgeridu, the world's oldest trumpet. You hear this instrument every time there's a commercial or documentary on T V about Australia. The aborigines have played didgeridu for a great many centuries. Essentially, it is a three- to four-foot-long piece of wood (traditionally a limb that has been hollowed out by termites) that is open at both ends. Players sit with one end of the tube in their lips and elicit its fundamental pitch, over which rapidly changing syllables are pronounced to give the instrument its distinctive rhythmic drone. Breath is taken in through the nose while being expelled through the mouth (a technique known as circular breathing) so that phrases can be very long and complex. Compositions are made up of many of these elaborate phrases.

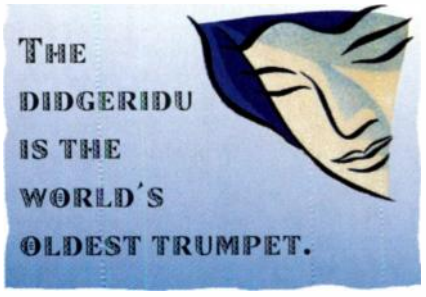
The didgeridu traditionally rests on the ground, but you might try having your player rest it on a light wooden stool. I've had success using an AKG C 414, placed about four feet away to capture some room ambience. Try using a second mic placed closer to the performer's face if you want to highlight

the distinctive lip sounds associated with this instrument.

JOURNEY'S END

It has been a fascinating journey for me over the last half-dozen years searching for and recording these wonderful instruments. I wish I had more space to tell you about some of the unique percussion instruments I have had the pleasure to sample as well as some fascinating Irish instruments such as the haunting Uillean pipes and Irish whistle. Suffice it to say that there are a lot of fantastic instruments in the world besides guitar, bass, and drums.

The beauty of sampling is that it allows us to add an enormous palette of distinctive, often one-of-a-kind sounds to our own personal musical vocabulary. What's more, aside from the end product (hundreds of megs of samples



and hundreds of hours of DAT tapes that I have yet to sample from), I feel fortunate just to have been in the presence of some of the great players who have been kind enough to allow me to record their performances.

Though there are a number of commercial sample libraries now available that offer the sounds of many of these instruments, I feel it is much more enriching to actually have the opportunity to hear these instruments for yourself and create your own samples, even more so than with any contemporary violin, clarinet, cello, etc.

Who could ever have imagined that it would take the latest breakthrough technology to finally bring the wondrous sounds of these ancient instruments into all of our lives?

Jim Miller's *samples have appeared in many sound libraries and ROM-based instruments, including those from Sweetwater Sound, Alesis, Roland, InVision Interactive, and Peavey. He has recently learned to play the didgeridu, though not very well.*

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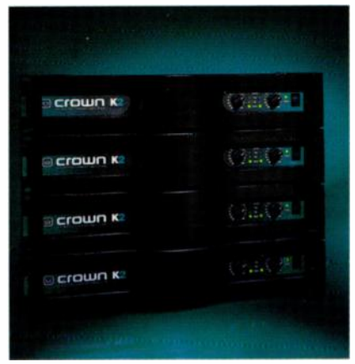

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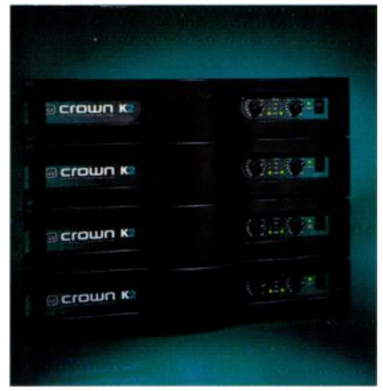

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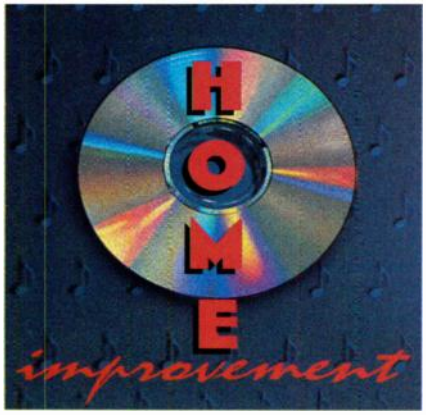
By Greg Pedersen

Illustration by David Bishop



THE ACOUSTIC FACTOR

The pop stars, producers, and business people who hire designers to build dream studios often have the luxury of creating the structure from scratch—or at least, they have the budget to gut an existing space and refashion it to their needs. The build-to-order process eliminates the need to deal with many acoustical nightmares because the problems are typically solved in the design.

Unfortunately, home recordists who seek to turn a bedroom or a garage into a studio seldom, if ever, have the option of knocking down a few walls. You're basically stuck with a room full of reflective surfaces and parallel walls that can twist and turn frequencies until the sound you perceive from your monitor speakers is pure fantasy. Obviously, this is not the place for critical recording and mixing.

"Ideally, you want a space with a frequency response as flat as possible," says George Newburn, a principal in the design firm Studio 440, who has designed studios for MCA Music Publishing and such artists as Danny Elfman and Peter Frampton. "You don't want a room with severe bumps or nulls at various frequencies because you'll never get an accurate representation of what you're hearing."

So how do you diminish your room's acoustic hellions without rebuilding the entire space? The solution is to minimize the room's effect on the listening position.

THE MIXING ZONE

Luckily, there are several ways to ensure that reflections and other audio anomalies will not dance around your ears. First, you must find the optimum spot to install your mixer because doing this will establish the all-critical listening position. Chips Davis, a world-renowned room designer whose current projects include the MGM Grand Hotel's EFX show in Las Vegas, recommends setting up the mixing area approximately one-third the distance from the front of the room.

"The mix position is supremely important," he says. "You can invest a lot of time and money creating a perfect room, but if you set up your console and speakers incorrectly, you'll blow the whole ballgame. For example, if you set up the mixer too close to the rear of the room, you'll get a lot of reflections that can cause frequency coloration bouncing back at you. By setting up your reference monitors toward the front of the room, however, the reflections will take some time to bounce back from the rear of the room, which causes your brain to kick in with the Haas effect [see the sidebar "The Haas Effect"]. The Haas effect essentially cuts off the rest of the room, so you're able to focus on the source sound from your speakers."

"In a bedroom, it's best to have the majority of the acoustic space behind you," agrees Gary Hedden, designer of The Bakery in Hollywood and many artists' studios, including rooms for Adrian Belew and Chester Thompson. "If you have an 11 × 14 bedroom, the speakers should be placed where most of the 14-foot dimension is behind you. You want the sound reflections to disperse as far away from your mix position as possible. The worst thing you can do is put your back against a wall because the sound from the monitors will immediately reflect off the wall behind you."

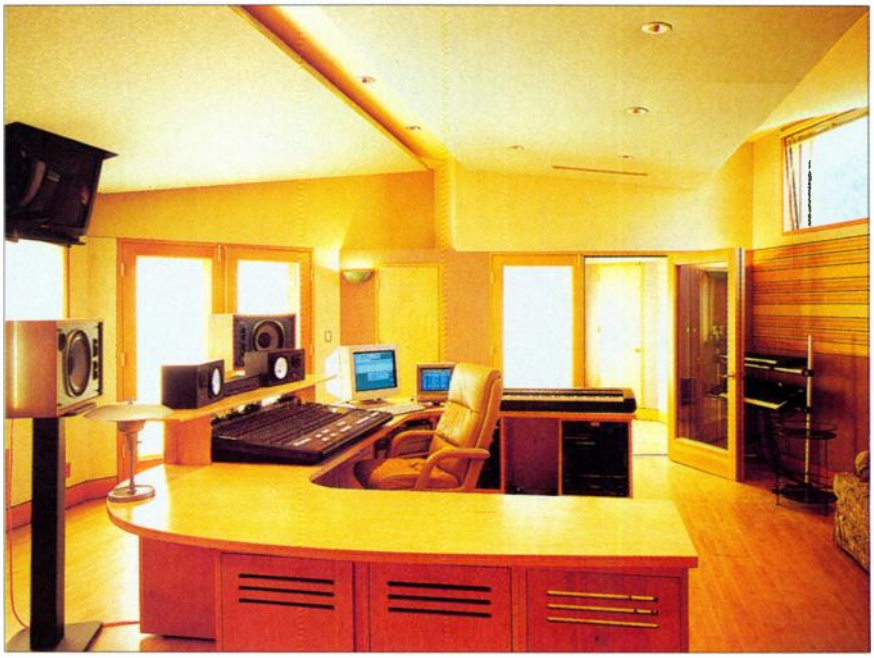
SPEAKER PLACEMENT

The positioning of the reference monitors in relation to the listener, the console, and the rest of the room also plays a substantial role in the presence of reflections within the mix field. To avoid having reflections mess with your tonal decisions, you need to have symmetry between each speaker and between each speaker and you.

"You need to create an equilateral triangle," says Jack Jacobsen, a designer and studio builder for members of Night Ranger and Jefferson Starship and for Windham Hill's production facility. "The distance between each speaker and the distance from each speaker to your head must be the same. For example, if the speakers are six feet apart, your head should be six feet away from each speaker. Otherwise, you will not have an optimum perspective of the stereo field, and secondary reflections may compromise critical listening."

The face of your console may also fall prey to reflections from speakers placed right on the edge of the console bridge. The solution? Get 'em back, Jack!

"You can't really put the speakers in a position where you don't have console reflection, but you can minimize it," says Bob Hodas, an acclaimed acoustician who has tuned hundreds of rooms as owner-operator of Bob Hodas Acoustic Analysis. "Setting your speakers on top of the console bridge is about the worst



JOSHUA WHITE

At Murielle and Joel Hamilton's studio in Pacific Palisades, California, studio bau:ton's designers moved the reference monitors back from the mix position to minimize console reflections.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thing you can do because the sound from the speakers bounces right off the console and into your face. If you place the monitors on stands and move them back eight inches from the console, you will change the angle at which the sound waves intersect with the console. Then most of the reflections will bounce underneath you and will not affect your frequency perception."

Davis also recommends attaching an absorptive cloth cover pad on top of the console meter bridge to minimize reflections from the edge of the bridge.

"The pad should be approximately one inch thick and can be mounted using Velcro," he says. "It should also hang over as much as possible without cutting off your view of the meters. The more the pad hangs over, the more it will prevent reflections off the face of the console. Posture counts when avoiding reflections, too. Make sure your seat is high enough so that your knees are almost up to the arm pad of the console. Sitting in this position will keep any reflections that bounce off the face of the console from hitting near your ears; they'll hit your stomach and chest instead."

STUDIO SYMMETRY

The glorious concept of harmony is critical in both life and relationships, and it's essential in the recording studio, as well. Changes in mass on either side of your listening environment can affect the balance of sound produced by your monitors.

"Symmetry is one of the most important aspects of building a studio," maintains Hodas. "If you don't have symmetry in the studio, there's a good chance you'll be plagued by bad stereo imaging from your monitors and the bass response will be screwed up, too. Let's say you have a bench on one side of the studio but not on the other. It will be more difficult for bass frequencies to move through the side of the room containing the bench because you have a greater amount of mass on that side. Obviously, the situation can interfere with an accurate assessment of bass levels while you're mixing.

"You also want any acoustical treatments to the space to be symmetrical," continues Hodas. "If you have an untreated window on one side of the room and you've mounted some foam on the other side to kill reflections, that imbalance will really mess up your imaging—especially if this setup is in the front part of the room, where you're also getting first-order reflections bouncing back to the mix position. What will happen is you will get this comb-filtering effect that will skew the frequency response of one side or the other."

Peter Grueneisen, ace designer at studio bau:ton, whose credits include Heart's famous Bad Animals studio in Seattle, agrees that it's critical to ensure equal, balanced frequency response from the left and right monitors. "When you're sitting at the board in the mix position and you look to each side, the conditions should be the same," he says. "For example, you don't want to have an enclosed wall on one side and a big opening that goes to another part of the house on the other."

Hedden elaborates further: "If you draw a line through your mix position to the middle of the stereo spectrum, everything on the left should be identical to everything on the right. This includes wall treatments and the shape, position, and amount of equipment. Any reflection, absorption, or resonance that happens near the speakers, or between the speakers and you, should be identical on the left and right. Otherwise, your perception of center as well as your perception of left-right balance will be funny. If it's not possible to maintain strict symmetry, you can reduce the sonic damage by positioning asymmetrical elements below the level of your shoulders. Setting up the studio in this manner should ensure that any unbalanced reflections occur below the mix field."

BASS BUGABOOS

Within the studio environment, bass frequencies can generate enough sonic gremlins to earn public-nuisance status. Parallel walls, for instance, can cause bass frequencies to bounce around the room, producing nasty peaks and dips in low-end response. However, this wily enemy has a weakness: bass frequencies tend to build up in the corners of a room, so an absorptive apparatus called a *bass trap* can be positioned in these areas to minimize low-end woes.

Unfortunately, bass traps can usurp space that's not readily available in a cramped home studio. The depth of a bass trap determines what frequencies it can absorb, so the longer the length of the bass wave, the more depth you'll need to counteract it.

BENNY CHAN

In this room designed by George Neuberg of Studio 440, note how symmetrically the mix position has been laid out. Consistent elements on the right and left help maintain proper stereo imaging.

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"Bass traps often require quite a bit of space to be effective," says Hodas. "For example, at 100 Hz you have to deal with a 10-foot wavelength, so you're going to need two and a half feet of space to absorb it. Whatever length the bass wave is, it will take a bass trap that is one-quarter the size of the wave to adequately absorb it. So if you have a big bump at 40 Hz—which produces a rather massive sound wave—you probably will not have the available space in your room to construct a trap large enough to tame the bass."

If lack of space is a serious problem in your studio, you may want to consider using a parametric equalizer to help your playback system reconcile the room's tonal idiosyncrasies. An outboard, dual-channel equalizer can be inserted into your console stereo bus to cut (or boost) problem frequencies until the sound produced by the monitors is relatively flat within your specific listening environment.

"A parametric EQ, such as Meyer's CP10S, is one way to control bass frequencies if you don't have the room to build proper bass traps," says Hodas. "Equalization will not solve the problem acoustically, but it can, at least, help you create a space you can mix in with some confidence."

If you're lucky enough to have a big room, preconstructed bass traps are available from many manufacturers (see the sidebar "Studio Helpers"). These can be rather expensive, so budget-conscious recordists may want to consider building their own bass traps (see the sidebar "DIY: Corner Bass Trap" on p. 72; also check out "DIY: Build a Better Bass Trap" in the June 1995 **EM**).

THE HAAS EFFECT

The Haas effect relates to the localization of a source sound when the signal reaches a listener's ears at slightly different times. "Basically, the brain perceives reflections that reach the ear from a source sound within nineteen milliseconds as part of the direct sound," says acoustician Bob Hodas. "Essentially, the brain can't identify those reflections as being separate from the direct sound. This helps the quality of sound within the home studio because some problematic reflections may be 'canceled out' by the brain before they compromise the listening environment."

The Haas effect can be classified as an example of sensory inhibition because the response to one stimulus (in this case, the direct sound) causes the response to another stimulus (the reflections) to be inhibited.

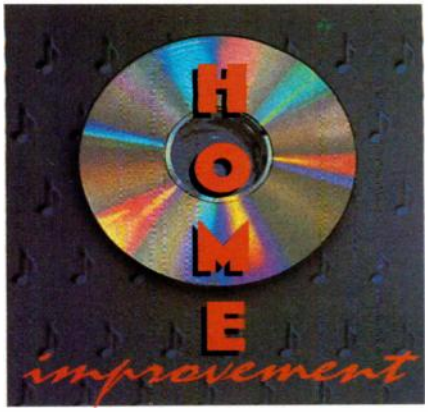
It's important to note that any home-brewed bass traps often require spin glass (the term *Fiberglas* is actually a trademark of Owens Corning), particles of which can float loose in the air and cause a health hazard. To keep the air in your studio as free of spin-glass particles as possible, be sure to wrap these materials in fabric.

"Make sure to use fabric that's acoustically transparent," says Newburn. "A quick test for this is seeing whether you can breathe easily through the fabric. If you can, then sound can move through it, too. Upholstery fabrics aren't a good idea because they're actually dense enough to become reflective."

Well-designed studio furniture, such as this Pro Station by Omnirax, can help you work within your monitoring sweet spot, minimize system clutter, and prevent problematic tonal reflections and resonances.

For the construction-challenged studio owner, Hedden suggests an extremely easy-to-build bass trap that could probably be assembled by a class of earnest sixth-graders. "The simplest form of bass trap is a fiberboard or plywood panel placed diagonally across the corner of the room," he says. "You just place Fiberglas behind the panel, and you're done. The depth of the panel will, of course, determine at which frequency the trap works. I would suggest a typical arrangement of making the panel one foot deep on one wall and two feet deep on the other wall to create a sort of triangular cavity."

According to Jacobsen, one of the cheapest bass traps may already be located in your room—a bed. "Futon beds make great bass traps," he says. "Futons are nonreflective and absorb



low frequencies very well. Any soft furniture, in fact, that is placed in the corners of the studio can help control bass frequencies."

HIGH-FREQUENCY HORRORS

Now that we've dealt with some low-frequency reflection problems, let's explore some solutions to high-frequency reflections. For example, if you think a mirror does wonders reflecting that cute mug of yours, you'll be happy to know it can also expose mid- and high-frequency reflections that can compromise the listening environment.

"Above 400 Hz, sound waves act like light beams, so you can accurately predict the location of first-order reflections by using a mirror," says Hodas. "Sit at the mix position and have a friend take a mirror and move it around the studio's side walls and ceiling until you see the front of the monitor. If you see the speaker reflected in the mirror, you're also getting a sound reflection from that point."

You should also use the mirror trick to hunt for reflective surfaces on all your equipment racks. Most racks are constructed of metal alloys, and metal *loves* to reflect. The angles of your racks and other gear, in relation to the reference monitors and the mix position, are usually responsible for how much these surfaces reflect problematic sound waves back to the listening area.

"You can use two mirrors and a flashlight to test whether or not the surface angles are going to cause reflections in the mix position," says Newburn. "Put a mirror on a reference monitor and another mirror on a piece of equipment you believe may be reflecting sound into your mixing area. Then,

DESIGN COUNCIL

If you're diving into building your own studio but want a "lifeguard" handy, the designers interviewed for this feature are available for consultation. However, please keep in mind that these people are busy professionals. In other words, don't embarrass this magazine and your fellow **EM** readers by trying to wrangle free advice.

Bob Hodas Acoustic Analysis
tel. (510) 649-9254
fax (510) 649-1882
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tel. (510) 827-3263
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e-mail chips-davis@worldnet.att.net
Gary Hedden
tel. (615) 794-0550
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studio bau:ton
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from the mix position, point the flashlight at the speaker. If the light reflects off the speaker mirror, then off the other mirror, and back to the mix position, you have a problem."

If you discover a rack that is tossing "nasties" into your ear space, Davis suggests orienting the rack (or other piece of furniture) so that the sound waves are reflected straight to the back wall and away from the console. "I'd also try to absorb some of the reflections by putting 1-inch thick, cloth-covered Fiberglas on the side of the rack facing your speakers," he says. "Even if you've taken care to ensure that a rack reflection won't come back into the mix position, it'll still bounce someplace else."

Windows are also notoriously reflective surfaces, but it's often difficult to banish them from your studio space. After all, a nice view (and natural light) provides a soulful contrast to the lifeless aura of a room crammed with recording gear. Newburn has a solution for how you can enjoy a window to the world without risking rampant reflections: "Put up a heavy-duty velour curtain that you can pull across when you want to do some critical listening. Velour is a dense fabric and will tend to absorb certain frequencies rather than allow them to pass through and reflect back from the glass."

THE ISOLATION FACTOR

Now that we've covered the concepts and materials inherent in improving your room's acoustics, it's time to focus on the concept of sound isolation. Obviously, in most home-studio situations, housemates and neighbors are close enough to be driven mad by the noises (both good and bad) that you produce.

In a worst-case scenario, eviction can be the reward for your finely honed musical masterpieces. Good manners—such as turning down the volume during early morning and late-night hours—can help, but courtesy is usually a doomed "quick fix." Sooner or later, the noise factor is going to upset somebody. Hedden suggests wearing headphones and close-miking acoustic performances to keep sound levels on the soft side.

"In the studio, if you monitor through headphones and put microphones close to what you're trying to record, you'll achieve pretty good sound-level management," he says. "Try to keep the volume levels of the performances down so that the acoustic levels and the monitoring levels are relatively equal—which means equally *quiet*."

Of course, you can't always manage sound levels so precisely. A snare drum overdub, close-miked or not, is going to be loud enough to upset a sensitive neighbor. And most vocalists can sing you out into the street, too—especially if you're running the same section of a song over and over to get the part right. One way to avoid these problems is to add enough mass to your room so that bass waves and other frequencies can't simply stroll through your walls and disturb the peace.

"Doors and windows are the weakest link, in terms of construction isolation, because you can't insulate them like you can a solid wall," says Hedden. "However, there are workable solutions. A solid-core exterior door—that is, a wooden exterior door measuring 1¾ inches thick with a jam and sill—is a cost-effective way to have a door capable of creating a relatively airtight seal. Thermal seals that are designed to keep warm air in and cold air out are also effective options for making a door as airtight as possible. In addition, retracting thresholds that come down against the carpet to seal the bottom of the door are available. Door handles can also be a problem because common, through-the-door assemblies invite sound leakage. It's better to have a door that can be pushed open from one side so that no handle is necessary. This way, there is no front-to-back opening in the door through which sound can escape."

Newburn suggests mounting weather stripping to cover the tops and sides of the door. (Of course, the stripping must be installed on the entry *opposite* the side where the door opens into the room.) "You should also put a seal on the bottom of the door, but this can be difficult in a carpeted room because a carpet doesn't provide a solid surface upon which to form a tight seal," he says. "However, you can work around this by stuffing towels under the door while you're working."

As mentioned earlier, windows can be an acoustic problem because they are a reflective surface. But they can also cause isolation problems because single-pane glass is far from soundproof.

"To try and keep sound from leaking out of windows, I make a foam block that can be inserted into the window sill," says Jacobsen. "I use rigid installation foam, the kind that has an aluminum backing, and I put the aluminum side to the glass because it reflects light and heat back outside. You can just set the foam block in the window to achieve isolation while you're recording and mixing and then pull it out when you're finished working."

DIY: PORTABLE ISOLATION BOOTH

Space is always limited in the typical bedroom studio, so it's usually impossible to build a vocal isolation booth into a personal studio. Fortunately, you can make a portable iso booth that can be set up when needed and broken down for easy storage when the session is over.

"First, get three unfinished sliding closet doors," says studio designer and musician Jack Jacobsen. "The doors are lightweight and hollow on the inside, so they're easy to work with. Then hinge the three doors together with hinge pins. To ensure the sound of the booth is conducive to cutting vocals, it's essential you attach absorptive foam on the inside of each door to help diminish sibilance and high-frequency reflections. If you feel it would help the performer to have visual cues, you can easily cut an eye slot into the door.

"Now, position the three-sided booth in a corner of the room to form a five-sided area," Jacobsen continues. This configuration minimizes the possibility of reflections from parallel walls. You should also hang shipping mats on the back wall to help absorb the lower frequencies of the voice that occur at around 200 Hz or 300 Hz. Again, the use of absorptive materials is critical because you want this area to be as nonreflective, dead, and anechoic of an environment as possible. When you're done, you just pull the hinge pins, break the booth down into pieces, and store the unit away."

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TUNING YOUR ROOM

Unfortunately, the mirror trick we discussed earlier isn't powerful enough to be the sole crusader in the witch-hunt for frequency anomalies.

To really get the nitty-gritty on your room's acoustical problems, you need to have the studio "tuned." This process—which is also referred to as "shooting the room"—involves testing the response of your monitors as they react to the room's natural modes.

"The height, length, and width ratio of any space has natural resonances, or *modes*," says Davis. "If you're singing in the shower and you hit a note that fills the whole space without your having to put any energy into singing it, you've discovered a natural resonance of that space. Or let's say that a room's natural resonance is at 39.7 Hz, a low E. If I play an open-string, low E note on an acoustic guitar, the natural resonance of the space will promote and sustain that note."

Obviously, if certain frequencies are accentuated by your room's natural modes, you may be "tricked" into cutting those frequencies when you mix because they'll sound more prominent than they really are. Thus fooled, your EQ tweaks may actually weaken the tonal spectrum of the mix. Once again, the acoustic personality of the room is inhibiting your ability to make critical, informed sound-sculpting decisions.

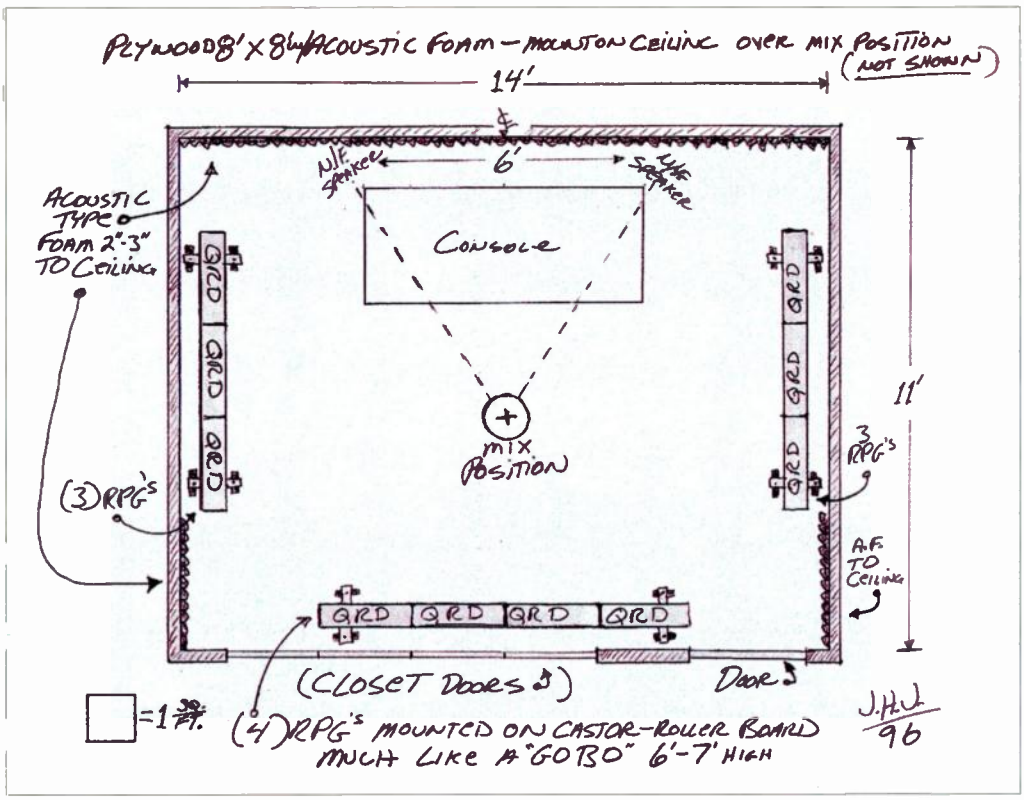
In addition to revealing problematic resonances, shooting your room is a good way to find out whether the phase relationships of your speakers and other recording equipment are correct. The process can also determine which surfaces of the room need to be treated with frequency-absorbing material (to make the space more acoustically "dead") or frequency-diffusing material (to scatter reflective waves without absorbing them).

"If you're serious about what you're doing in your studio, getting the room tuned is critical," says Hodas. "If you can achieve a fairly linear frequency response between your room and your recording system, you'll be able to trust your ears, work faster, and have more fun. For example, it's a lot of grief if you always take your mixes out of the studio and play them on three or four different systems—your car, your friend's home stereo, a boom box, and so on—to ensure that they translate to the outside world. You should be able to produce good mixes on your speaker system and also be able to trust that the sounds you're hearing are relatively accurate. Of course, you can always get used to the idiosyncracies of your room and do whatever it takes to make it work for you, but what if you're working with other people? You'll be constantly explaining why nothing sounds quite right in your studio."

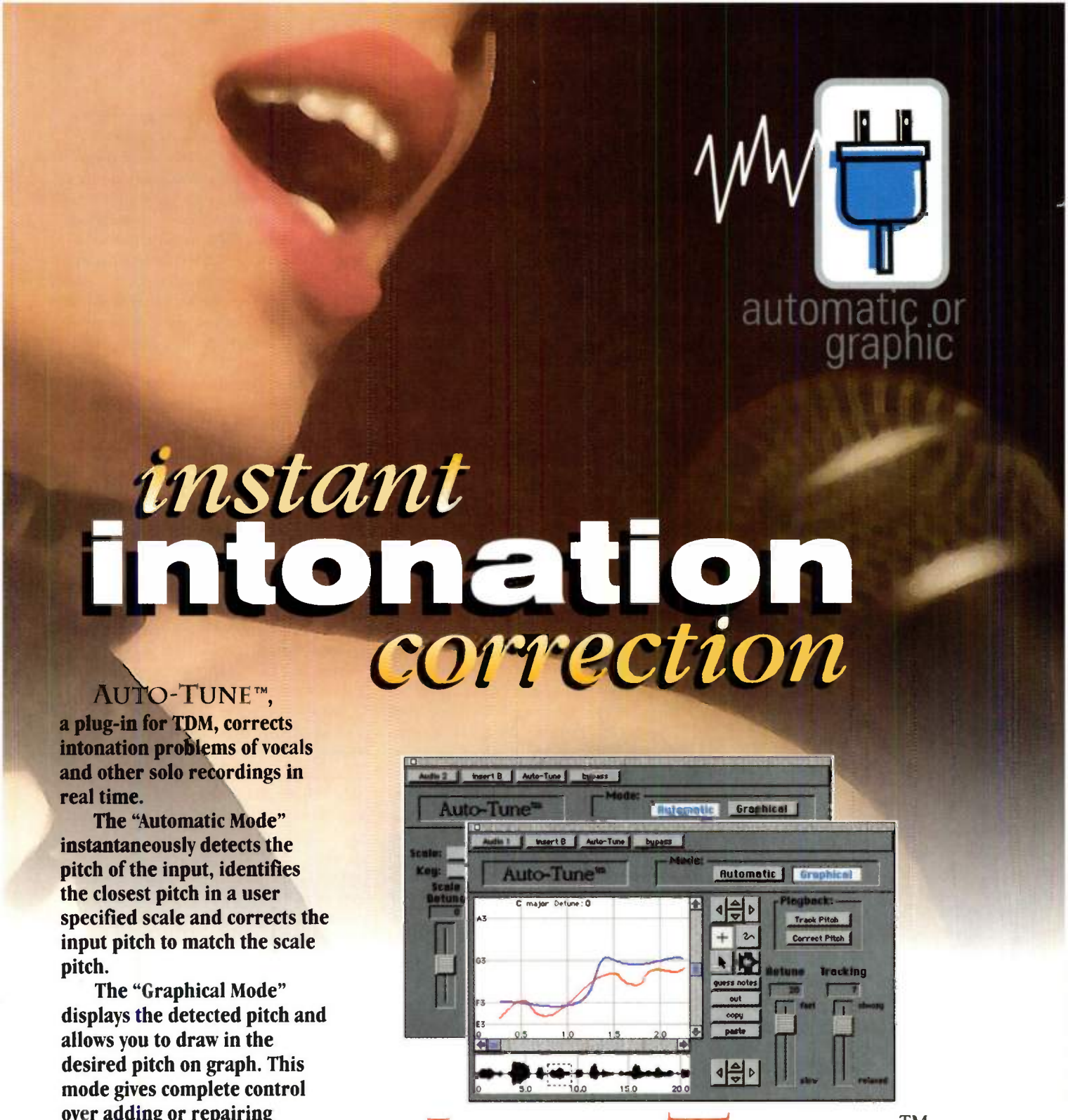
The decision as to whether you shoot the room yourself or hire a trained professional is largely dependent on how much bread you can spare and the breadth of your knowledge concerning the science of sound. If you endeavor to tune your own room, start the process by listening to some favorite albums in the space. Ideally, these productions should represent the styles of music and instruments that are likely to pop up in your recording situations. As the music plays, try to determine any boosts or drop-offs in frequencies that you do not hear when listening to the piece on headphones or other playback systems.

Although this is a *very* basic way to test room response, a casual listening test should reveal potential tonal lapses in your recording space. You should also purchase a test-tone CD, such as the *Mix Reference Disc* (available from Mix Bookshelf; tel. 800/233-9604 or 908/417-9575; fax 908/225-1562) to check your room's reaction to tone bursts at specific frequencies.

"A test-tone disc gives you an idea of the modal stability of the space: whether certain notes are going to sustain or die," explains Davis. "Essentially, you're listening for the results of one tone burst frequency to another tone-burst



Jack Jacobsen drew up this basic design for a bedroom studio exclusively for EM readers. Thanks, Jack!



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frequency. Let's say the 40 Hz burst goes off and you hear this 'buuung' sound that really sustains. Then, the next tone burst at 50 Hz dies really quickly. Now you know that your room's modal resonance is supporting 40 Hz, which is why that frequency is ringing on you. Obviously, that shows 40 Hz is going to be more prevalent in the space when you're recording and mixing."

A less subjective method of measuring the response of your room is to rent a ⅓-octave-resolution real-time analyzer (RTA). This device is essential for do-it-yourself room shooting because it measures sound-pressure levels at 31 different frequencies. A good RTA can cost you $900, but you can rent units for approximately $100 per day.

Although an RTA will not tell you *all* the information needed to tune a room, it will uncover problem frequencies with more precision than the untrained ear.

"Even a cheap RTA has a variety of uses beyond tuning a room," says Jacobsen. "When I'm in the studio, for example, I leave the RTA's microphone turned on the entire time I'm working. If some frequency suddenly starts feeding back, all I have to do is look at my RTA and—bingo!—the unit pinpoints the offending frequency. You can also use an RTA as an ear-training device to help you locate specific frequencies within the sonic spectrum. Pretty soon, when you start to EQ a mix, you'll be able to determine whether your tweaks are actually doing something.

"However, it's important to note that an RTA is not an end-all solution," Jacobsen continues. "If you equalize your monitor system until it delivers flat frequency response when measured on an RTA, that doesn't necessarily mean you have improved the sound of your room because an RTA doesn't tell you anything about reverberation times or phase relationships."

Jacobsen's statement shows the danger of taking room-tuning into your own hands: what you don't know *can* hurt you. To enhance the room's sound, it's important to understand the science of acoustics so that you know why you're messing with a given frequency.

"It will do no good to try to EQ a playback system to fill every hole you find in a room's frequency response if the problem is a physical one," says Davis. "If you just try to EQ the holes, you will smear the audio spectrum immensely. For example, never take an RTA, set it on a mic stand at the mixing position, and then start EQing the low end. If you have an 8-foot ceiling, you are dealing with a natural floor-to-ceiling frequency cancellation. That's the laws of physics at work; you can't make that problem go away by using EQ."

Given the pitfalls of DIY room shooting, hiring a professional may be the best route for you and your room. You should be prepared, however, to spend anywhere from $500 to $1,000 to contract the services of a local acoustician. To get a better idea of what you're getting for that fee, we asked Hodas to chronicle some of the services he provides and the equipment he uses when tuning a room.

"I tune studio spaces with a Meyer Sound SIM System 2, and I had to pass a rigorous course by Meyer Sound Labs to be accredited on the system," he says. "Meyer understands that anyone could go out, buy a test system, and do bad work, so their course is very thorough. The SIM System makes it possible for me to do my analysis at an extremely precise ¹⁄₂₄-octave resolution. By comparison, the ⅓-octave resolution of many less-expensive systems affords a very cursory kind of analysis. If I used one of those systems, it would be like telling someone, 'I will tune your piano, but I'll only tune twelve of the keys!'

"When I tune a studio, I'm looking at 245 points in that room," continues Hodas. "For example, if you have two or more sets of reference monitors, I'll shoot them all to ensure the polarity is correct on each and every one. This is critical, because if your speakers have an absolute polarity shift, elements such as vocals will move back and forth when you switch from one set of speakers to another. You want the left and right speakers properly in phase and all the tweeters moving in the same direction. With SIM, I can spot a horn out of phase quickly because I'm able to look at phase relationships constantly in real time."

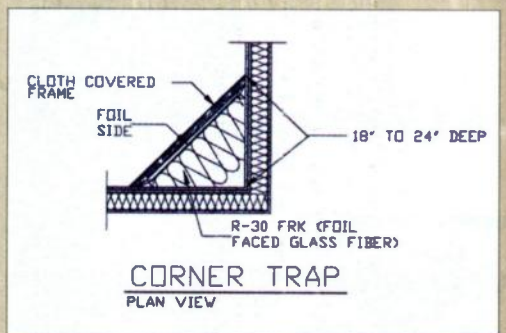
Hodas also does a tone-impulse-response test to identify the reflection patterns in the room. He even checks the console stereo bus to determine whether any electronic malfunctions are messing with your tonal perceptions.

"If you've got a left and right mismatch in your console that is more than two-tenths of a dB off, it's going to pull the stereo image to one side or the other," says Hodas. "The SIM system has resolution down to one-tenth of a dB in amplitude, so I can look at the phase of the console and see whether any of its capacitors are starting to go bad—a situation that can throw off phase and frequency response."

DIY: CORNER BASS TRAP

Building a bass trap can be as easy as going down to your hardware supply store and buying a bundle of Owens Corning R30 Fiberglas. Then build a simple wooden frame, and mount it diagonally, approximately 24 inches out from the corner of a wall (see diagram below). The frame should be tall enough to cover the entire corner; you want the low-frequency focus taken out of the listening environment. Now stretch a cloth over the frame to hide the Fiberglas, but be sure to keep the foil side of the Fiberglas facing out into the room. Mounting the Fiberglas "foil out" is really important because tests show that this foil acts as a membrane absorber that knocks the heck out of bass waves.

—Chips Davis



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POWER STRUGGLES

Acoustic anomalies are not the only factors a studio designer must take into account when building a recording facility. These days, even a modest home studio is a neural net of wires, patch cables, and power cords, and every piece of electronic gear must plug into a wall socket. Electrical power is the lifeline for your recording gear, and the studio's power sources must be absolutely uncompromised. If not, you may be plagued by buzzes, hums, and other noises that can ruthlessly trash a session. In addition, power surges, brown outs, and blown circuit breakers can put expensive recording and music gear at risk of serious damage. (For a more comprehensive tutorial on power and grounding, see "On Solid Ground," parts one and two, in the September and October 1992 issues of **EM**.)

To start the process of creating a safe, effective power source, Hedden suggests studying the branch circuits supplying electricity to your home. "It's important to know what else is plugged into the outlet that you're using for recording," he says.

Jacobsen has an easy method for identifying shared circuits that could cause power problems in your studio. "The first thing you want to do is go to your breaker box and find out which breaker goes to the room you're using as a studio," he says. "Turn on a small table lamp and plug it into all the sockets; if it goes off every time you throw the breaker, then all the sockets in that room are on the same breaker. There are usually only four sockets in a conventional bedroom, but because you can put eight sockets on a single breaker, two rooms may share that one breaker. It's critical to determine whether your studio is on the same breaker as another room. For example, if you share a breaker with your kitchen and someone turns on a blender while you're mixing, you'll probably have a horrible buzz and hum coming through your monitors. If you're tracking, that sound will get recorded along with the instrument or voice."

To prevent this frustrating scenario from messing with your patience and productivity, you should dedicate—and isolate—one breaker to your recording space. (Be sure to identify it with a label that says "studio.") You'll probably have to call in a professional electrician to make the adjustment, but the expense is worth its weight in studio hours. After all, you obviously can't work if dinner preparations, house cleaning, and other activities are constantly throwing noise into your recording system.

"You also don't want to plug *anything* in that draws a lot of current, such as a portable heater or fan, on the same line," warns Jacobsen. "There may be only twenty amps going to that circuit, and your breaker will click off if you draw more than that twenty amps. And guess what? A 1200-volt heater alone draws ten amps. Luckily, you can have a massive quantity of stuff—say, a setup with three ADATs, a big

mixing console, two racks full of outboard equipment, a pair of speakers, and a monitor amplifier—and probably draw only seven or eight amps of power. But if you start plugging in bass amps, subwoofer systems, and anything else that swings a lot of current, you'll be popping the circuit breaker and interrupting your session."

Some homes mix 15- and 20-amp circuits, and **EM** Senior Editor Steve Oppenheimer strongly suggests having the 20-amp circuits service your studio. Check your breaker box to confirm the power situation.

"I've run home studios on 15-amp circuits for years, but 20-amp circuits give you more power," says Oppenheimer. "It's better to have some headroom so that you're not stressing your power with whatever you're doing in the studio. An electrician can switch you over very easily to twenty amps. It's really no big deal, and you should have it done."

The quality of power in your studio is another factor to seriously consider. "You certainly want to protect your thousands of dollars worth of equipment by plugging into good electricity," says Jacobsen. "*Bad* electricity is caused by certain situations such as a socket without a neutral wire, the ground and the neutral wires being tied together, or reversed neutral and hot wires. I recommend getting a phase-test plug so you can check the wiring yourself. These units only cost about five bucks, and they can save your gear. You just plug it into every socket in the room, and it will tell you whether any of the outlets are wired wrong. Of course, if you discover any problems, you should hire an electrician to correctly wire the system."

To further protect your recording gear, Hedden recommends buying an isolation transformer. "These units cost approximately two hundred bucks and will help improve your system's signal-to-noise ratio considerably," he says. "For your computers and related equipment, I would also recommend an uninterruptable power supply [UPS]. You can get one for about a hundred and fifty bucks, and they can really save you a lot of grief if your house power shuts down. If you're sequencing, for example, and there's a little glitch in the power, you won't lose your RAM. The UPS has a battery backup, so if there's a power failure, you'll have a few moments to save your work and turn everything off."

ILLUMINATION

In the rest of your house, light dimmers can provide a subtle hint of mood lighting. In your studio, however, the only moods these dimmers will produce are anger and frustration. Light dimmers typically create annoying buzzes that can toy with your concentration and even sneak into your recordings.

"That buzzing sound can get into everything, including mixing consoles and microphones," says Hedden. "If you do have dimmers in your studio—and I recommend that you don't—a good test to see whether you've got a problem is to grab a battery-operated AM radio and tune to a place where there isn't a station. If all you hear is hiss, you're okay, but if you hear a buzz, it's probably coming from the dimmer. You also want to avoid fluorescent lights because their transformers produce a horrible hum.

"In addition, be careful when using halogen lights because some models employ a switching power supply to make a lower voltage," he continues. "I plugged one of these low-voltage lamps into my desk and had a computer fail as

a result. I wasn't able to use my mouse. I would stick to the standard 120-volt lamps rather than the low-voltage lights. All in all, incandescent lights are really the best choice."

THE COMFORT ZONE

Let's face it: the space in your bedroom studio is pretty darned cramped, and it's often difficult to place all of your equipment in convenient and easy-to-reach places. Every time you reach for a piece of outboard equipment, it's two feet away from your mix position, so your body contorts like William Hurt's half-man/half-ape character in *Altered States*. Not good! Fortunately, well-designed studio furniture can help you keep everything within easy reach.

"Musicians buy all this gear, but they don't think about where they're going to put it, so most of the stuff just sits on the floor," says Phillip Zittell, vice president of sales and marketing at Omnirax. "We offer an alternative where everything in your studio can be positioned so that it's ergonomically sensible. We even measure the overall height of our workstation units to ensure that reference monitors can be placed in the right position."

And speaking of position, we already know that the mix position in your studio is critical. However, many recordists will compromise ergonomics by plopping a console onto a cheap folding table or a makeshift bench. Such kludges invite problematic frequency reflections and may force you to move outside of your monitors' sweet spot to make sound adjustments. Specially designed studio furniture can alleviate mix position ills, but a new type of furniture—the console housing—can do that *and* make your Mackie 8•Bus or Yamaha 02R look as slick as an expensive Neve.

"Like a lot of people, I bought a Mackie 8•Bus," says David Atkins, founder and president of Argosy Consoles. "My wife had hoped I was getting something that wouldn't have all these wires hanging out the back, so I told her I'd try to find something nice to put the console in. But because I couldn't find anything that professional, we starting sketching out a housing for the mixer to emulate a large-format console.

"We discovered that by dropping the 8•Bus into the housing, everything could be on one control surface, which creates a much more organized work space," Atkins continues. "For example, you can integrate your peripheral equipment into the console work surface instead of having to put your outboard gear into a rack that's out of the speaker plane. Obviously, it's inconvenient to reach behind you or to the side every time you want to tweak a compressor. This way, you can just reach up on the console and adjust compression parameters without moving out of the monitoring sweet spot."

However, whether you use home-built or custom studio furniture, keep in mind that any furniture can resonate at frequencies that can cloud

your perception of the audio spectrum. "The furniture in your studio should be acoustically transparent," says Hedden. "Stands that support the gear where you want it but don't have a lot of resident cavities are good choices. A simple test for resonance is to simply knock on the furniture. You shouldn't hear any ringing tones, hums, or rattling. Obviously, the overall construction should feel very solid, too."

Comfort involves more than just ergonomics, however. Spending hours (or days) in your now-soundproofed home studio—sometimes with other musicians or technicians present—will definitely increase the "funk" factor. And we're talking about sweat and stuffy air here, not rhythmic grooves.

"There will be more heat in your recording space than normal due to the gear, lights, and people," says Hedden. "You'll also have reduced air flow because you've sacrificed ventilation for soundproofing. To keep the studio from becoming a sweat box, you can install an independent cooling unit outside the room with vents that can blow in cool air. These systems have all the

FEAR OF FLUTTER ECHO

Do you remember the echoing "chu-chu-chu-chu" sound effect in those *Friday the 13th* horror movies? The eerie tone is an example of flutter echo—a distortion of a source sound that drives some studio engineers and producers to distraction. The easiest way to check your recording space for this acoustic imp is to walk around the room clapping your hands. If you come to a spot where you hear the "Jason effect," you're the proud parent of a flutter echo. However, according to designer Chips Davis, the presence of flutter echo doesn't mean you're working in an acoustically unfit studio.

"It cracks me up sometimes when I see people walk into a room, clap their hands, and pronounce, 'Oh no, there's flutter echo in here!'" he says. "If you clap your hands in a space and you hear flutter echo, you have to be directly in the line of the echo for it to be harmful. If it's bouncing at the other end of the room, you will not hear it when you're mixing or recording. I mean, have you ever heard flutter echo on a speaker? If you think so, just take your drum machine, call up a snare sound, and run it through your speaker system. Now, if you don't hear the flutter echo bounce around where you're sitting to mix, don't worry about it. You should only be concerned with the listening environment you're in.

"If you *do* hear that echo, simply change the angle of the speaker, and it should disappear," Davis continues. "Even if you have the worst case of flutter echo imaginable, all you have to do to cure the problem is to treat a front or back wall and one side wall with absorptive foam and angle your speakers correctly. Do that, and people will be able to come in and clap their hands all they want!"

noisy parts in the box that's mounted outside, so the noise levels inside your studio can be kept fairly low. You can also add some vents to your central air-conditioning system—if you have one—that service the studio space."

"A lot of the guys I know have separate, window-mounted air conditioners in their studios," offers Hodas. "These are single-room air conditioners, much like the type you find in some cheap motels, that you can run periodically to keep the temperature down. Obviously, you don't want to run the system while you're doing critical listening, but you can usually keep the room comfy by turning on the system while you're setting up or composing."

In a really tight-budget scenario, you can keep the air moving by having small desktop fans blowing across equipment racks and the mix position. Again, you don't want the fans chattering when you're mixing or recording, but keep

STUDIO HELPERS

If you're really ambitious enough to build a serious home studio, you'll need to hook up with some manufacturers that make (or distribute) the necessary materials. Please keep in mind that this is not designed to be a comprehensive list; it's just a quick reference guide to get you started on your buying spree.

Acoustic Sciences Corporation
Tube traps
tel. (800) 272-8823 or (541) 343-9727
fax (541) 343-9245
e-mail ttrap1@aol.com
Web www.continet.com/bdcrow

Acoustical Solutions
Acoustic foam
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fax (804) 346-8808
e-mail acoustic@richmond.infi.net
Web www.acousticalsolutions.com

Argosy Consoles, Inc.
Housings for Mackie 8•Bus mixers
tel. (573) 348-3333; fax (573) 348-2769
Web www.argosyconsole.com

Auralex Acoustics
Acoustic foam
tel. (800) 95-WEDGE or (317) 842-2600; fax (317) 842-2760
e-mail auralex@auralex.com
Web www.auralex.com

Bryco Products
Racks and storage cabinets
tel. (310) 305-0317; fax (310) 305-9167
e-mail bryco@primenet.com

Folded Space Technologies
Acoustic panels
tel. (404) 633-2507; fax (404) 321-5094
e-mail fspace@mindspring.com
Web www.mindspring.com/~fspace

Illbruck, Inc.
Acoustic foam
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fax (612) 521-5639
e-mail 73133.1205@compuserve.com
Web www.illbruck/sonex.com

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Noise-control doors and windows
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Web www.industrialacoustics.com

Littlite/CAE, Inc.
Console work lights
tel. (810) 231-9373; fax (810) 231-1631

Omnirax
Racks and workstations
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fax (415) 332-2607

RPG Diffusor Systems, Inc.
Acoustic foam
tel. (301) 249-0044; fax (301) 249-3912
e-mail info@rpginc.com
Web www.rpginc.com

Rack Craft
Desktop racks
tel. (913) 262-3949
e-mail rackcraft@aol.com

Silent Source
Acoustic foam
tel./fax (413) 584-7944
e-mail silent@crocker.com
Web www.crocker.com/~silent

Taytrix
Modular gobo systems
tel. (201) 222-2826; fax (201) 222-5457

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them running every single second your ears are not on duty. It also helps to crack open the available windows and doors while the fans are going so that the hot air can escape.

Now taking frequent breaks to "fan out" the room on hot days may not seem like a productive use of studio time, but cranky musicians and engineers often work very, very slowly. (And falling asleep on the mixing console because you've gotten all warm and toasty isn't very productive either, is it?)

END OF SESSION

Hopefully, these tips, tricks, and applications will help you make those long and arduous hours spent in your home studio more enjoyable. When everything sounds great, you'll obviously be more confident about the quality of your music productions. In addition, your critical-listening chops should improve

"Set up your mix position incorrectly, and you'll blow the whole ballgame."
—Chips Davis

when your room isn't throwing aural impediments into your workspace. In short, a good room is a great thing.

And armed with all the know-how that these pro studio designers have encapsulated into the pages of this article, building your ultimate dream studio shouldn't be an *impossible* dream.

Freelance author, guitarist, and gonzo bedroom-studio recordist **Greg Pedersen** *(gregscrnr@aol.com) maintained his esprit while researching and writing this massive feature by inventing open tunings and jamming to old Joe Walsh records.*

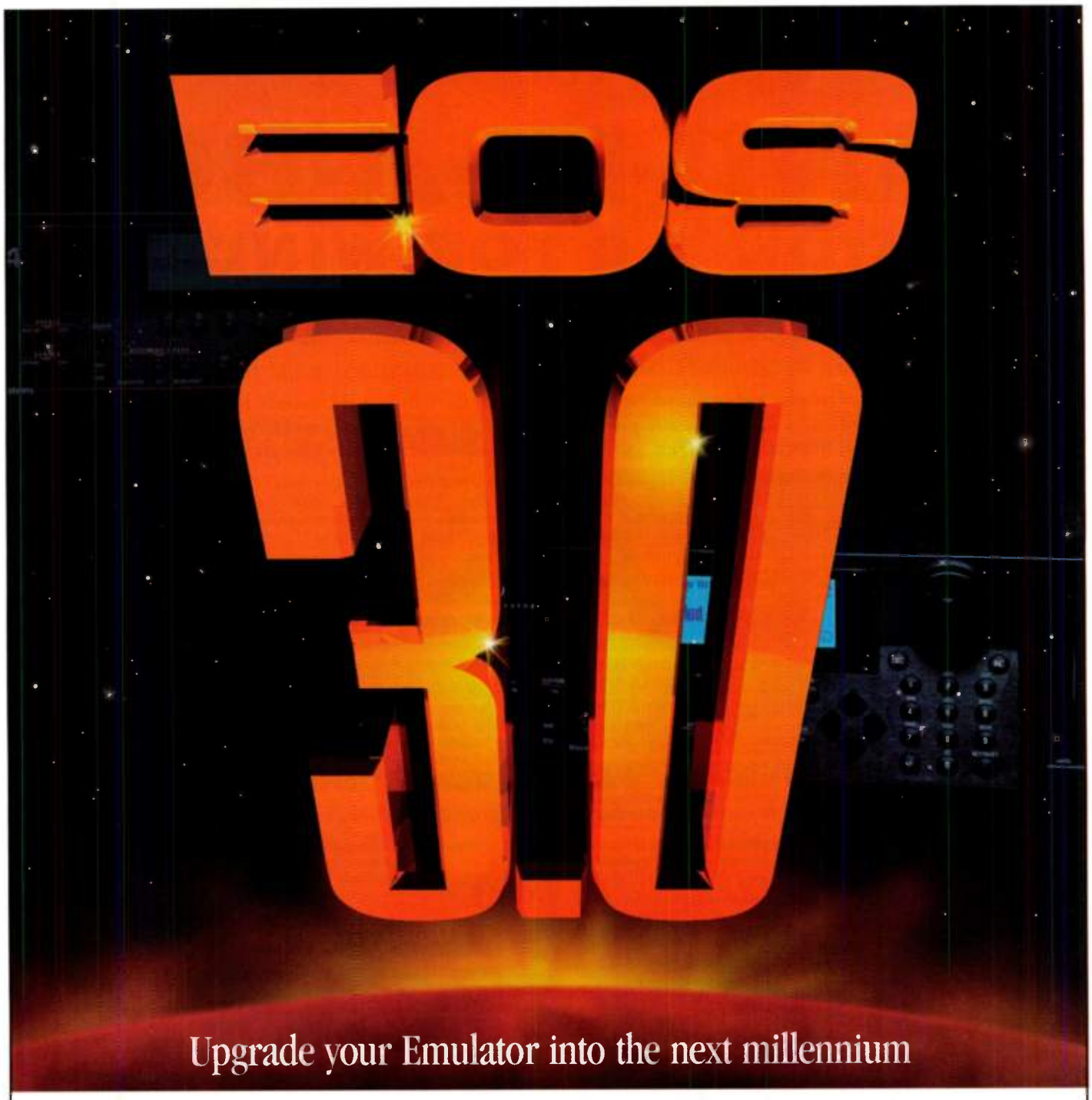

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ILLUSTRATION BY TROY THOMAS

Join us for a safari through the tangle of audio-file formats.

the audio-file jungle

By David M. Rubin

In the heyday of analog tape recorders, capturing audio was easy. All you had to do was slap a tape on the machine, hit Record, and jam. If you wanted to share the recording with someone else, you simply dubbed the tape to cassette and sent it to the lucky recipient.

These days, computers are supposed to make musicians' lives easier, but this isn't always the case. For example, we can now record audio directly into a computer with excellent fidelity and edit the file with unprecedented precision. We can also send these files to anyone on the Internet. However, these digital audio files are stored in many different formats, which are usually supported by some programs and incompatible with others.

If you're planning to produce soundtracks for desktop presentations, collaborate with other electronic musicians, or distribute music over the Internet, you'll soon enter the tangled undergrowth of the audio-file jungle. Even if you're just downloading some cool sound files from your favorite Web site, it helps to know what formats are floating around in cyberspace.

Unfortunately, there are no clear maps through this jungle because some file formats have not been thoroughly explored and others have changed over time. But if you understand which creatures you are likely to encounter, what pitfalls to avoid, and where to find the game trails, watering holes, and tools you need to get through, you can survive and even thrive in this wild environment.

AUDIO-FILE ANATOMY

Macs, PCs, and other computer platforms utilize distinct file formats to store audio data, which some programs can read and others can't. Furthermore, some formats are more widely supported and easier to convert than others. As a result, it's important to know the characteristics and the capabilities of different file formats before you begin a project.

As with most types of files, digital audio files usually include an introductory clump of data called a *header.* The header includes important information that enables the computer to properly read the contents of the file. Common header information for audio files includes such things as the number of audio channels, the resolution (8-bit or 16-bit), the sampling rate, the type of data compression used (if any), and other, more arcane information.

Not all audio files have headers. Some files consist of nothing but raw audio data, without any additional information. To use these files (which most programs can't import), you must determine the proper sampling rate, resolution, and other parameters. A few programs, such as Sonic Foundry's *Sound Forge* for Windows, can open raw files and save them in other formats.

Mac files utilize a unique, 2-part structure consisting of a *data fork* and a *resource fork.* Digitized sounds can be stored either as data in the data fork or as resources in the resource fork of an application, document, or even the operating system itself. (Resources are the building blocks that make a program look and act the way it does, e.g., menus, icons, and dialog and alert boxes.) Not all files have both forks; a file can include just a data or resource fork.

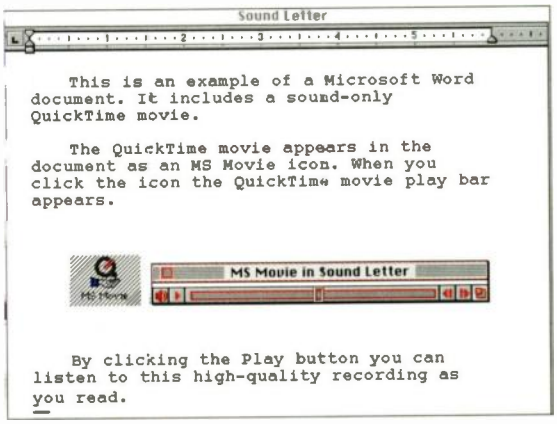


FIG. 1: Sound-only QuickTime Movies are a great way to distribute your music. When a QuickTime file is imported into a document, it appears with its own set of playback controls.

COMPRESSED FEELINGS

Many audio-file formats use one or more compression schemes to reduce the amount of memory or disk space needed to store sound data. In addition to saving space, this allows the files to be transmitted from one computer to another more quickly. However, this advantage comes at a price. Most of the commonly available compression algorithms used in audio files are *lossy;* that is, they lose information in the process of compressing the files. As a result, the sound quality suffers.

The exact nature and severity of audio degradation depends on the type and amount of compression used. In addition, compressing and decompressing files takes up processor time. Finally, some compressed files are not as widely usable as their uncompressed counterparts.

Three of the most common types of compression for computer-based audio are Macintosh Audio Compression/Expansion (MACE), Adaptive Differential Pulse Code Modulation (ADPCM), and µLaw (pronounced "myu-law"). MACE is included in the Macintosh system software. It works with 8-bit digital audio files and supports compression ratios of 3:1 and 6:1. Although compression and decompression are extremely fast, the resulting audio quality is not the greatest. A ratio of 3:1 is marginal for applications where 8-bit music is normally acceptable, and the 6:1 ratio is strictly for speech.

ADPCM is a moderately fast compression scheme for 16-bit sound. It is not as fast as MACE, but it preserves audio quality considerably better, which makes it suitable for a wide variety of applications. However, different algorithms can be used with ADPCM, and Microsoft's ADPCM algorithm is not compatible with the version that is approved by the Interactive Multimedia Association (IMA). You might encounter other variations, as well. With a ratio of 4:1, IMA/ADPCM has become a popular form of compression for multimedia applications, with broad support from software developers.

The µLaw compression scheme is arguably the most common type found on the Internet. It's a fast 2:1 compressor for 16-bit audio that's used mainly with Sun and NeXT computers. This scheme compresses 16-bit files to 8-bit, nonlinear resolution that offers better dynamic range than standard, linear, 8-bit audio files. However, µLaw only supports a sampling rate of 8 kHz. That's great for keeping file sizes manageable, but it only offers the sound quality of a typical telephone receiver.

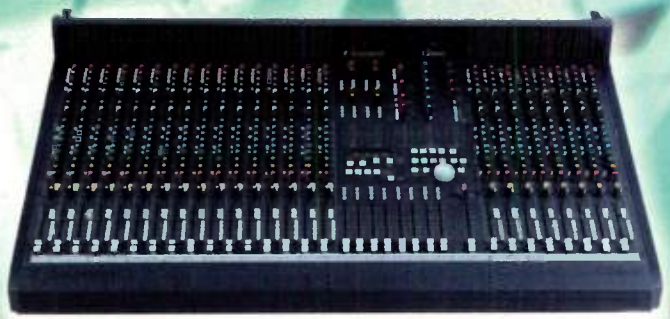
Nonetheless, µLaw enjoys strong cross-platform support with playback software for just about every operating system, including Mac and Windows. Because of the Internet's UNIX/academic legacy, and because files stored on the Net are usually relatively small, µLaw has become widespread despite its poor fidelity.

FIG. 2: Once installed, the *RealAudio Player* pops up from within your Net browser whenever you click on a RealAudio file link. The slider lets you randomly access parts of the sound file. Text fields provide information about the audio clip.

FILE-FORMAT TAXONOMY

Over the years, many file formats have come and gone. Others have continued to garner support because of their unique characteristics or because they fulfill a certain need. Let's examine the popular formats that you're likely

We made the console,



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● "I just sold my Mackie 8-bus and purchased the new Ghost console. The difference is amazing. The Ghost is the warmest sounding board I've ever used." - Mike Perkin (The Lab Recording Studio, Emmaus, Pa)

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● "I love the desk, the EQ is just marvellous. Ghost is the best 8 bus recording desk on the market." - Lee Hamblyn, Engineer.

● "Intuitive handling, flexible routing, great Soundcraft sound." - Melvin Fernandes, Recording Engineer, CMM Studios, India.

● "I use the Ghost for several radio shows doing live performances. The EQ is amazing, I'm on air in 5 minutes! Doing dance stuff is one, doing live stuff is another. But I use only one board for both of them, The Soundcraft Ghost." - Barney Broomer, Sonic One Rotterdam.

● "Ease of operation and the numerous in-line inputs for my synthesizers and samplers is why I purchased the Soundcraft Ghost console." - says President of Saban Entertainment and producer of Mighty Morphin Power Rangers Shuki Levy.

● "I didn't know how useful mute groups could be and how good the EQ had to be until we used the Soundcraft Ghost." - Stefaan Windey, La Linea Musicproductions b.v.b.a., Belgium.

● "It sounds great and the EQ is very precise which makes it very easy to pin-point the frequencies I need to work on. Ghost enables me to finish mixes on the console at home, without having to use any other studio." - Phil Kelsey (Remix Engineer)

● "The console is very user-friendly and is constructed so well that it can easily withstand the rigors of even the most hectic of production schedules." - Corey Dissin, Producer at Paul Turner Productions.

● "Both myself and our Production Director Jeff Thomas used the console for PowerStation and were equally very, very impressed. For the money, the console is fantastically versatile, has good headroom and a very impressive EQ." - Alex Lakey (Engineer for PowerStation)

Ghost

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to encounter at Web sites or when working with multimedia producers and other musicians.

The name of each format is followed by the corresponding file-name extension, when appropriate. Although the formats appear in alphabetical order, they are somewhat interrelated, so some formats are mentioned before they are fully described. However, rest assured that they will be described in short order.

AIFF (.AIF). The Audio Interchange File Format (AIFF) was established in 1988 by Apple Computer with the cooperation of several software developers. It is based on an earlier standard (EA IFF 85) from Electronic Arts. AIFF supports uncompressed mono, stereo, and multichannel audio at many different resolutions and sampling rates, including the CD standard of 16-bit, 44.1 kHz. It's designed to serve as a universal interchange format that allows one program to open a digital recording created by another program.

Because AIFF supports high-quality audio and facilitates cross-platform file swapping, it is an excellent choice for high-end audio applications where storage space is not a major concern. AIFF is supported by pro-level Mac audio software, such as Passport *Alchemy* and Digidesign *Sound Designer II*, as well as by such multimedia applications as Macromedia *Director*. A growing number of PC programs support AIFF, including *Sound Forge*, and the format is also popular on the Silicon Graphics platform.

AIFF only includes uncompressed data, so Apple introduced AIFF-C, a format that supports both compressed and uncompressed audio. AIFF-C can use MACE and IMA/ADPCM compression with ratios as high as 6:1, but the sound quality suffers.

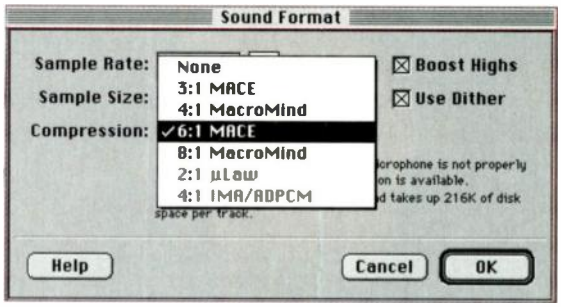

FIG. 3: *SoundEdit 16* 2.0 offers MACE compression and Macromedia's compression algorithm for 8-bit sounds as well as IMA/ADPCM and µLaw compression for 16-bit sounds.

AU (.AU). This file format was developed by Sun Microsystems for its UNIX workstations, but it can also be used on other UNIX platforms. Some PC and Mac audio applications also can deal with it. This format supports stereo and mono files with a resolution of eight or sixteen bits and sample rates from 8 to 48 kHz. It can encode linear (uncompressed) files or use µLaw or ADPCM compression.

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check out the *sox* (sound exchange) application, which can convert just about any audio file format to any other format. However, this command-line program can be difficult to use, especially given its poor documentation.

MOD (.MOD). The module (MOD) format originated on the Amiga and soon spread to other platforms, including the Mac and PC. MOD files are a cross between MIDI and digital audio files. As with MIDI, MOD files store information for playing a sequence of notes. Unlike MIDI, however, MOD files also include a user-defined group of digitized instrument sounds that are used to play the music.

The instrument samples consist of raw, 8-bit, mono audio data (at various sample rates) and can be compressed. As each note in the MOD file plays, the appropriate instrument sample is played at the appropriate rate to provide the proper pitch. MOD files can contain up to 31 instrument voices, although only four are typically played at once. Each instrument can have its own volume, and a few simple effects are possible, including echo and pseudo-reverb (which are achieved by repeating notes), tremolo, and pitch bend.

Because they contain digital audio data, MOD files are much larger than MIDI files. On the other hand, MOD files offer much of the audio quality and playback predictability of a digital audio file without demanding the storage space of a full digital recording.

There are many MOD playback utilities and extensive collections of MOD files on the Internet. In fact, MOD composition has become an art form in itself. Be aware, however, that there are different varieties of MOD files. Most files contain four channels of music, but there's a recent trend toward 8-channel MOD players and music, and compatibility problems may arise.

MPEG. The Moving Picture Expert Group is responsible for the MPEG standard, which was formalized in 1992 for compressing animation, audio, and video files. This is actually a compression scheme that includes its own file format.

Because of this scheme's ability to dramatically compress large files without overly compromising quality, MPEG has become one of the most popular compression standards for use on the Internet. MPEG playback applications are currently available for just about any platform, and numerous sites now distribute music in the form of MPEG audio files.

There are three layers of MPEG encoding that involve increasing levels of complexity; each layer has its own format. The most complex and processor-intensive format, Layer III, takes longer to compress, but it offers higher ratios while retaining much of the original audio fidelity. Layer I is faster, but it offers less compression for comparable quality.

Layer II is a popular compromise for use with audio files. With ratios of 5:1 to 12:1, Layer II files retain much of the original sound file's quality. At a ratio of 8:1, for example, a 16-bit, 44.1 kHz sound exhibits little audible loss.

Higher ratios are possible, but the sound quality begins to degrade noticeably. Layer II files are often identified with the .MP2 extension.

As described here, MPEG refers to the original MPEG standard that is currently in widespread use. There is also an MPEG-2 standard that is not commonly found at the consumer level. It typically requires expensive hardware to use effectively.

QuickTime. Apple's QuickTime is a system extension that's included with the Mac OS software. It provides a means for playing movies with synchronized video and sound on any Macintosh. However, many people don't realize that QuickTime Movies (documents in QuickTime format) need not include pictures. Sound-only Movies are a great way to distribute audio files. Simply import an AIFF, SND, or SoundEdit file into a QuickTime editing program, such as Adobe *Premiere*, and save the file as a QuickTime Movie.

When someone opens the Movie, they see the standard set of QuickTime controls that lets them play the music, pause, access the music randomly, and adjust the volume. You can even import the QuickTime file into a word-processor document so as to combine printed text with music or narration (see **Fig. 1**). In addition to an array of multimedia programs, QuickTime is supported by a number of audio applications, including Macromedia *SoundEdit 16* and *Deck II*, BIAS *Peak*, and Opcode *Audioshop*.

QuickTime Movies are also playable on Windows computers. To make a Movie playable on a PC, use Apple's *Movie Converter* utility that comes with the QuickTime Starter Kit. This lets you convert the Movie into a self-contained file that is still in the QuickTime format but is playable on non-Mac computers. You can then import the Movie into a large number of Windows applications, including word processors and multimedia programs.

Audio Video Interleaved (AVI) is Microsoft's answer to QuickTime. It provides many of the same capabilities as QuickTime, but it is not compatible with Macs. The Microsoft AVI Development Kit includes a QuickTime-to-AVI conversion application.

RealAudio (.RA). RealAudio files use a proprietary format designed specifically for playing audio in real time over the Internet. The RealAudio system was introduced by Progressive Networks in 1995 and consists of a server application, an encoder, and a player. The player, which works within your Web browser, is available from retail outlets or for free by downloading from the RealAudio Web site (www.realaudio.com). However, the current version of RealAudio (v. 3.0) does not support the Mac version of the America Online browser.

With the *RealAudio Server*, which is available from Progressive Networks, Web site developers can stream audio files directly to anyone with a player. Downloading files is unnecessary, and there's a relatively short delay before playback begins.

This audio-on-demand system is made possible by RealAudio's unique compression scheme, which the player decompresses in real time. When you click a RealAudio link on a Web page, the player automatically opens, providing transport-like controls to start, pause, rewind, fast forward, and stop the audio file. Other controls let you randomly access the data and

change the volume (see Fig. 2).

Normally, the *RealAudio Player* delivers 16-bit audio, although an 8-bit option is available. RealAudio uses several different compression algorithms to optimize sound quality for different modem speeds. At the low end, the 14.4 kbps algorithm offers audio quality that resembles a typical mono AM radio. At the high end, the Dual ISDN Stereo algorithm (available with version 3.0) provides near CD-quality audio in real time (with a frequency response up to 16 kHz).

The *RealAudio Encoder* (free from the RealAudio Web site) allows you to create RealAudio (.RA) files from WAV, AU, and PCM files. In addition, *Sound Forge* can save audio files in the RealAudio format, which lets you easily convert from other formats. However, once converted to the RealAudio format, a file cannot be converted back. And without the *RealAudio Server*, you must have the file on your hard disk to play it. In other words, you can't stream RealAudio sound directly from a Web page without the *RealAudio Server*.

Red Book. Red Book audio refers to the audio format used by compact discs. It consists of uncompressed, 16-bit, 44.1 kHz audio data. The name comes from the color of the cover on the specification developed by Sony and Philips. There are other colors, but they're not audio-specific. For example, Yellow Book deals with CD-ROM, Green Book specifies CD-I, and Orange Book defines CD-MO (magneto optical) and CD-WO (write once).

SND Resource (.SND). Macintosh sound (SND) resources are stored in the resource fork of a program or document. The SND format supports mono and stereo files with 8- or 16-bit resolution in a range of sampling rates and compression ratios.

Actually, there are two types of SND resources: Format 1 and Format 2. Format 2 sounds are used by *HyperCard*. Format 1 sounds are used for everything else, including Macintosh system

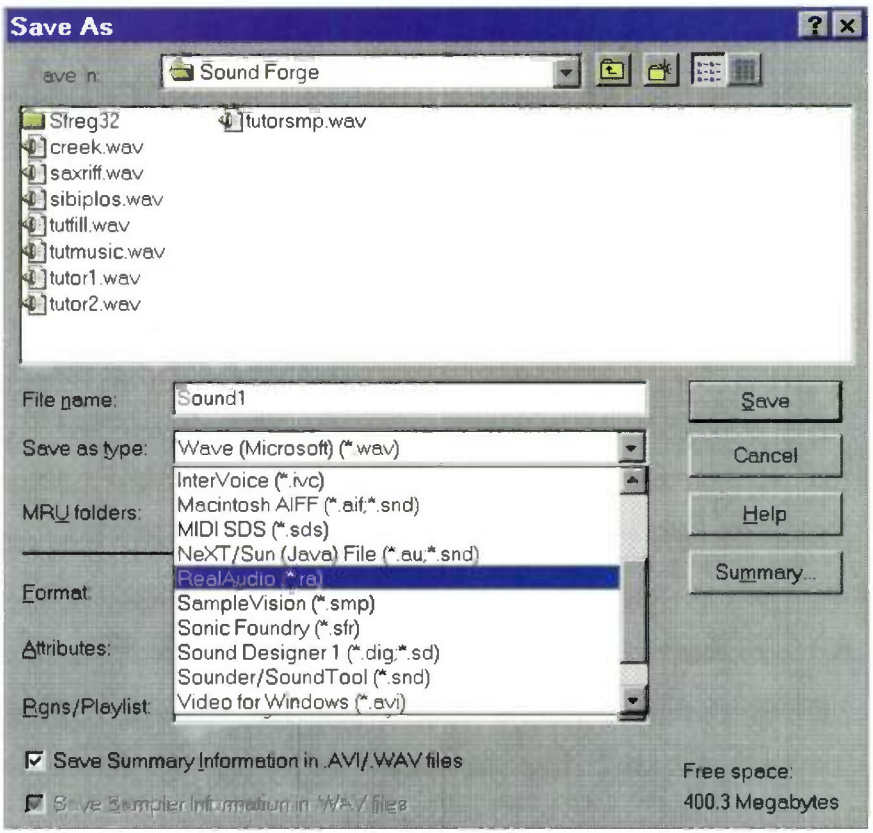

FIG. 4: The Save As dialog box in Sonic Foundry's *Sound Forge* 4.0 includes a drop-down menu with eighteen file formats from different computer platforms.

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sounds. SND resources are common for sounds that play from within a Mac program, such as a multimedia presentation, especially when the sounds or music clips are relatively short and don't require too much storage space.

You should keep in mind that the SND extension is quite ambiguous. It is also used (sometimes incorrectly) with a variety of other formats that are not Macintosh sound resources.

Sound Designer. Digidesign's Sound Designer file format is another proprietary file format that has become a de facto standard on the Macintosh platform. For many years, Digidesign has been virtually alone in marketing high-end digital audio cards for the Mac. *Sound Designer,* Digidesign's original audio-editing application, supports uncompressed, 16-bit, mono sound files at several sampling rates. Although it's now somewhat outdated, you may still encounter this format.

The current version, *Sound Designer II,* is widely used in recording and post-production work because it's included with Digidesign's sound cards and recording systems. The Sound Designer II (SDII) format supports 16-bit stereo files with sampling rates up to

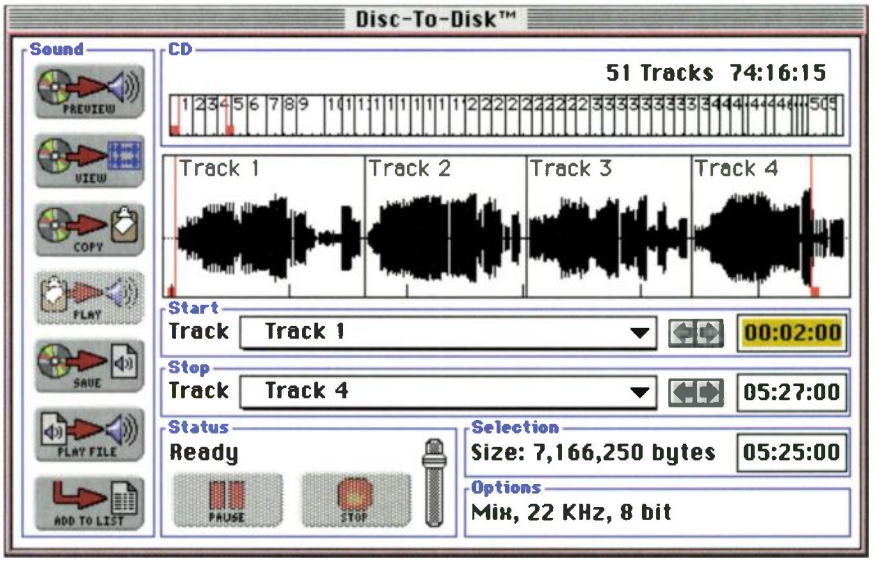

FIG. 5: OMI's *Disc-To-Disk* uses a colorful graphic interface that shows the contents of CD tracks as waveforms. You can select part of a track or several tracks and convert the selection to any of five file formats.

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AIFF and SDII are both popular formats for exchanging high-quality Mac audio files. AIFF is the best choice when you want a generic format with maximum cross-platform compatibility. SDII is a good alternative if the files are to remain on the Macintosh platform or if they might need further editing in *Sound Designer II*. A large number of audio programs support both SDII and AIFF files, including *SoundEdit 16, Peak, Audioshop, Studio Vision Pro,* and *Deck II*.

SoundEdit. In 1987, Farallon introduced the MacRecorder, an inexpensive audio digitizer that could record 8-bit sounds for playback on the Macintosh. The MacRecorder package included a simple but quite capable audio-editing program called *SoundEdit*. In the ensuing years, just about everyone with a Macintosh was creating, swapping, and collecting wacky sound clips to enhance their desktop activities. The SoundEdit format soon became a standard for exchanging 8-bit mono or stereo sound files at four sampling rates from 5.5 to 22.255 kHz.

Compression is available for mono files at 3:1, 4:1, 6:1, and 8:1 ratios (see **Fig. 3**). The actual audio data resides in the file's data fork; the resource fork contains information describing various parameters of the sound and the way it's displayed onscreen, including loop and selection locations, labels, and pitch settings.

Over the years, *SoundEdit* (now owned by Macromedia) has undergone several upgrades, including support for 16-bit sounds with sampling rates up to 64 kHz. The newer SoundEdit 16 format extends the original SoundEdit format with additional information in the resource fork concerning tracks, cue points, spectral data, printer preferences, and Mac System 7 information. The SoundEdit 16 format is supported by a large number of Macintosh audio and multimedia programs, including Macromedia *Director* and Adobe *Premiere*.

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VOC (.VOC). The Voice (VOC) format was designed by Creative Labs as the native digital audio format for their Sound Blaster audio cards and became a popular proprietary file format for the PC. Originally, VOC files were always 8-bit, but with the introduction of the Sound Blaster 16 card, the format was extended to accommodate 16-bit stereo sound.

The VOC format supports compressed or uncompressed data at a wide range of sample rates up to 44.1 kHz. Compression options include 4:1, 3:1, and 2:1 ratios for 8-bit sounds and 4:1 and 2:1 ratios for 16-bit sounds.

VOC files can include a number of special features, such as markers for looping, synchronization markers for working with multimedia programs, and silence markers that replace sections of silence to reduce storage requirements. Despite its association with the Sound Blaster family, the VOC format is not as widely used as the WAV format, although it's supported by a number of editing programs, audio utilities, and multimedia applications.

WAV (.WAV). The Waveform Audio (WAV) format was introduced with Windows 3.1 as the standard format for multimedia sound applications. WAV is the most common of several file types that conform to the Resource Interchange File Format (RIFF) specification adopted jointly by Microsoft and IBM. As a result, you might see it referred to as the RIFF WAV format.

Because of its association with Windows and its "interchange" orientation, WAV has become one of the most popular formats for exchanging digital audio files on the PC. In fact, although the headers are a bit different, WAV and AIFF are similar in the type of sound data they support. WAV files can include mono or multichannel audio at 8- or 16-bit resolution with a variety of sample rates, including 44.1 kHz. The format supports different compression schemes, but the most common is IMA/ADPCM. A 4:1 ratio is often used for 16-bit sounds.

WAV files are universally supported

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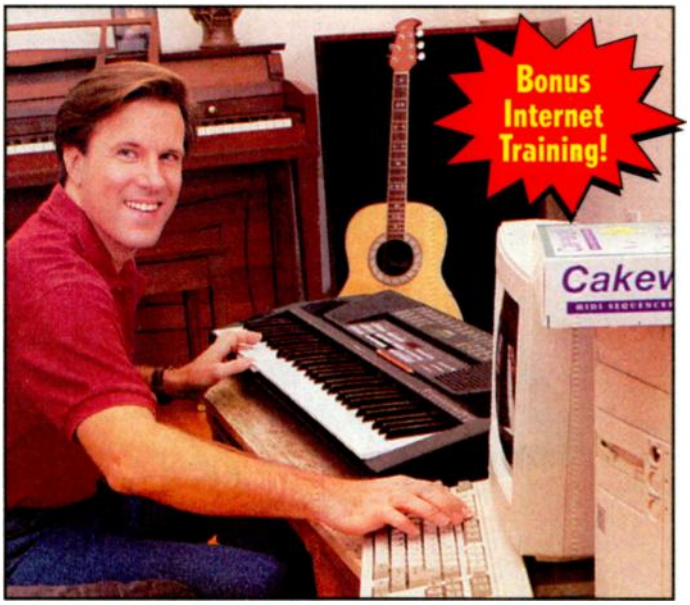

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by Windows audio-editing programs, such as *Sound Forge* and Steinberg's *WaveLab*, as well as multimedia applications and the Windows *Sound Rezcorder*. There is less support for this format on the Mac, although a growing number of audio programs, such as *SoundEdit 16* and *Peak*, can import WAV files.

FOLLOWING GAME TRAILS

In the past, each audio-file format existed solely in its own little universe. Mac users had their favorite formats and PC people had theirs. Even on the same platform, files of different formats were often difficult (if not impossible) to exchange.

Fortunately, things have changed for the better. In today's cross-platform world, it's easier than ever to convert one file format into another. Simply open a file in one format and save it in a different one. In many cases, you can change the resolution and sampling rate of a file as it's being converted.

PC users who work with multiple file formats should check out *Sound Forge* 4.0, which offers a veritable Rosetta stone for audio formats. *Sound Forge* can open and save an impressive list of eighteen formats from Macintosh, PC, and other platforms. You can even save in any format while changing the stereo/mono, bit-resolution, and sampling-rate characteristics (see **Fig. 4**).

SoundEdit 16 version 2 offers similar options for Mac users. It supports ten file formats, including QuickTime. One valuable feature offered in version 2 is the ability to import a Red Book audio track directly from an audio CD and open it in a *SoundEdit* window. From there, you can edit and save it in another format. Opcode *Audioshop* (Mac) also imports audio CD tracks along with seven other Mac formats.

There are several programs that specialize in transferring Red Book audio tracks to your hard drive. *Disc-to-Disk* from Optical Media International (tel. 800/347-2664 or 408/376-3511; fax 408/376-3519; e-mail sales@optmedia.com; Web www.optmedia.com) offers an intuitive graphic interface for selecting parts of tracks to transfer (see **Fig. 5**). The program, which is available for Mac and Windows, lets you change the sample rate and resolution, and it supports MACE compression and five formats: SND, AIFF, QuickTime, SDII, and WAV.

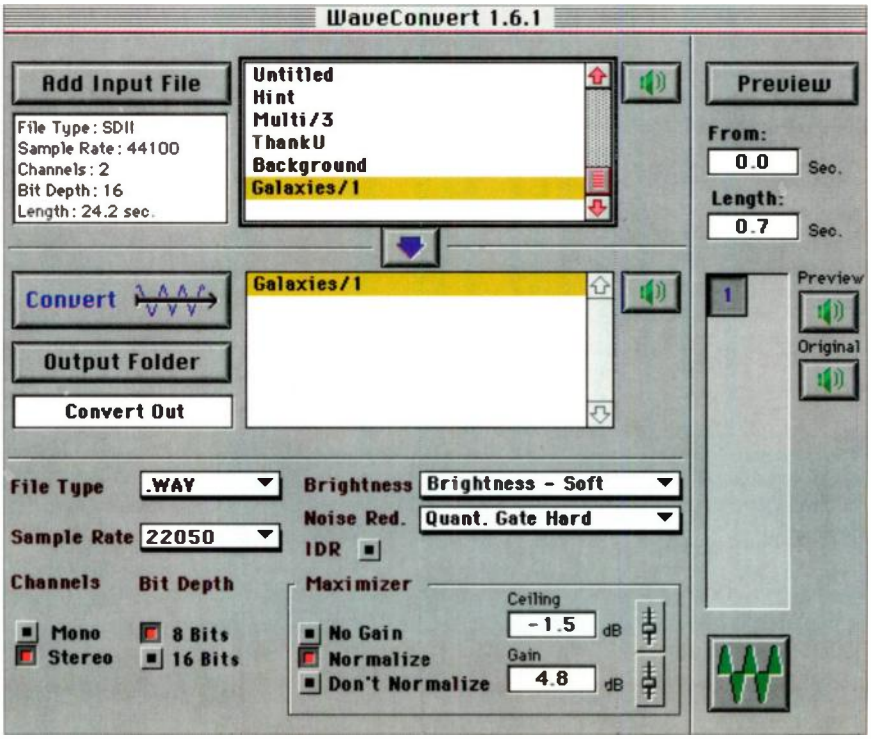


FIG. 6: *WaveConvert* offers several audio-processing options to improve the results of downsampling and 16-to-8-bit reduction. Sophisticated algorithms maximize sound quality during conversions between file formats.

If you're concerned about maximizing audio quality as you convert from one format to another, consider *WaveConvert* from Waves (tel. 423/689-5395; fax 423/688-4260; e-mail waves@waves.com; Web www.waves.com). The Mac version features sample rate, resolution, and stereo/mono conversion, and it supports AIFF, SND, SDII, QuickTime, raw audio, and WAV formats. The Windows version offers the same features, but it only supports the AIFF and WAV formats. When you buy the program, you get both the Mac and Windows versions.

WaveConvert offers an array of audio-processing options to improve the results of downsampling (reducing the sample rate) and file conversion (see **Fig. 6**). The program's conversion functions are based on proprietary algorithms that optimize dynamic range, perform peak limiting, add gain, and reduce noise while providing subjective control over sound quality.

Another audio-file conversion utility is Synclavier Company's *S/Link* for the Mac (reviewed in the December 1995 **EM**), which is distributed by Ilio Entertainments (tel. 818/883-4546; fax 818/883-4361; e-mail ilioinfo@ilio.com; Web www.ilio.com). It supports AIFF, AU, Mac System 7 (which is similar to SND), QuickTime, raw audio, Sound Designer and SDII, SND, VOC, WAV, and more. The program can also convert sample rates and resolutions, and it can read (but not write) MOD files.

NATURAL SELECTION

The audio-file jungle is teeming with exotic species, and the formats described here are not the only ones you'll find. They're simply the ones that you are most likely to encounter while surfing the Net or working with other musicians. Natural selection has made dozens of older formats obsolete as more popular formats, such as AIFF and WAV, gain prominence.

Meanwhile, an increasing number of applications are supporting more formats, and the Internet is filled with shareware programs for working with and converting audio files. For now, diversity is still the name of the game in the audio-file jungle.

David M. Rubin *owns a computer-music studio in the Los Angeles area, where he composes for film, video, and multimedia. His latest book,* The Desktop Musician, *is published by Osborne/McGraw-Hill. You can reach him at dmrmusic@earthlink.net.*

DANNY'S

Danny Elfman embraces the

Big

dark rites of film composition.

adven

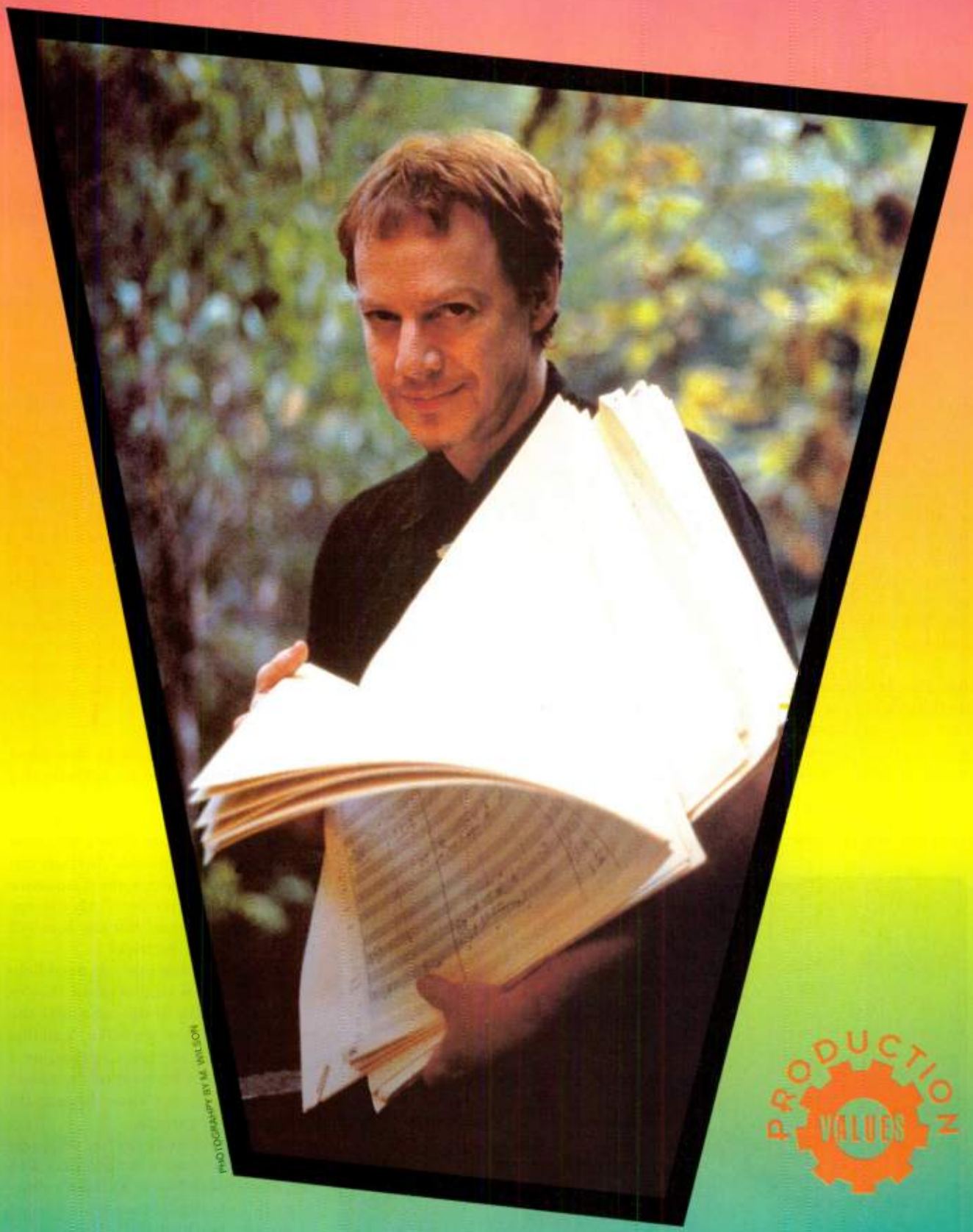
By Greg Pedersen

THE ODDS ARE SLIM that even a tremendously talented musician will succeed in the brutally competitive film-scoring industry, but in 1985, Danny Elfman beat the odds. That year, after a lengthy stint as the leader of modern rock's commercially underappreciated wild bunch, Oingo Boingo, Elfman jumped ship to score Tim Burton's directorial debut, *Pee Wee's Big Adventure*.

That first stab at film composition was a smash success, and it was only the first in a series of scoring triumphs that include *Batman*, *Beetlejuice*, *Mission Impossible*, and *Edward Scissorhands*. Elfman has conquered the small screen, too, creating memorable themes for TV shows such as *Tales From The Crypt* and *The Simpsons*.

In fact, Elfman has been so prolific that he recently released his *second* compilation of film scores, *Music for a Darkened Theater*, vol. 2. The 2-disc set includes music from the composer's last ten films (including *Sommersby*, *Black Beauty*, *Dolores Claiborne*, and *The Nightmare before Christmas*); a score for *Freeway*, a low-budget film that he recently did for his best friend from high school ("I got paid a dollar," he says); and some unreleased material from television series such as *Amazing Stories* and *Pee Wee's Playhouse*.

As usual, Elfman is currently deluged with scoring projects. But as he braved a hellish deadline to complete the soundtrack for Burton's upcoming *Mars Attacks!*, he graciously took a break to share some insights with **EM** readers.



PHOTOGRAPHY BY M. WILSON

PRODUCTION VALUES

tres

DANNY'S Big adventures

What's it like being a film composer after spending so many years as a rock musician?

Well, being a film composer requires a level of discipline that far exceeds anything I had to deal with as a musician making records. As a composer, you have to write every day. Period. You have to write approximately two minutes of music every single day whether you feel great or you feel like crap. And you have to be *inspired* every day.

And then there's always a certain point—as the clock is ticking toward the deadline when the score must be finished—where I feel like I'm stepping off a cliff. But I must have faith that when I take that step, something will appear at the very last second to catch me. That's what diving into a score often feels like.

Are the deadlines really as tough as they're reputed to be?

I try to get six weeks to write a score, with maybe a couple of extra weeks added so I can toy around with ideas. Sometimes you get that much time, and sometimes you don't. Part of being a successful film composer is being able to work extremely quickly and effectively under pressure.

For example, in 1996, I scored four major films, so that was a particularly tight year for me. The workload on those movies—as with any film—depended upon how much the director wanted to get involved. Brian DePalma, who directed *Mission Impossible,* was very particular and wanted to hear every cue I wrote before we got to the scoring session [for the orchestra]. Gus Van Sant, on the other hand, did not need to hear every cue I wrote for *To Die For.* Then, of course, there are scheduling complications, such as when I was brought into *Mission Impossible* really late and was given just five weeks to complete the entire score.

How do you begin working on the scoring process?

I experiment with different ideas to get the score's thematic material laid out, and then I play them for the director. If he or she likes the direction the music is going, then sooner or later I'll have to start writing the score, or I'm in deep trouble.

Once I'm actually pumping out music, Steve Bartek gets dragged in. Steve does the final orchestrations after I have sketched out the music as elaborately as I can. He was the guitarist in Oingo Boingo and has been my orchestrator since day one. When I got the *Pee Wee's Big Adventure* gig, I called him up and asked, "Steve, didn't you take an orchestration class at UCLA once?" He answered, "Yeah," so I said, "Well, do you want to work on a film?" It was pretty funny, because Steve, Tim Burton, and I were all babes in the woods. It was the director's first film, the composer's first film score, and the orchestrator's first orchestration.

Do you still have to be able to write music, or does notation software render that skill superfluous?

When I started scoring in 1985, you *had* to write music. There was no other way to communicate the parts to the musicians. These days, theoretically, if you perform and sequence the score close enough to the way you want it, you can print it out with some type of notation software.

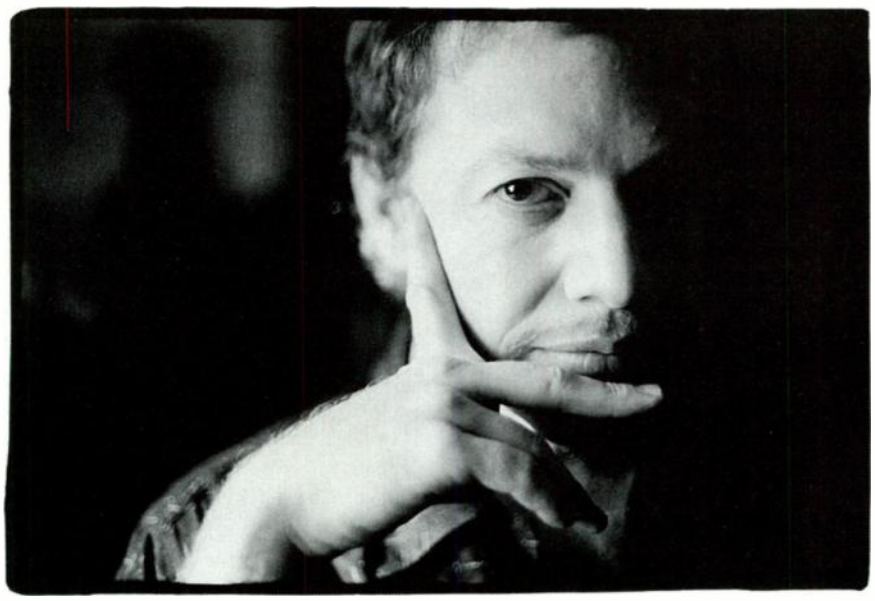
However, the first time I tried notating a piece using MIDI, it didn't come out right. I realized that, when you're looking at a piece of music bar by bar, you learn some things about the composition that you'd miss if you just hit a keystroke and printed it out. You add certain things and you think a slightly different way when you're working on paper. Although I have notated some scores directly through MIDI, I'm really glad I taught myself to write, because it is an important part of learning how to compose.

What would you say is the most critical element of an artistically successful score?

The ability to find the tone of a film can be the most important thing a film composer can bring to the job. Anybody can hold long chords with some dissonance and say, "That's tension." My eleven-year-old can do that! But can you nail the tone? That's the trick.

You see, sometimes people need help figuring out what kind of movie they're watching, and the music can nudge the viewers one way or another. Visually, viewers may be interpreting a scene a certain way, but because of the music they're hearing, they know they should be smirking or gasping.

For example, on *To Die For,* there was a dilemma because test audiences had trouble reading Nicole Kidman's character [who entices some wayward teens to murder her husband]. Her character was strange because it was wicked but not evil in the way that people are used to seeing a wicked character. As a result, audiences were very unclear about whether they were allowed to



Elfman has successfully jumped from the concert stage to the scoring stage.

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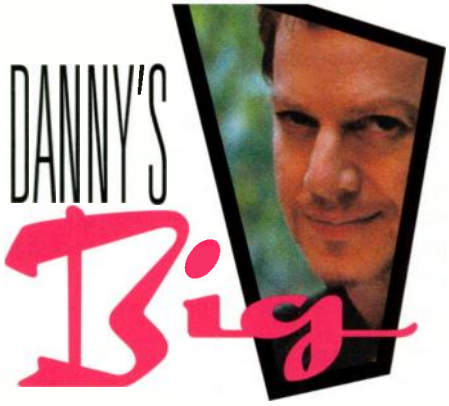


DANNY'S Big adventures

laugh or not because the film's plot revolved around murder. Obviously, we didn't want to wash over the dark side of what was happening in the movie, but we also wanted to keep the audience aware that they were watching a strange, quirky *comedy*. So I came up with a theme for the Nicole Kidman character that had a lightness to it.

On that point, do film composers rely on specific instruments to produce a sense of happiness or dread?

A talented composer can create a light or a dark tone with *any* instrument. For example, strings typically have been used for romantic scores, and yet the person who most influenced my style, [composer] Bernard Herrmann, used strings for the most chilling score I know: *Psycho*. He also scored one of the scariest episodes of the original *Twilight Zone* television series using a harp, a glockenspiel, and a bass clarinet. These three very diverse instruments would not seem conducive to a scary score, but Herrmann used the juxtaposition of two light instruments against one dark instrument to produce something that was absolutely terrifying.

GENE TRINDL

Director Tim Burton (left) and Elfman in the studio during the recording of the score for *The Nightmare before Christmas*.

All the films that Herrmann scored had strong, identifiable patterns that he could twist and turn a million ways. His work taught me that there are no rules; there's no scary instrument or happy instrument.

What types of things can lead you to uncover the right tone for a scene?

Well, for *Beetlejuice*, the craziness of the character said to me: "Russian dancing, kicking out the legs, wild and drunken fun!" But for *Edward Scissorhands*, the film's fairy tale–like quality moved me to keep a sad undercurrent running throughout the music.

Batman, however, was inspired by the set. When they were about halfway done with filming, I was able to walk around the set at night and get the whole vibe of the movie. As far as the characterizations went, the Joker had a schizophrenic feel, so I didn't give him a specific theme. One moment, his music would be comic, the next moment it would sound kind of music box–like, and later there would be some dissonance.

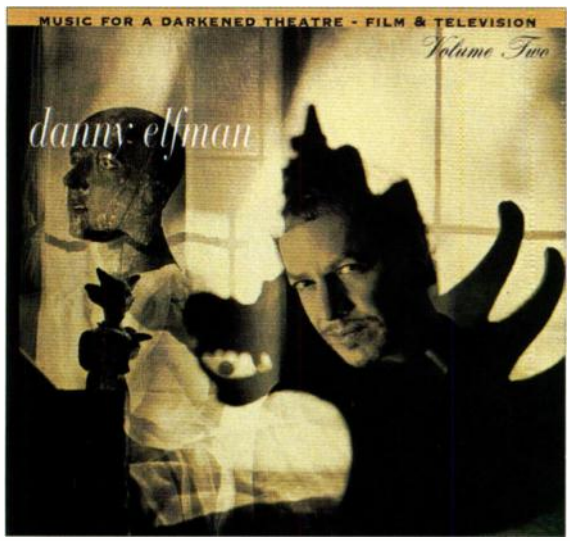
In *Batman Returns*, the characters were more consistent, so we felt three of the four characters should have very identifiable themes. Consequently, the soundtrack turned out to be more of an old-style, thematic score that switched back and forth between characters.

Catwoman was simple. I went right for a 1960s, slinky, bending style of strings to convey a TV-show sound. I think Julie Newmar [who played Catwoman in the original TV series] was my first crush! Both the Catwoman and Batman themes are in a minor key, but hers is whimsical whereas his has a tragic, quasi-operatic quality. Even though the Penguin was evil, I wanted his theme to be evangelical, almost religious, with a bit of a hymn quality. I definitely wanted something that could easily turn comic or tragic.

Moving to the small screen, I very much wanted the tone of *The Simpsons* to reflect a 1960s family TV show. I wanted it to feel like it came out of another era, so I had some fun with that retro sound. I used xylophones, muted trumpets, and pizzicato strings.

When you actually get down to composing a music cue to picture, do you usually follow any particular methodology?

The system I developed during *Pee Wee's Big Adventure* is the one I use to this very day: I watch a scene until I start hearing a piece of music in my head. Then I'll time the scene out by locking my sequencer to the time code on the work video.



Elfman's prolific output as a film composer has necessitated a second compilation CD of his scores, *Music for a Darkened Theatre*, vol. 2.

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DANNY'S Big adventures

Let's say I'm hearing something that I feel is roughly 126 beats per minute [bpm]. I'll input the SMPTE number for the beginning of the scene into Mark of the Unicorn's *Performer* and run a 126 bpm click track locked to the video. Now I can experiment with tempos until I find what I like. Sometimes I'll use one tempo throughout a scene, and sometimes I'll use half a dozen, so it's also very important that I block out a tempo map for the entire scene.

At this point, I usually select a piano sound and sketch out the gist of what I want to happen against the scene musically. Much of this phase is improvised, but I'll typically end up with a rough sketch of what I want the music to do. For example, let's say there's a 4-bar melody that I really want to end on a door slamming, but it's a beat off. With *Performer* locked to the work video, I can move the phrase one beat and adjust things until I'm satisfied. Ultimately, I'll have a complete map of the scene, whether it's 20 measures or 220 measures. I've got all my hit points, meter changes, and tempo changes marked.

Before I start notating the score on paper, I'll write out that entire map on the top staff—even if it's twenty pages long—with all my bars and all my meter changes. Then I indicate everything I want to hit with a system of little arrows. If the hit is on a downbeat it gets an arrow, and if it's on an upbeat it gets a crooked arrow. Now, I can just look at the entire music cue at the top of the page and see all my bar lines *and* everything I want to hit. This makes it easier when I begin writing the music down because I can make sure everything is metered correctly and, as I see a hit point coming up, make sure the music works up to that point. I'm actually writing the music and following the map on the top staff simultaneously.

Many of your scores are not purely orchestral, however, so how do you incorporate synths and samplers into the orchestra cues?

Up until last year, I used to pre-record all of my synth and percussion stuff on a Sony/MCI 2-inch, 24-track in my home studio—which I call "MIDI Hell"—and bring the tape to the scoring session. This tape, which is a recording of all the sequenced, nonorchestral instruments I want in the score, is then slaved to another 24-track machine that is used to record the orchestra.

The problem with this prelay system is that I have to lay the click tracks down on tape to ensure that the orchestra stays in sync with my pre-recorded tracks. Obviously, if we discovered any problems during the scoring session—or if we simply wanted to make some changes—we were stuck, because we couldn't adjust anything. The tracks were already on tape.

On *Mission Impossible*, it was pretty obvious that I needed more flexibility. So, instead of pre-recording my tracks on tape, I sequenced everything the way I normally would and brought along the synths, samplers, and sequencer to the session. Now *Performer* could receive time code from the video, and both my sequences and the orchestra could sync to a master click track created by the music editor. The tracking method is basically like this: the music editor builds a click using this tempo-processing software called *Auricle*, from Auricle Control Systems (for more information

MISSION POSSIBLE

"I just finished scoring my 26th film," says soundtrack wizard Danny Elfman, "and you'd think it would get easier, but I'm no less insane when I'm composing. And it's still just as hard to play music for the director for the first time!" However, for **EM** readers itching to enter the celluloid madhouse, Elfman offered some tips for surviving film-scoring gigs with a few shreds of your sanity (and stomach lining) intact.

Work those themes. When I'm starting a film, I make sure I compose a primary theme and at least two secondary ones that can be turned a number of different ways. I'll take the theme and figure out whether I can play half of it and still recognize it. Then, does it work in a major and a minor key? Can I turn it from funny to spooky? Can I cut it down to just three notes and still make it recognizable? These are some of the acid tests I put a theme through while I'm composing.

Be empathetic. The problem between every director and composer is that music is not easy to describe in words. You can spend hours talking about the score, but from the first moment the music is played, it either works or it doesn't. Then you realize all the talking you've done doesn't mean a thing! I try to approach everything from the standpoint of a director's eyes so I can compose something that's satisfying for me and yet still connects with the director.

Define the demo. Make sure the director understands that, in the demo stages of an orchestral score, the synth brass are going to sound like car horns and the synth strings are going to have a peculiar edginess. I usually play the director a demo of another score, such as *Batman*, and say "Here's the demo score I played for Tim [Burton], and here's the real thing. As you can hear, the real orchestra is going to sound ten times better than the synth orchestra."

Argue early. Show the director some themes and sketches as soon as you can and engage their opinions. If there are any problems, the goal is to duke it out in the studio during the writing process. You don't want conflict on the scoring stage when there are 90 musicians sitting there! Ideally, when you get to the orchestra session, all the director should be hearing is a better version of what he or she heard in your home studio.

Let the film conduct. Most film editors have an internal metronome going, so if you can tap into that rhythm, it will make scoring a lot easier and more fun. In fact, if you're out of sync with the film editor, it will be very hard to make the picture work. Once I lock into the editor's rhythm, though, it's easy to make all my hits match perfectly to the action onscreen.

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on how *Auricle* is used in scoring sessions, see "An Orchestrated Escape" in the October 1996 **EM**). *Performer* is slaving to that click, and my sequences and the orchestra are playing together live and being recorded onto a master multitrack reel.

The flexibility of this system is great. For example, if we want to go a little quicker in, say, bar 57 of the score, and notch up the tempo three bpms, we can lose two beats at the end to make up the difference. My sequences will simply follow whatever changes are made. Or if I want to change the tempo for the last eight bars and do an accelerando or a retard, I can simply edit the sequence on the spot.

With pre-recorded cues on tape, you can't do that because tape doesn't like to suddenly start changing speeds. I can also accommodate the director if he or she says, "I know that I told you come out here, but can you keep the music going for two more bars?" Once again, I just edit the sequence.

Obviously you use quite an array of gear to presequence your scores. How much equipment do you actually bring to the scoring session?
Typically, I'll bring a rig with an Apple Power Macintosh 9500 running Mark of the Unicorn's *Performer*, two Emulator EIV samplers with 128 MB of RAM, and two Roland S-760 samplers. Sometimes, I'll also bring my other Emulator samplers: two 3XPs and two ESI-32s. The S-760s are playing guide stuff—the orchestral string and horn mockups that I've sequenced at home—that I can refer to, but these tracks seldom end up as a part of the final score. The EIVs and the 3XPs are the machines actually laying down what will end up on the final score as the drums, percussion, and so on.

Stylistically speaking, you've definitely broken convention and forged a unique style.
I guess I *must* be breaking some rules, judging by the incredibly negative reactions I get from other composers and how frequently I hear myself imitated in other composers' scores!

"The ability to find the tone of a film is the most important thing a film composer can bring to the job."

When did you start noticing composers imitating your approach?
Right after *Pee Wee's Big Adventure*. For that film, I took an Italian, Nino Rota–like approach to an American comedy because I didn't like the fact that most contemporary comedy scores were so jazzy and poppy. So, of course, for about the next five years after *Pee Wee's Big Adventure*, all these comedies were suddenly becoming Italian!

And the imitations didn't stop after *Pee Wee*, either. My other scores that are imitated the most are probably *Beetlejuice*, *Edward Scissorhands*, and *Batman*.

Unfortunately, my generation of composers are by and large the most plagiaristic in the history of film. And, you know, composers have a wonderful way of rationalizing when they plagiarize another living composer.

For example, John Williams' 2-note pattern for *Jaws* has been imitated thousands of times in tense scenes, but composers will simply rationalize that he didn't *invent* those two notes. Well, that doesn't matter! It was Williams who brought those two notes to a certain style of movie, and that's really the key issue.

The bottom line, I believe, is that it's just too easy to imitate your musical contemporaries. It's like taking a free ride. I think that composers who really care about the *art* of film composition will tend to draw more from the work of dead masters than from their living cohorts. ❀

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JUST SAMPLE IT

DESKTOP MUSICIAN

Getting Away from It All

Reduce ambient noise by stashing your CPU and drives in the closet.

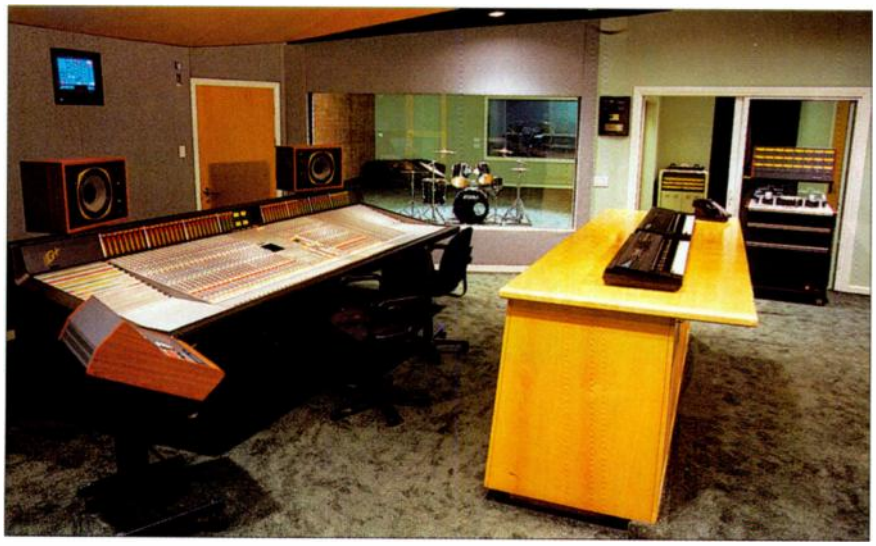
By Paul D. Lehrman

Sometimes you just have to get away from your computer. No, not because your spouse tells you it's time to stop working, or the kids are clamoring to go to the beach, or you've been staring at the monitor for so long that your glasses are beginning to develop screen burn-in. The problem is that sometimes your computer just makes too darn much noise. So you have to get away from it or, even better, move it away from you.

Computers have fans, hard-disk drives, removable-media drives, and various other peripherals that produce noise, both constant and irregular. For the majority of computer users—business people, scientists, Web surfers, or even graphic designers—this noise is relatively innocuous, a minor annoyance that is far outweighed by the convenience of having the on/off switch, reset button, floppy drive, CD-ROM drive, and other media close at hand.

But to a musician, a high ambient sound level in a music studio is a major impediment, much like a fluorescent paisley pattern on the walls of a room where an artist is trying to paint a masterpiece. No matter whether you're recording tracks, designing sounds, or mixing, a high noise level is going to interfere seriously with your ability to hear, which will cause your work to suffer.

Unlike mainframe computers or pro video decks, which are usually in a machine room separate from their operator's primary work area, most personal computers are designed to be on your desktop. Remote operation rarely figures into the thinking processes of computer designers. The cables that generally come with personal computers—monitor, SCSI, and keyboard—are short, and if you ask a computer retailer for a longer cable, you'll probably be met with a blank stare.

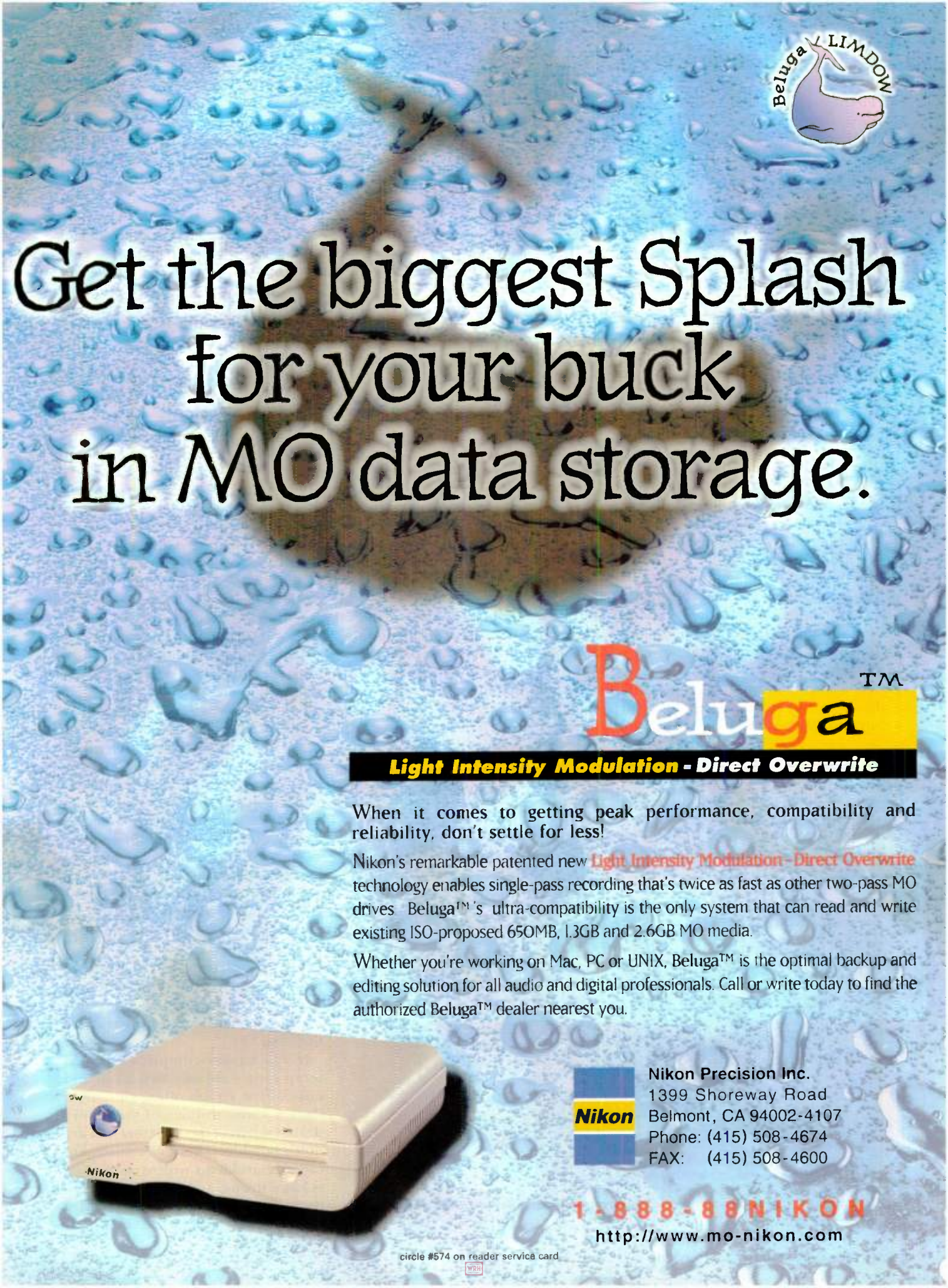


ED COLVER

Pro studios, such as the L.A. Recording Workshop in North Hollywood, employ a machine room (visible behind the keyboard station) to prevent noise from invading the control room. Desktop recordists can modify a closet to silence their disk drives. (Courtesy studio bau:ton)

STASHING THE CPU

Fortunately, with some ingenuity, some of these limitations can be overcome.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First things first: let's move the computer out of the studio, and then we'll figure out how to hook it up. If you have another room adjacent to your control room, think about stashing the computer over there. Put the computer close to the common wall (although not necessarily *on* the wall, because that might allow vibration to come through), and then punch a hole in the wall big enough for all of the cables to go through.

A more convenient solution is to use an existing closet in the control room or build one. The most important consideration when stashing a computer in a closet is to make sure it gets plenty of fresh air. So, instead of simply shutting the equipment up, cut away about six or eight inches at the very bottom and top of the door to create a kind of chimney. Cover the gaps with foam from an air-conditioner filter to keep the noise level down and to keep dust away from the machinery. Build a shelf or use a small table to get the computer off the floor, but make sure it's not so big that it cuts off the vertical air circulation. Put the computer and other hardware on some more foam to cut down on vibration transmission.

You may even want to put a small, low-noise fan in the closet to draw air upward. Mount the fan on a wall (not the door) above the computer. To access removable media, you can simply open the closet door. Alternatively, you can build a small door directly in front of the drive bays. Use Velcro fasteners or a wooden hasp to hold the door closed; a metal latch might vibrate.

REMOTE CONTROL

Now that the computer system is safely out of earshot, how do we control it? We need cables for the keyboard and mouse, monitors, and serial devices.

Serial cables for a Mac or PC are a no-brainer. They can be pretty much as long as you like, assuming you start with quality cables. If you need to change cables a lot—say, between a MIDI interface and a modem or printer—and the computer's new location has made this difficult, bring a cable from the serial port out to an external serial switch box mounted in a convenient place, and connect your peripherals to the switcher.

Keyboard cables for a PC are similarly easy to extend, but on the Mac things are a little more complicated. The Mac's Apple Desktop Bus (ADB), which connects the keyboard, the mouse, a trackball or touch pad, some modems, and the occasional hardware dongle to the CPU, uses a special, round connector with four pins and a rectangular "key" to make sure it is oriented correctly. Although ADB runs can be quite long, dedicated cables longer than six feet are hard to find, and they tend to be expensive.

If you go to a video supply store, however, you will find perfectly good ADB cables of many lengths at quite reasonable prices. That's because ADB connectors are identical to connectors used for S-VHS video. Your local Blockbuster or Radio Shack won't have them, but a dealer or mail-order supplier who caters to video professionals should have S-VHS cables in abundance. You should be able to find it in lengths of anywhere from five to 50 feet, perhaps even longer. Go for quality, and don't worry about the cost. For example, Markertek Video Supply lists heavy-duty, 15-foot cables in its catalog for a mere $21.95.

FUZZY SCSI

Unfortunately, there's an inherent limitation on cable length for one of the most important protocols of the personal computer world: SCSI (Small Computer Systems Interface). The SCSI specification limits the total length of a SCSI chain—that is, all of the cables connecting all of the devices in the chain, including any internal drives—to six meters, or about 19.5 feet. That's not very far.

If all of your SCSI drives are crammed into the closet with your computer, cable length is not an issue (assuming it is within the aforementioned limit). But you should use the shortest cables you can get away with, not just to keep down the clutter but also to keep the weight and odd angle forces caused by longer cables from straining the connectors. If you have devices stacked on top of each other, you may be able to use rigid C connectors, which literally clamp in place behind two SCSI devices and form a solid connection while minimizing the signal-path length.

Sometimes, however, you need to run SCSI cables to devices that are located some distance from each other. For example, you might have a sampler that you want to use with drives that are attached to your computer, or maybe you'd like to exchange samples between the sampler and the computer via SCSI. If the cable is going to be more than six feet long, get the best cable you can find. Make sure the cable is at least double-shielded with both copper braid and aluminum foil protecting the conductors. A cheaper cable may seem to work, but SCSI problems often come and go intermittently, and a cable that is marginal may be okay one day and throw a tantrum the next.

If you have a complicated SCSI setup or if you move devices in and out of your system often, you might want to invest in some of the sophisticated analyzer/terminators that are now available. These can automatically fix problems with termination, voltage drops, and impedance conflicts. Another cool device is a hot swapper, which lets you switch SCSI

Gefen Systems' TSE100 allows you to place a Macintosh monitor, keyboard, and mouse up to 500 feet from the CPU.

Sound Forge Heats Up With ActiveMovie

Sound Forge 4.0, the award-winning digital sound editor for Windows, now supports ActiveMovie audio plug-ins. Sonic Foundry has selected ActiveMovie (a component of Microsoft's Interactive Media technology) as the foundation for the Sound Forge plug-in architecture. ActiveMovie plug-ins will be supported by a variety of audio software companies including Sonic Foundry and Waves.

Sound Forge also supports the ActiveX Streaming Format (ASF) – the new audio and video Internet and intranet streaming standard used by Microsoft's NetShow On-Demand.

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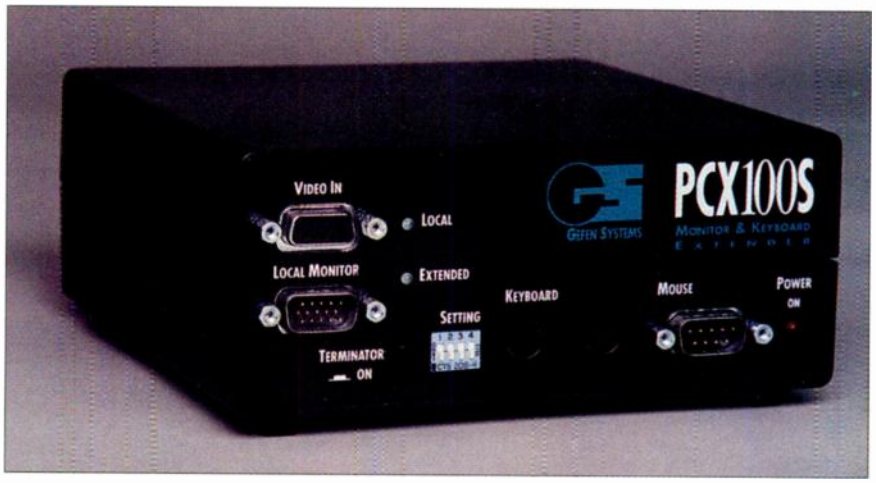

Gefen's PCX100 extends the length of Windows PC monitor, keyboard, and serial-mouse runs up to 250 feet.

devices on a chain without powering the whole system down. Some of the best SCSI accessories I have found, including rigid connectors, hot swappers, and double-shielded 12-foot cables and extensions, are available from APS Technologies.

Glyph Technologies also offers a number of useful SCSI solutions, including the GSS-210S SCSI Switch/Extender ($495). This product allows you to switch your computer between two SCSI chains (one of which can include the sampler) or to switch one set of drives between two controllers (the computer and the sampler). The product also acts as a terminator and repeater that lets you run SCSI 1 chains of up to 20 feet and SCSI 2 chains of up to 40 feet. Glyph offers several models of the SCSI Switch/Extender.

THE LONG VIEW

Finally, we come to monitors, and here is an area where it seems that "nobody knows nothin'." I discovered this unfortunate fact in a recent redesign of my studio: it took about two months to piece together all of the solutions. Granted, my needs were rather special. I have an office and a studio in separate rooms, each with a keyboard and monitor (one 17-inch VGA, one 19-inch BNC). These are connected to one CPU that lives in a closet between the two rooms. But even if I were just trying to get my monitor away from my computer, I would have come up against the same problems.

Macintosh monitors use a DB-15 connector, which has fifteen pins in two rows. VGA monitors have a DB-15HD connector, which arranges the pins in three rows. In either case, if you just stick an extension cable onto a monitor cable, chances are you will have to deal with signal loss, line noise, interference, and other factors that will seriously degrade the picture quality. The end result may be washed-out color or ghosts. Although the former is ugly, the latter can cause serious eyestrain if you stare at the screen too long, and staring at a computer screen for too long is something desktop musicians do a lot. If your monitor is an energy-saving model, you may find that the signal level drops so low that the monitor sometimes won't even kick out of sleep mode.

Once again, using top-quality cables will help. A double-shielded cable may not be good enough; an individually shielded cable, in which each of the internal cable pairs has its own shield, is what you want. Using one long cable is far better than adding extensions: the connectors serve as entry points for interference, so the fewer connectors you have, the better. The best source I've found for excellent monitor cables is Tekserve, which sells lots of hard-to-find Mac gizmos and puts out a very good, free book of frequently asked questions about the Mac.

Many monitors, particularly those made by Sony or NEC, use multiple BNC connectors. They need to be hooked up to your computer with a breakout cable that splits the signal from the VGA or Mac video connector into three, four, or five individual components: red, green, blue, and possibly horizontal and vertical sync. Each component has its own shielded connector—again, individually shielded cable is the key to making long cable runs work.

But rather than use an extension for the DB-15 or DB-15HD, use as *short* a 15-conductor cable as you can, putting the breakout as close to the computer as possible. For the rest of the run, use multiple coaxial cables with BNC connectors on each end. It's not as neat as using a single cable, but your eyes will thank you. Runs of 50 feet or more with no picture degradation are easily achieved. A good video supply house will have BNC cables in the lengths you need. Prices will run from $1 to $2 per foot, per conductor, so a 10-foot set of three cables costs between $30 and $60.

If you are using a Mac and having trouble with monitors in general, a sync adapter from Griffin Technologies can be a big help. This little in-line device takes care of a lot of problems by fooling the Mac into thinking one kind of monitor is hooked up; meanwhile, it

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Glyph Technologies' GSS-210S SCSI Switch/Extender allows you to switch your computer between two sets of SCSI devices and extends the SCSI chain's maximum length.

converts the video signal so that the picture can be displayed by almost any monitor at any resolution and sync speed. It is essential for, and is often included with, any monitor that measures nineteen inches or larger. Griffin also sells a flexible Mac-to-VGA adapter for using VGA monitors with a Macintosh.

Finally, if you have to deal with really long monitor cable runs—or multiple monitors, as in my studio—you should look into a video booster/distribution amplifier. This device raises the signal level so that the signal can easily overcome all sorts of noise problems and therefore can run great distances. Various configurations are available, in both Mac and VGA formats, with up to eight outputs.

Many folks will tell you that the only place these are available is Black Box Corporation, but don't believe them. Although Black Box has an impressive catalog of all sorts of hardware for networking, hardware organizing, displays, and so on, its prices are quite high. It looks like the type of place large corporations go when they need to get everything from one supplier and saving money on the equipment itself is a secondary consideration. For example, video amplifiers start at around $300. Still, Black Box is a source worth considering.

I've had much better luck with Datatek, which sells simple amplifiers for between $100 and $160. Like Tekserve, Datatek has a whole catalog of useful, reasonably priced Macintosh products, including a very cool, very affordable, push-button serial-port switcher.

Beware: if you have a short video cable from an older Mac that was made before Apple supplied monitors with permanently attached cables, *don't* use it between your computer and a video amplifier. Those older cables did not have all the pins wired, and they won't work with any monitor or peripheral that needs all of the pins, which most do today.

ONE-STOP SOLUTIONS

Fortunately, one company is working to simplify this entire process. Gefen Systems' TSE100 for the Macintosh ($395) is a complete system that allows extension of the Mac monitor, keyboard, and mouse up to 500 feet from the CPU. The hardware supplies the amplification required to send monitor and keyboard signals up to the maximum distance. The sender connects to the Macintosh via standard monitor and ADB cables, which are supplied with the TSE100 system. The receiver is placed next to the monitor and keyboard and connects with standard monitor and ADB cables, just as if you were connecting to a Macintosh CPU. The system is powered by an external 9 VDC supply.

The TSE100 sender and receiver communicate via multiple cables. The monitor extension cable is an RGBH&V cable made of five coax cables snaked together. The ADB extender uses standard RJ-11 6-conductor telephone cable. Cables are a separate purchase and are available from Gefen. The most expensive of these are the monitor cables, which will set you back anywhere from $109 for a 25-foot run up to $750 for a 300-foot cable. RJ-11 link cables range from 25 to 250 feet ($9 to $55).

Windows PC users can extend their monitor, keyboard, and serial mouse cable runs up to 250 feet via Gefen's PCX100 ($395). The system supports PS/2 and RS232 serial mouse devices and all PC keyboards. As with the TSE100 system, you get sender and receiver units and an external power supply. A cable that carries the coax video and all keyboard and mouse signals is available separately.

I did not have the opportunity to test Gefen's extenders because I only recently learned about them, but it appears that the company could have saved me a lot of headaches. These systems don't solve the SCSI problem, but we have already examined a few solutions to that.

Gefen offers numerous other systems and cables for connecting multiple monitors and keyboards, switching between computers, and using a remote Mac keyboard and mouse without the video extender. More information on Gefen systems and cables is available on the company's Web site.

SKELETONS IN THE CLOSET

With your computer and peripherals stashed in an equipment closet, you will have made a good start toward getting your noise level down below a dull roar. This isn't a panacea for all noise problems in the studio, of course; for example, you may still have to deal with room-ventilation fans, floor and wall vibration, and other ambient noise. (See "Home Improvement" on p. 58 of this issue for a discussion of these problems.)

Perhaps someday low noise levels will be a common computer-design goal, and specs that include noise levels will be available. Then we can bring our computers out of the closet and put them back on the desktop. For now, these hints will help you keep the noise in the closet and out of your face—and, perhaps more importantly, away from your microphones.

Paul D. Lehrman *is the author of* Getting into Digital Recording *(Hal Leonard Publishing) and coauthor of* MIDI for the Professional *(AMSCO Publications). In November 1996 he celebrated his tenth anniversary as an* **EM** *writer.*

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RECORDING MUSICIAN

Glorious Acoustic Guitars

Capture stunning sounds from unplugged six-strings.

By Brian Knave

Getting superlative acoustic guitar sounds on tape is not a simple step-by-step procedure like changing the oil in your car. In fact, the acoustic guitar is one of the more challenging instruments to record well. Think of it as an orchestra in a box. From stomach-rumbling lows to hair-curling highs, an acoustic guitar produces a wide load of timbres, tones, and frequencies. Documenting that sonic rainbow requires not only good gear but good ears, good technique, and, on occasion, the patience of Job.

BOB BURNS

FIG. 1: Veteran Nashville engineer Bil VornDick, shown setting up to record Jack Jezioro's Martin D-28, is a fervent believer in stereo miking. Here, the left mic, a Milab 96, is positioned to document the player's perspective, and the right mic, an Audio-Technica AT4031, captures the instrument's high end.

Let's assume your session is blessed with a talented musician and a good-sounding guitar. The first challenge you'll face is finding the ideal room, because *where* you record the performance may be as important as *how* you record it. To really nail the nuances of an acoustic guitar, you should match the recording environment to the artist's performance style.

For example, when recording a classical guitarist, independent producer-engineer Jeff Saltzman booked time at San Jose's Lick Observatory, a cavernous, perfectly hemispheric room. He stereo miked the nylon-string guitar with his favorite mics for recording acoustic guitar: two Neumann KM 84s fitted with KM 83 omnidirectional capsules. "I pulled the mics about six feet back," says Saltzman. "It's amazing what the room did for the sound. It was like having the most expensive Lexicon reverb with all the diffusion and predelay perfectly dialed in, only better."

On the other hand, an acoustically dead room is usually better for tracking guitar parts that are destined for a crowded mix and/or intense signal processing. Too much room sound in these applications may cause the acoustics to appear diffuse or "washy," and can effectively destroy any contribution the guitar was supposed to make to

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the audio spectrum. In addition, putting effects on a track that already includes a room sound means that the room, as well as the source sound, will be effected. Saltzman, who has worked extensively with Francis Ford Coppola's movie studio, American Zoetrope, solves these dilemmas by recording the acoustic guitar in a padded isolation booth.

"I sometimes go for the classic 1970s, AM-radio acoustic guitar sound, which requires a lot of compression," he explains. "And that sound also requires doubling the guitar part. Therefore, if I was to record the guitarist in a reverberent space, the compressor would increase the volume of the low-level room sound. Then, when the player doubled the part, all the different reflections in the two performances would make the guitar sound really 'comb filtery.'"

If your goal is to capture a realistic, natural-sounding portrait of the acoustic guitar, your best bet is to record in a natural-sounding room. That's simply because, in the real world, people listen to guitars in ordinary rooms, not iso booths or observatories. When it comes to recording unsullied, organic acoustic guitar tracks, one recognized master of the genre is Nashville veteran Bil VornDick. Originally chief engineer for country-music legend Marty Robbins, the award-winning VornDick has recorded countless artists, including James Taylor, Doc Watson, Chet Atkins, Rickey Skaggs, Tony Rice, and Alison Krause. "I like a medium-sized, neutral-sounding room, not overly dead or live," says VornDick. "And it needs to have a hardwood floor to get some natural reflections."

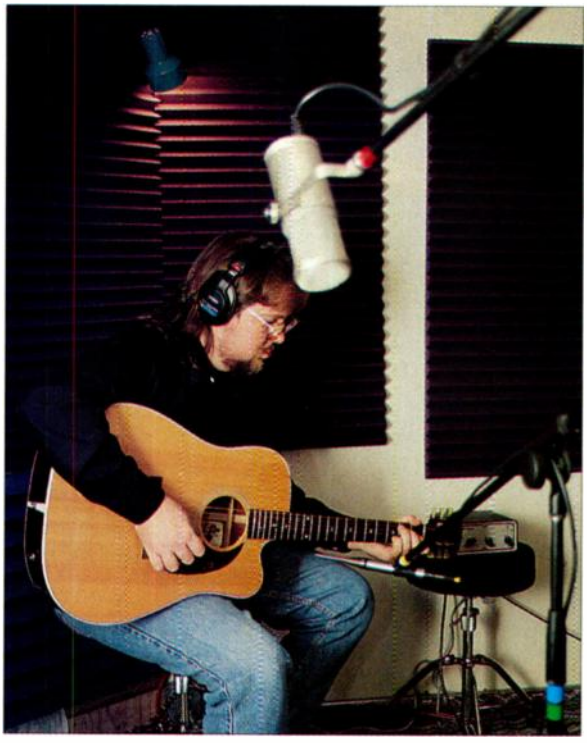
If you don't have a wood floor or are forced to record in an acoustically dead space, you can liven up the sound by placing a large wooden board on the floor beneath the guitarist. I keep several boards handy for this purpose and have even erected them vertically on either side of the guitarist to create reflections.

LET MIKEY TRY IT

Second in importance only to the quality and musicality of the sound being made in the studio is the choice and placement of microphones. Three parameters should govern your miking strategy: the type of microphone best suited for the job, which polar pattern is most appropriate, and whether you are recording a source sound in a monaural or stereo perspective.

TYPE CASTING

The choice of microphone usually comes down to dynamic or condenser, and due to the complexity of sound emanating from an acoustic guitar, a condenser is almost always the better choice because it is more sensitive. However, for aggressive rock or filmscore tracks, a dynamic mic can be used to produce a cranking midrange tone that will cut through almost anything. Depending on the model used, the dynamic microphone also may not reproduce the low-frequency rumble of the guitar as well as a condenser would—and this can be a good thing when the acoustic is adding mud to a track.



BRIAN KNAVE

FIG. 2: You can create a more complex stereo picture of the acoustic guitar by using two different mics. Here, an Earthworks OM1 is aimed just below the strings to capture sparkle and attack from songwriter Mike Lawson's Gibson J-30C, while a Rode Classic, set to omni, reproduces the broader, ambient sound of the guitar.

POLAR REGIONS

The two most commonly used polar patterns are cardioid and omnidirectional. Obviously, the cardioid pattern is useful if you're recording other instruments in the same room simultaneously, because the pattern's rear rejection minimizes bleed from non-source sounds. However, if there is no interference from other instruments, be sure to audition an omni mic. The omnidirectional pattern typically gives a bigger picture of the sound, which can be especially desirable if you're using a single microphone.

Of course, opinions vary about the use of cardioid versus omnidirectional microphones. Saltzman feels that using a cardioid pattern on an acoustic instrument produces a "pinched" sound, so he prefers to use omni mics. "A mic capsule is naturally omnidirectional," he explains. "To make the mic cardioid, a manufacturer has to do acoustical porting and other modifications. So a cardioid mic is basically a crippled omni. A cardioid doesn't capture all the sound coming from an acoustic guitar. When I put a mic on an acoustic, I want to pick up as much sound as possible, so I go with an omni."

Another advantage of omnidirectional mics is their lack of proximity effect, which means bass frequencies don't multiply like bunnies as the mic gets closer to the sound source. This attribute lets you position the mic closer to the sound hole without increasing boominess. A corollary benefit (for small-diaphragm models only, because they employ a single capsule that does not "hear" behind itself as well as a dual-capsule, large-diaphragm condenser) is the relationship of mic-distance to ambient sound: the closer the mic is to the guitar, the less room sound it hears. This is an important consideration if all you have to work in is a crappy-sounding room. I recently recorded a vintage Lowden guitar with a single Earthworks TC30K (small-diaphragm omni) positioned only inches from the juncture of fretboard and sound hole. The result was a stunningly natural, in-your-face acoustic guitar track with no hint of room sound.

TWO ON ONE

The issue of whether to record a mono or stereo perspective of an acoustic is

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FIG. 3: A Rode Classic tube condenser microphone set to the omnidirectional pattern is positioned to capture the blend of warmth (from the sound hole) and high end (from the strings) of a Gibson J-160E.

typically decided by aesthetic considerations and available tracks. Let's discuss tracking in stereo first because, although it offers more tonal and spatial options than mono, it is at least twice as hard to get right. For one thing, the addition of another mic introduces potential phase cancellation problems. If you detect a hollow or thin sound, activate your mixer's phase reversal switch (on *one* of the input channels only) or move the mics further away from each other until you hear a full-bodied tone. Of course, if phase cancellation is responsible for a strange but wonderful tone, go with it.

One advantage of stereo miking an acoustic guitar is that you can position one mic to capture the instrument's treble qualities and the other its bass. In this case, you're listening for two sweet spots, each with a different character. In fact, you can further emphasize the different tonalities by using unmatched mics. For example, you could train a small-diaphragm condenser near the strings and use a large-diaphragm near the bridge or sound hole, or vice versa.

When recording acoustic guitars, VornDick swears by stereo miking. ("It gives the sound dimension," he says.) He aims one mic, pointing downward, at the sound board between the top of the guitar and the bridge (see **Fig. 1**) and positions the mic so that it's about the same distance from the guitar as the player's right ear is (assuming the guitarist is right-handed). "In essence, this mic picks up what the player hears," says VornDick. "Of course, I then move the mic around until I find the sweet spot.

"The other mic," continues VornDick, "is aimed about three or four inches below the neck, between the sound hole and where the neck joins the body, but not pointing at the strings. It's about five to seven inches away from the guitar. With this mic I'm trying to capture the high side of the instrument's ambient structure without emphasizing the sound of the strings on the fretboard. Naturally, the positions may change depending on the guitar."

During his years of recording acoustic guitars, VornDick has developed allegiances to certain microphones. Some of his favorite mics for the job are Sanken 31s or 32s, Milab 96s, and Neumann KM 84s. Because these are expensive models, he recommends Audio-Technica's affordable AT4033 and AT4031 for the budget-conscious recordist.

In the mix, VornDick pans the stereo tracks in a V. "I never pan them hard left and right," he explains. "That would sound too spread out. The widest I would ever get is one track at nine o'clock and the other at three o'clock—and then, only if it is a guitar-based song. As more instruments are added to the mix, the size of the V gets smaller."

Another way to mic an acoustic guitar in stereo is with one mic up close and the other five or six feet back, positioned as an overhead looking down on the guitar. Again, you can use a matched pair of mics or two different mics, if you want to broaden the tonal picture (see **Fig. 2**). The two signals can then be blended to taste in the final mix. This is an excellent application for a guitar-based song with little or no other instrumentation because it creates a spacious stereo image.

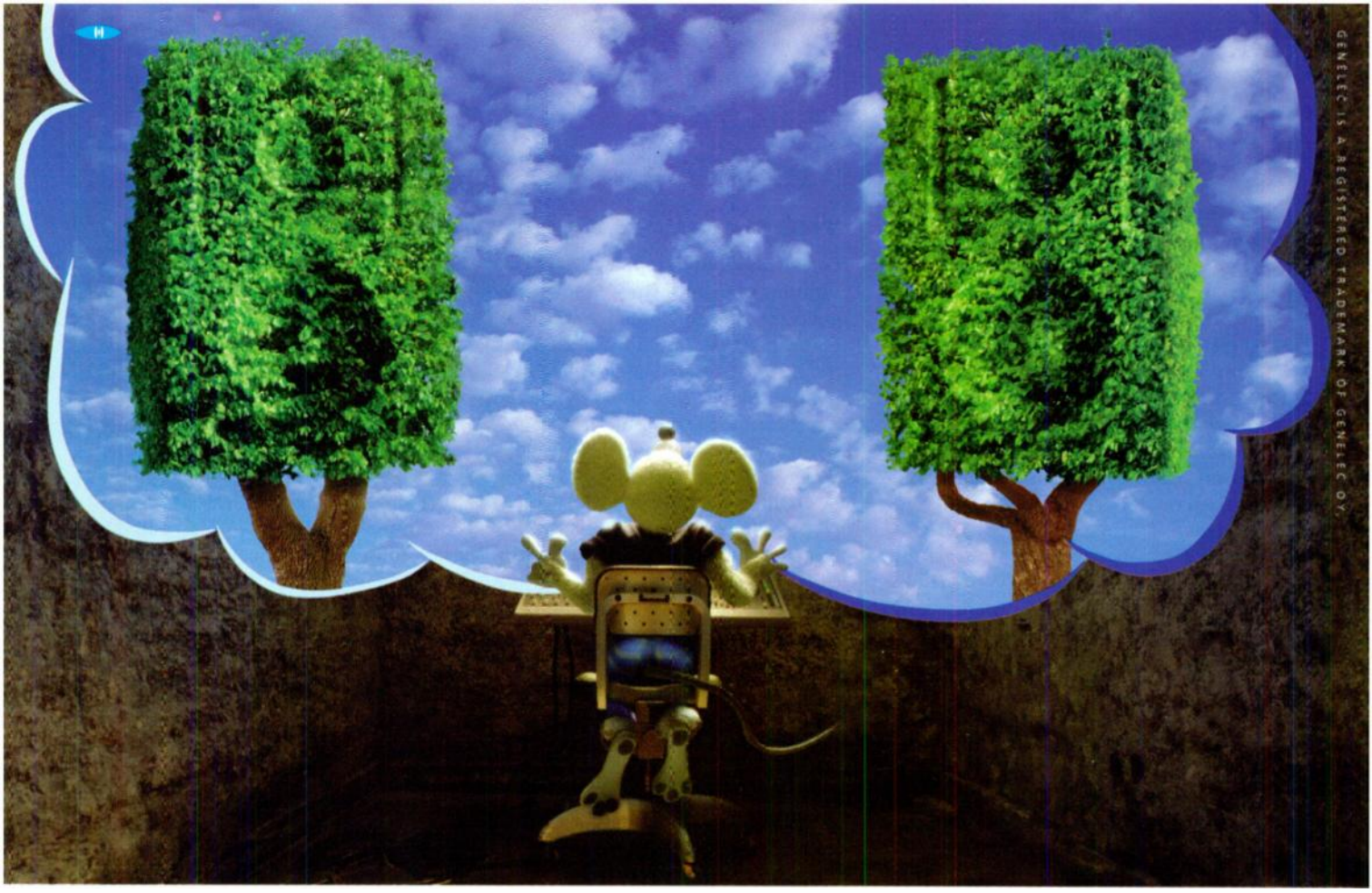
ONE ON ONE

For monaural recordings of an acoustic guitar, try to find the spot with the best balance of low- and high-end frequencies. This is usually somewhere between the sound hole and upper frets, with the mic far enough back—six to twelve inches—to capture a blend of warmth from the sound hole and zing from the strings (see **Fig. 3**). If the guitar sounds thin, angle the mic toward the sound hole; if the tone is boomy, angle the mic more toward the strings. Also, move the mic up and down. Try several positions, and keep in mind that the slightest difference in mic placement can cause a big change in sound.

Legendary Beatles producer George Martin used to listen for the sweet spot by putting a hand over one ear to simulate the perspective of a single mic. In fact, whether you're tracking in mono or stereo, there's no better way to position a mic than by first listening to the instrument with the unaided ear. However, because a microphone invariably hears differently than the human ear, be sure to audition mic placement by listening through your monitors (or headphones), as well, to hone in on what the *mic* perceives as the instrument's sweet spot.

TWEAK CITY

Compression is the most common and useful type of signal processing for acoustic guitar, as it can enhance the impact of the guitar while maintaining a "natural" sound. Although you can compress while tracking, mixing, or both, engineers have different opinions about what works best. Saltzman, who likes using a Urei 1176 or 1178 for acoustic guitar, says that he usually avoids compressing while tracking.



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"When you close-mike an acoustic guitar," he explains, "you're going to pick up a lot of low-end frequencies, which can drive a compressor crazy. The compressor will be pumping and breathing while trying to hold back the low end. But in the mix, you can dial out excessive low end and make any other necessary EQ adjustments *before* compressing. That way, the compressor is only reacting to the final sound."

VornDick, on the other hand, prefers to compress while tracking. "But I only use the compressor as a catch-all," he clarifies. "I don't try to squish the sound. Usually a very light setting, around 2:1, is sufficient." VornDick, who favors the Aphex Compellor and Rane MC 22 compressors for acoustic guitar, also tweaks the EQ a bit while tracking. "Martin guitars," he explains, "often have a muddy bottom. So I'll use a highpass filter to roll off everything below 80 or 90 Hz. Or if the boom is really drastic, I'll switch mics until I find one that de-emphasizes the problematic frequencies."

Bay Area independent producer-engineer Tom Luekens, formerly Booker T. Jones' engineer and currently working with Windham Hill Records and Tiki Town studios, often compresses lightly while tracking and then again in the mix. "The denser the mix—and the more instruments there are playing similar rhythms in the same tonal range as the guitar—the more I'll compress," says Luekens. "But I rarely go above a 4:1 ratio for acoustic guitar because it starts sounding funny."

Luekens has been getting good results lately with the Focusrite Red 3 compressor. "The Red 3 is great; I typically set the attack and release controls on automatic, which works just fine for most applications. But if I do adjust the parameters manually, I usually like a medium to fast attack and release. If you have the release set too long, the compressor tends to pump because the next thing that comes down the audio stream will catch the tail end of the compression from the previous note."

Unfortunately, compression can't cure all ills. For example, a performance that features both fingerpicking and strumming can present problems due to varying dynamic levels. A compressor can level out the dynamic range to some extent but not always enough to optimize the nuances of each style. Saltzman has another solution (see **Fig. 4**): "I try to persuade the musician to perform the two sections on separate tracks. That way I can close-mike the fingerpicking for a wonderfully dramatic effect and then pull the mic back enough to get the strong, rhythmic impact of the strumming."

If you're looking to forge beyond *au naturel* acoustic guitar tracks, a wealth of effects and studio tricks are at your disposal. Most multi-effects processors contain a number of preprogrammed patches—typically chorus-based—designed to enhance the acoustic guitar. Of course, you can always experiment with other effects and develop your own processing tricks.

On a recent album by Japanese artist Jun Hirose, for example, Luekens and producer Scott Mathews created a

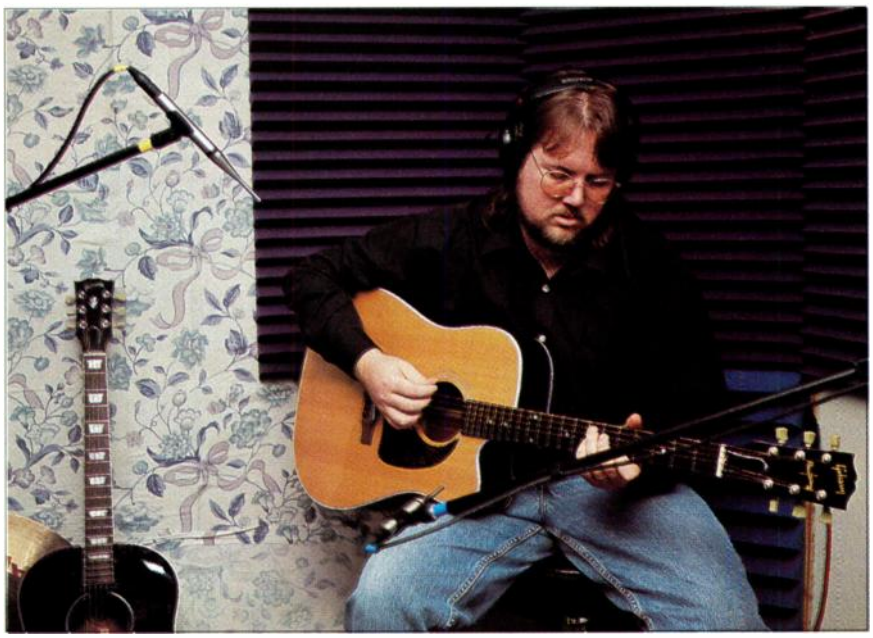
(continued on p. 201)

THE NASHVILLE THANG

I've always wondered how Nashville engineers create the wonderfully crisp acoustic guitar tone that has become a classic element of country music productions. I have been able to approximate the sound by dialing out a lot of low end, but in the process of researching this column, I discovered that it takes more than that to get the real thing.

"To achieve a classic country rhythm-guitar sound," explains VornDick, "you double the part with a high-strung guitar. A high-strung guitar has the three bass strings tuned an octave higher than standard tuning. You need to use thinner-gauge strings on the E, A, and D strings to do this. Then the guitar is recorded with a single mic positioned above the player and shooting down at the sound hole. At the board, the engineer cuts out the bottom end, boosts the top, and slams the signal (compresses it extremely hard).

"The goal is to have the tone of the acoustic match the hi-hat," continues VornDick. "You're going more for a rhythmic 'ching' than the full-bodied sound of the instrument. Country engineers also do this trick to clear a space for the piano, because the guitar and piano are so alike in their overtone structures."



BRIAN KNAVE

FIG. 4: To ensure near-perfect documentation of each style, engineer Jeff Saltzman recommends recording fingerpicked and strummed performances in two passes, on two separate tracks. Here, Lawson strums his Gibson J-30C while being stereo miked with a pair of Earthworks OM1 omnidirectional condensers.

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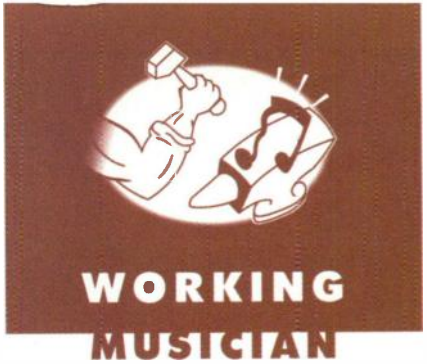


New Frontiers in Copyright

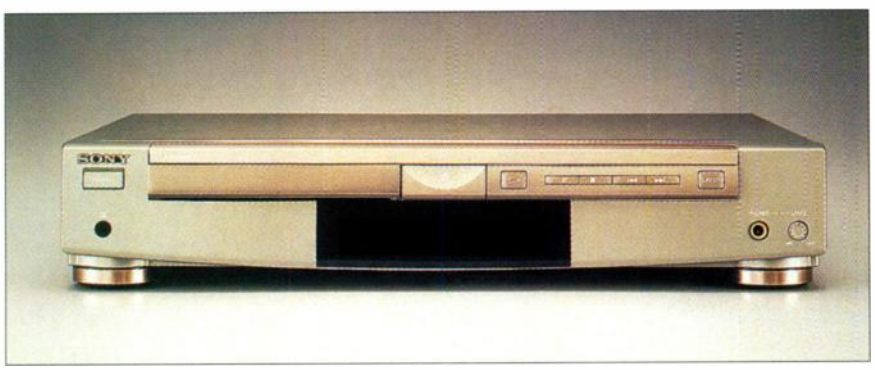
Technology marches on, but copyright law still controls the pace.

By Mary Cosola

The entertainment industry has an annoying habit of getting consumers all fired up about a new technology and then taking years to deliver it. Take digital versatile disc (DVD), for example. This new format crosses video, audio, and computer platforms, packing a ton of data onto a disc the size of a regular CD. In fact, one DVD can hold an entire motion picture. But the capital-V versatility of DVD also proved to be the thorn in its side when it came to bringing the format to market. A major problem that held up the delivery of DVD players for months was the implementation of copy protection, proving the adage, "The devil is in the details." Research and development can conquer new worlds, but the legal wrangling over details like copy protection can bring science to its knees.

"What it comes down to is that the computer industry and the motion picture industry were at odds over the copy-protection scheme," explains Mark Ely, DVD product marketing manager for Sonic Solutions, which at press time was getting ready to release *DVD Creator*, its DVD premastering system. "In its early phases, DVD went from a video-only format to a cross-platform medium. For the motion picture industry, the main question was, 'Do people with personal computers now have access to my movies?' They were especially worried about digital versions of movies floating around."

Ely also points out that the computer industry has pretty much accepted some piracy as a fact of life whereas the movie industry has not. Software companies have relied on built-in copy-protection schemes such as requiring a serial number during installation, but with DVD, the motion picture industry wanted a stronger approach. The problem is that a decrypter can easily be built into stand-alone DVD players, but what about existing computers? The computer industry lobbied against copy protection that would require a hardware addition to PCs, arguing that it would deter PC users from making the move to DVD. In the end, the two sides agreed on an encryption scheme



Shown above is a prototype of a Sony DVD video player. Hard to believe that such an innocent-looking box could be responsible for a copy-protection battle royale between the motion picture and computer industries. It took the two industries years to come to an agreement on a built-in copy protection scheme for DVD.

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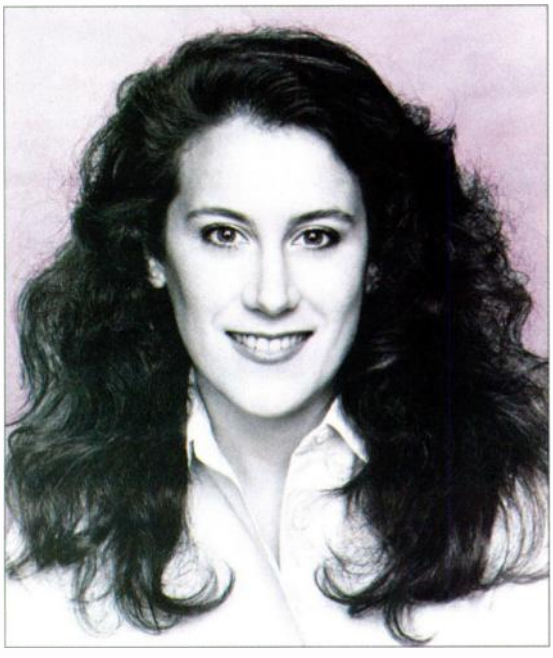
The lesson we can learn from the DVD dilemma is about the value companies place on their copyrights. As the computer and entertainment industries continue to create new delivery formats, more discussion on copyright will ensue. In addition to DVD, other electronic copyright issues have also recently emerged. These include a recent U.S. Copyright Office opinion on Standard MIDI Files and the use of copyrighted material in new media, such as Web pages.

STANDARD MIDI FILES

In September 1996, the Copyright Office wrote that SMFs are no different than other musical recordings and are subject to the same copyright laws. The SMF format has been around since 1988, five years after the introduction of the MIDI spec, but it has taken eight years for SMFs to come under the same royalty category as CDs, albums, and cassettes. The Office's opinion states, in part, "The output of Standard MIDI Files are works of authorship copyrightable as sound recordings, since the information in the file causes the sound device to render the pitch, timbre, speed, duration, and volume of the musical notes in a certain order, as does a player piano in conjunction with a piano roll or a compact disc player in conjunction with a compact disc." The rights to SMFs have always been negotiated on a case-by-case basis, but the Copyright Office opinion could mean that SMFs will now be subject to the statutory per-song licensing rates for mechanical royalties (currently 6.9 cents per song).

In January 1996, the MIDI Manufacturers Association (MMA) established the SMF Copyright and Licensing Committee, whose goal is to create a market for SMFs in the United States. The hope is that by working to standardize the practice of licensing and selling SMFs, the market will become more active—and therefore, more lucrative. Throughout Japan and Europe, SMFs on diskettes are sold in stores and via mail order in the same manner as CDs and cassettes. (According to an MMA press release, mail-order SMF outlets have been known to sell as many as 10,000 disks a month, at a per-disk price very close to that of a compact disc.)

The Copyright Office's opinion came about when the SMF committee was doing some fact finding and asked the Office where MIDI files fell in the scheme of things. Tom White, president and CEO of the MMA, points out that the Office's opinion is good news for the industry and musicians. He explains, "Now that it is clear MIDI files aren't legally any different from other music formats, musicians can be more comfortable about pursuing it as a way of distributing music. They now know that they have more rights with regard to registering their songs as MIDI files and collecting royalties. Also, music publishers were reluctant to pursue publishing MIDI files in the past because they didn't know what the rights were."



Jill Alofs, founder and president of Total Clearance.

Currently, the fastest growing market for PCs is in the home. And as the average consumer becomes more computer savvy, the market for SMFs should increase accordingly. White explains that music education is one area that could really benefit from the increased availability of SMFs. "Music education has always been centered around the use of popular tunes," he says. "Obviously people in the industry know where to get or how to create MIDI files, but the average user doesn't know where to find them or how to use them. Users will be able to buy SMFs and listen to or view arrangements of songs and study them. Also, the increasing popularity of digital pianos, many of them with built-in sequencers, means that people will want to buy SMFs for that purpose, too."

SCOTT L. O'NEILL

Tom White, president and CEO of the MIDI Manufacturer's Association.

SAMPLE CLEARANCE

One thing that shouldn't be a mystery to *any* **EM** reader is that it's illegal to use a sample of someone's work without getting permission and paying the proper royalties. (For more on royalties and other legal aspects of putting out a release, see "Working Musician: The Fine Print" in the February 1996 issue.) But with the onslaught of new media such as CD-ROM, Web pages, and DVD, there is a screaming need for content, so developers often opt to license audio or visual clips for their projects. The biggest challenges most developers face is where and how to obtain clearances.

"The tricky thing about getting clearance is that you have to know who to contact, how to prepare your materials, and which questions to ask," says Jill Alofs, president of Total Clearance, a company specializing in rights and clearance issues. The company works with traditional media such as film and video, but the booming side of the business is in new media.

"When we receive a request from a developer, we start by asking them an array of questions," she says. "We need

(continued on p. 201)

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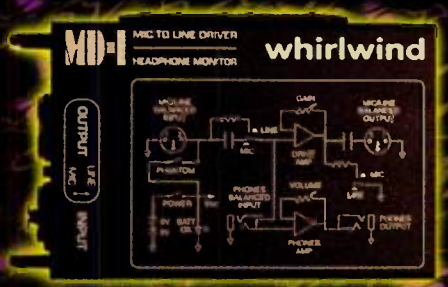

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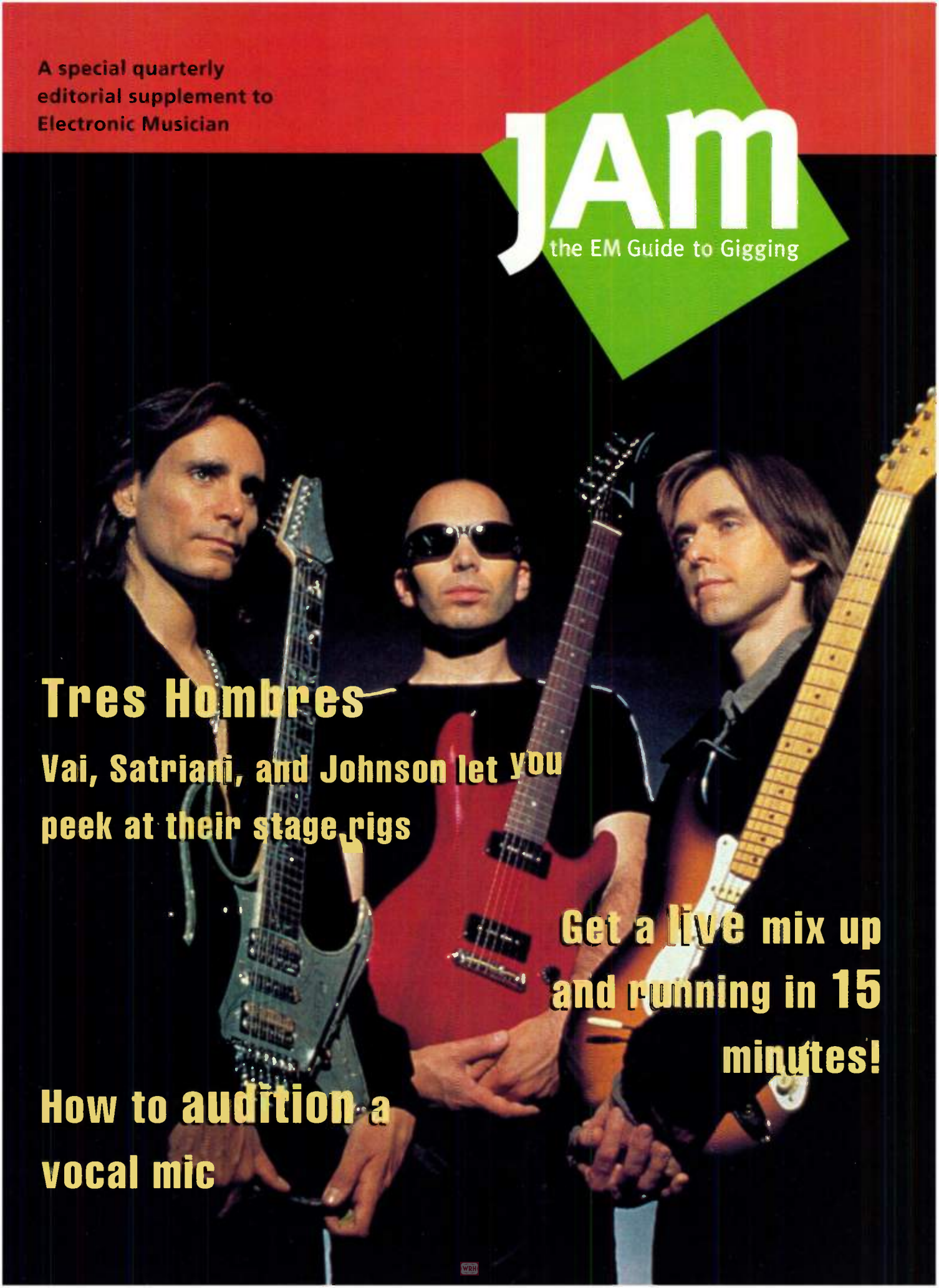

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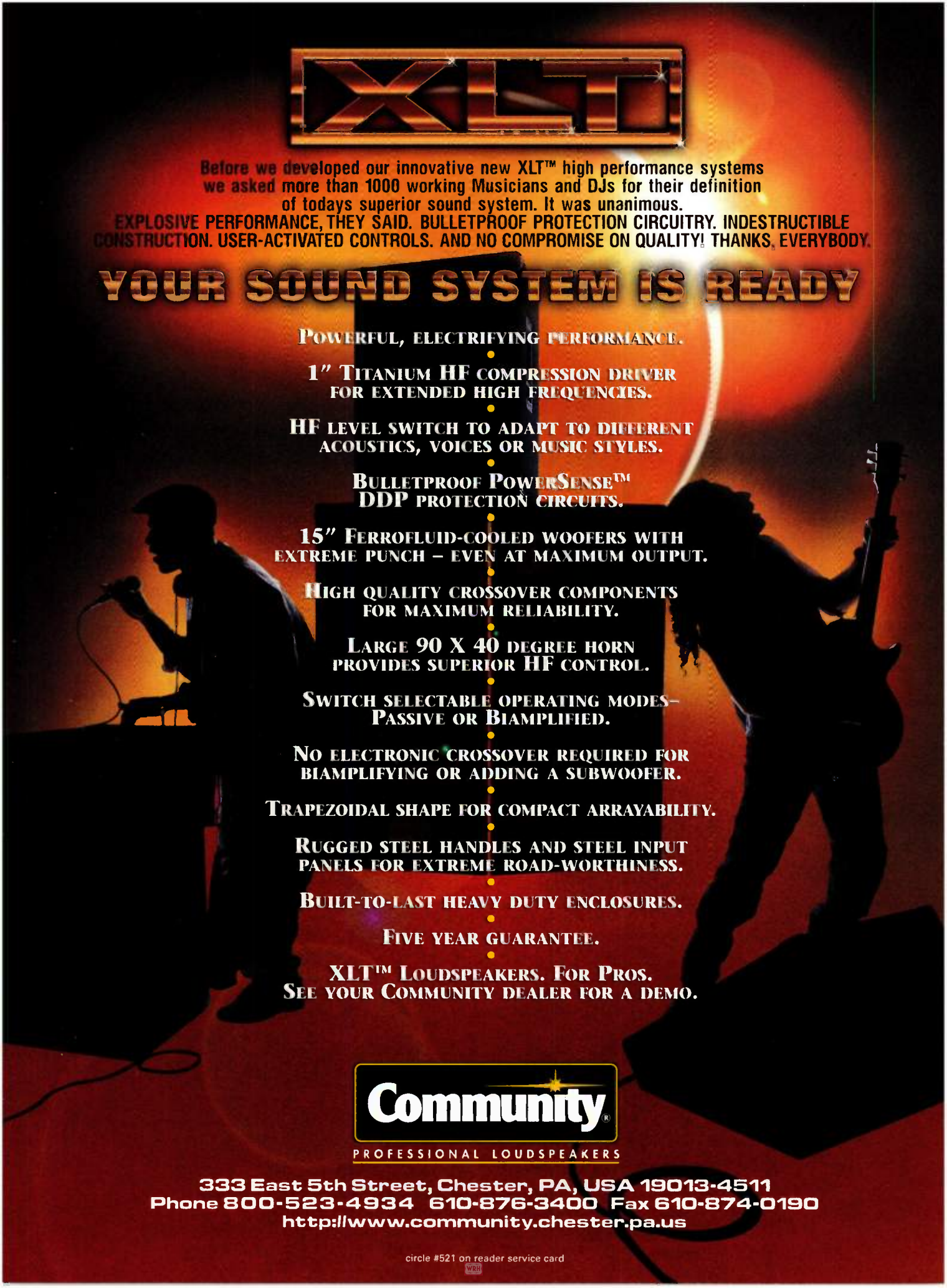
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Cover Photo: Neil Zlozower

intro

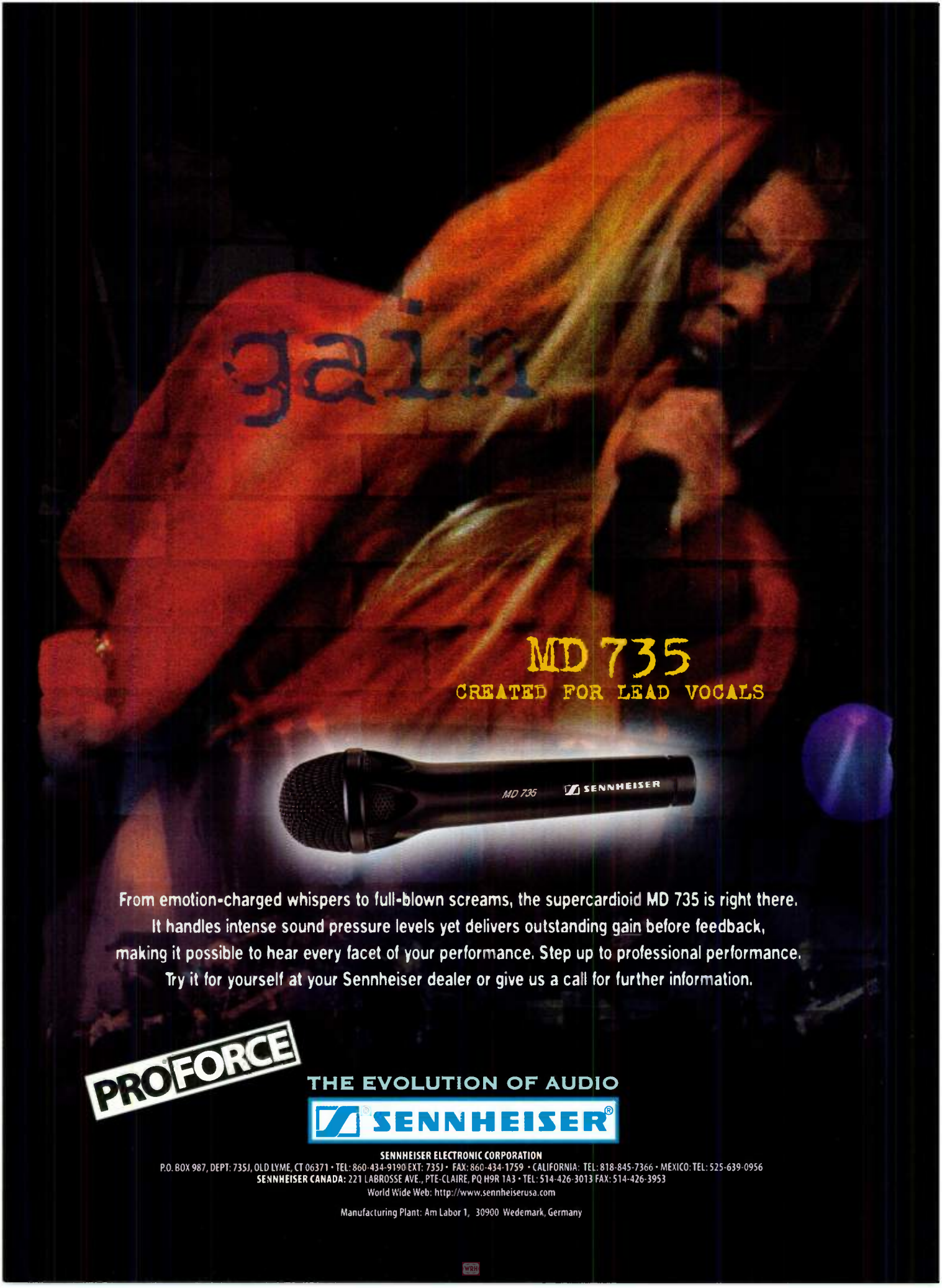
THAT *JAM* EXISTS IS A TEENY, TINY MIRACLE. The *idea* of it has been bouncing around the **EM** office for years—after all, nearly 60 percent of our readers perform live, and it was obvious they could benefit from a performance-oriented magazine. However, the *execution* was elusive. How do you create an entirely new editorial product when your staff is overwhelmed producing **EM**? Needless to say, when our publisher, John Pledger, expressed a desire to "front burner" the live project for our 1997 NAMM issue, it was panic time.

Due to the brutally compressed deadline, I established the editorial and basic design concepts of *JAM* during a long shower and developed the table of contents on my commute to the office. (My brain hurt a lot that day!) Then, I had to break the news to our talented but overworked art director, Linda Birch, that she had the, um, honor of designing a new magazine. I still can't believe that Linda came up with the graphic personality of *JAM* in less than a month and really nailed it. I hope you will agree that the visual impact of the supplement is absolutely dazzling—it's colorful, fun, and totally mod. (And check out the cool icons designed by artist/musician Robert Pizzo, especially the Gibson Flying V with whammy bar!)

Now, in order to keep the editorial staff from wiring a bomb into my little Toyota Tercel, freelance authors and editors were invited to contribute to *JAM*. (Four of those "lifesavers" are pictured stage left: ripping guitarist and guitar teacher **Jason Douglas**; our own Marketing/Operations Assistant—and classically trained flutist—**Meg Ryan**; frequent **EM** contributor **Greg Pedersen**; and the man who has made his mark on more than ten million albums, drummer/producer **Scott Mathews**.) I'm happy to report that, ultimately, I only needed a few hours of **EM** copy editor Diane Lowery's time and thus managed to keep my limbs attached to my torso. So, after two months of acidic tummies and disrupted sleep patterns, Linda and I breathed approximately 700 sighs of relief and *JAM* was born.

What's the deal on this new 'zine? Well, *JAM* will run quarterly, appearing next in our May, August, and November issues. We have made a tremendous effort, in the presentation of the supplement's editorial and graphic content, to evoke the adrenaline, power, and fun of playing live. We took this stance because most of the previous coverage of live performance, I believe, has been mired in staid, almost academic tomes about too much technology, when all musicians really want is to get their ya-yas out. As a result, *JAM* is filled with short, right-to-the-point features that *solve a problem* for you. No theory, no history, no overblown prose. Just the bottom-line approach on how to improve your live-performance chops. And the articles are short enough that you can toss *JAM* around during a break at a rehearsal and absorb some valuable tips before you pack up your gear. So what am I still blabbering for? Let's hit the stage! Here's to some great gigs...

POLAROIDS OF (FROM TOP) JASON DOUGLAS, MEG RYAN, GREG PEDERSEN, AND SCOTT MATHEWS BY MICHAEL MOLENDA



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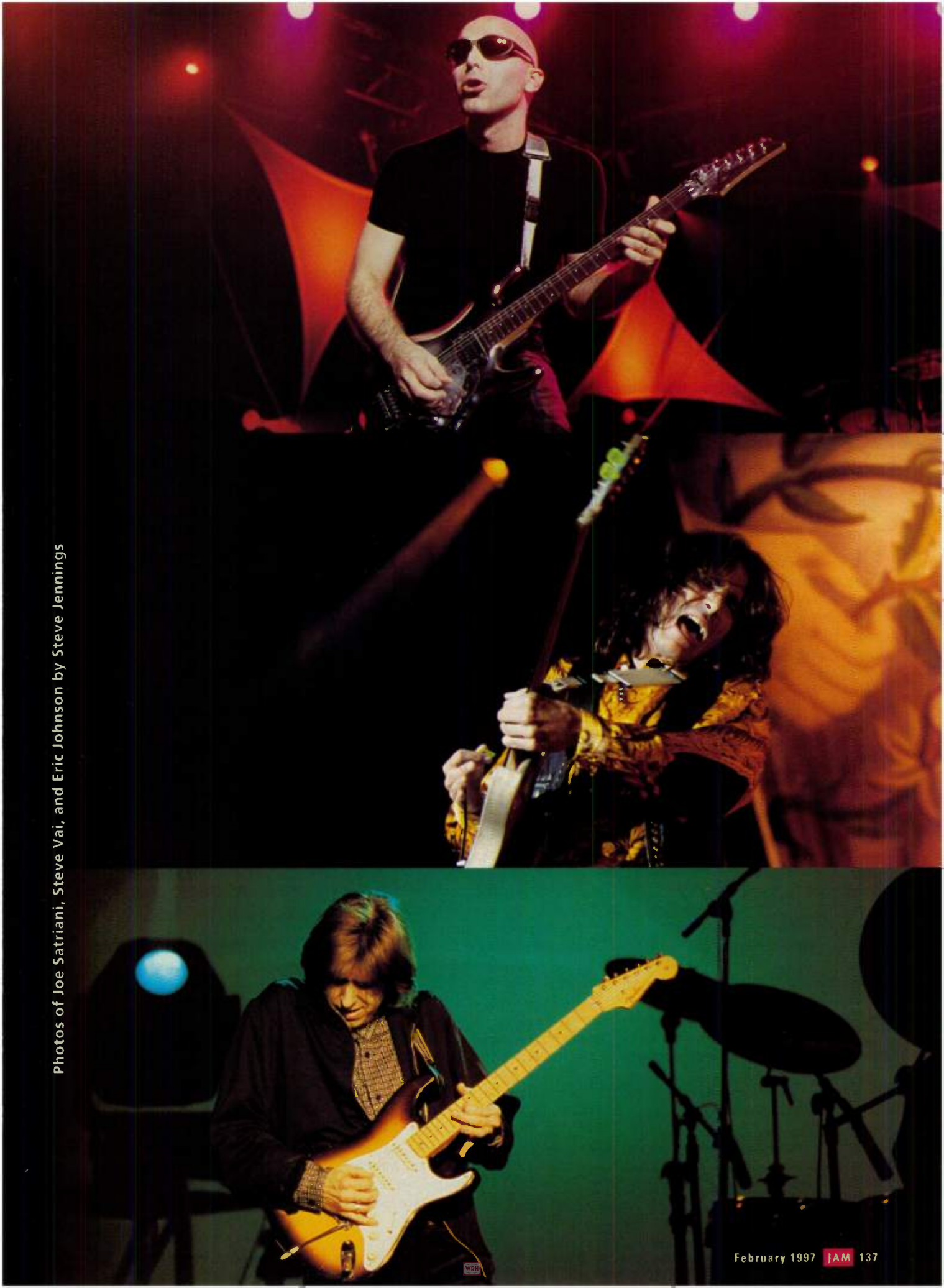
BY JASON DOUGLAS

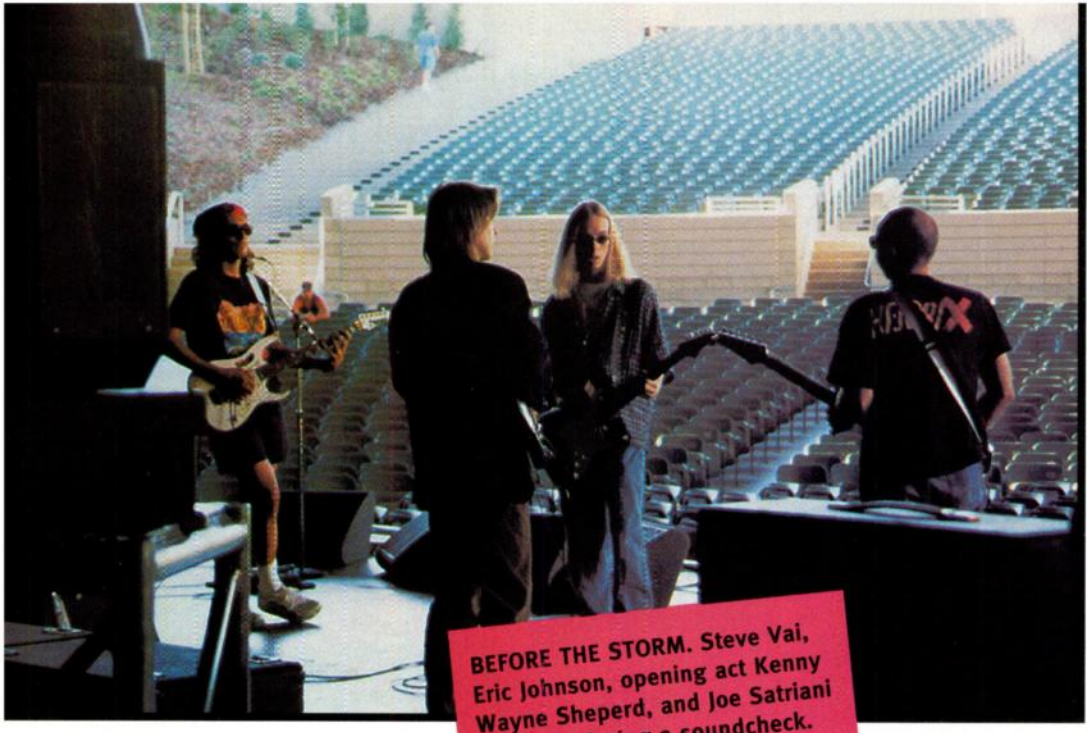
Thunder of the GODS

THE QUEST for the ultimate tone haunts all musicians. It doesn't matter if you're a guitarist, bassist, drummer, or keyboardist, *everyone* struggles to find a truly individual and awesome sound. Unfortunately, many of us are plagued with less-than-transcendent tones that seem to change at every gig. And not for the better, either! Luckily, we often have sonic deities we can look up to, musical heroes that show us what we should aspire to achieve. For the rock guitarist, Steve Vai, Joe Satriani, and Eric Johnson pretty much epitomize the ideal of "monster tone." To get some idea of how these guitar gods manage to produce such inspired soundscapes, we asked each tone master if they'd let us check out the systems they used on their recent G3 tour.

Three guitar heroes reveal the instruments of their tonal fury.



Photos of Joe Satriani, Steve Vai, and Eric Johnson by Steve Jennings



BEFORE THE STORM. Steve Vai, Eric Johnson, opening act Kenny Wayne Sheperd, and Joe Satriani talk tone during a soundcheck.

THE AMPS

JOHNSON: I'm using three different amplifier systems that I switch between for tonal variations. For clean sounds, I'm using a couple of black-faced Fender Deluxe Reverbs. When I want a crunch tone, I'll use a 50-watt Marshall that's running through a 4 x 12 cabinet to get some real dirty rhythm tones or another 50-watt Marshall—that's also connected to a 4 x 12 cabinet—that is set up for bluesy, Clapton-like lead tones.

The speaker cabinet that I use for the dirty rhythms is fitted with 25-watt Celestrion speakers that are wired at 16 ohms. For the solo work, however, I use a 16-ohm cabinet filled with 80-watt Celestrions that Marshall used to make in the 1980s. They make a reissue of the cabinet now, but it's wired for 8 ohms.

SATRIANI: I'm using a 100-watt, Marshall 6100 head. The speaker enclosures are two 4 x 12 Marshall cabinets loaded with 25-watt Celestion speakers.

VAI: My rig consists of two 100-watt Bogner Ecstasys. I split the sound of one amplifier into a stereo signal by patching it through the stereo processors in my effects rack. The stereo output from the effects is then routed to two cabinets that are positioned on the left and right sides of the stage. The other Bogner amp is placed in the middle of the stage and is used for a clean sound—no effects run through that amp at all. All of the speaker cabinets are Marshall 4 x 12 enclosures with 25-watt Celestrion speakers.

I've been using Bogner Ecstasys for about three tours now, and I've found them to be very consistent amps, sound-wise. This was not the case with Marshall amplifiers. I've been through just about every Marshall under the sun, and they are wonderful amps, but they sound so different every time you get on stage to play. I just could never rely on them to give me the same sound every night.

THE EFFECTS

JOHNSON: My main stage effect is a BOSS SC-2 chorus pedal. I also use an MXR digital delay to spread out the guitar sound a little with some echo. I basically set the delay and forget about it—the same effect parameter is running all night. In addition, I set up my reverb sound from the stage, typically with a Lexicon PCM-70, rather than leave it to the sound person.

SATRIANI: My live setup is actually pretty simple; I'm not into stumbling all over pedals and things. I usually use a Dunlop Crybaby wah-wah pedal, along with a Boss DS-1 distortion box, a DigiTech Whammy pedal, and two Chandler, stereo digital delays.

VAI: In the stereo effects rig, I rely mostly on Eventide's DSP 4000 and H3000, and a Roland SDE-3000 delay. However, manufacturers will, from time to time, give me some other effects just to try out, and those go in the rack, too. The main foot pedals are a Boss DS-2 distortion and a wah that I'm working on with Morley, called the Bad Horsie. Like Joe, I also use the DigiTech Whammy pedal.

THE MICS

JOHNSON: For the most part, I'm using Shure SM57 microphones. They always seem to get a great tone on distorted guitars. For clean tones, however, I'll often use AKG C 414s on each of my Fender Deluxe Reverbs. If I'm doing a clean kind of twangy thing, I may even plug direct into the mixing console.

For my clean rhythm tones, I'll put the mic right up against the grille, pointing directly at the speaker. On the Marshall, which is producing most of the distorted tones, I'll position the SM57 a little off-axis, more towards the edge of the speaker.

SATRIANI: For the guitars, it's pretty much all Shure SM57s. Doug, my soundman, has been with the band for quite a while, and he has an amazing knack for getting the real tough, natural sound of the band out there to the audience. He was the one who recommended using the SM57s.

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Joe Satriani www.satriani.com
Steve Vai www.vai.com

We position the mics just a little off-center from the speaker cone and approximately two inches away from the grille. Well, at least that's where they start. By the time the set has been going for a while, the mics have probably been bumped out of place three or four times! But we *try* to keep them as close to the speaker cones as possible.

VAI: We were using Shure SM57s, but we are currently in the process of changing over to another model. It's just a matter of trying new things and seeing what works best. I don't know what we may end up with.

But whichever mic we use, it is usually positioned about a quarter of the way from the center of the speaker cone at a 40 degree angle. I've found that this position delivers a good amount of top-end articulation without sacrificing

PHOTO BY BRAD NEVILLE

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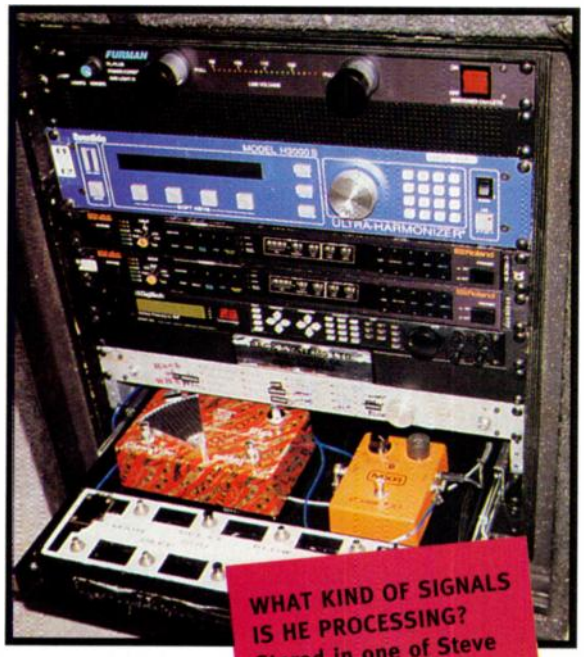

WHAT KIND OF SIGNALS IS HE PROCESSING? Stored in one of Steve Vai's effects racks, a rather arcane pedalboard includes three footswitches labeled "God," "Earth," and "Kill." Don't ask.

bottom end. In addition, I will usually pull the microphone back about three or so inches from the speaker cone. We only mic one speaker on each 4 x12 cabinet because I've discovered that miking several speakers on a single cabinet—which a lot of people seem to do—can cause a lot of phasing problems. The mics have to be in exactly the same position on each speaker or the phase relationships will be out of whack, causing your guitar sound to be too edgy and thin. This may seem strange, but I've found that you can put several mics on a single speaker and not get as dramatic a phase-cancellation problem as when you are miking different speakers on one enclosure.

PARTING SHOTS

JOHNSON: Well, as far as getting a good live sound goes, there is no shortcut to practicing with your rig and trying different combinations of all the elements involved. You really can't overlook anything—from your guitar and the strings you use, to the pickups, cords, amps, jacks, tubes, and signal processors—because it all adds up. How all of that gear interacts is what ultimately produces your guitar tone. Believe me, it all makes a difference.

Now, once you figure out an optimum setup, it's a good idea to take very precise notes on what you've done, to ensure that your sound remains consistent. It's especially true when you are touring that a consistent setup is the key to producing a good tone. What you really want to do is avoid the typical guitarist's nightmare, which is that your guitar sounds great one night and then terrible the next. The reason for such frustrating situations is usually because everything gets hooked up differently every night.

SATRIANI: The key to everything is sound. You simply must sound good! Even if you have the best band in the galaxy, if that band isn't producing a great overall sound, there's nothing anyone can do to save your show. People will not get it. They'll be too busy complaining about how crummy everything sounded. One thing that you can do to help your sound is to keep your stage volumes at a reasonable level. In other words, don't get into a volume war with the other players in the band. I also recommend getting a talented sound person to work with your band, and you should learn how to take his or her advice. And, finally, always learn how to handle your own gear.

VAI: I believe that the sound is in your head and in your fingers. To exploit this inner talent, you should start out real simple and try to get the best sound you can from the ground up. Begin with just the guitar and an amplifier and work to make that combination sound great. Then, as you add sonic elements, make sure they are what you want. Be extremely picky. If something just isn't doing it for you, backtrack and find another device that will do exactly what you want.

It's always a good idea to try and stay on the cutting edge and get the very best quality you can within your budget constraints, because if you sound good, it helps you play better.

But ultimately, as long as you are writing good melodies and chord changes, the overall sound will not matter. So, don't ever forget that the song has to be there first! ◆

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Vox

By Michael Molenda

Choose the Perfect Vocal Mic

Worn a trash bag to work lately? Of course you haven't. The typical, rational human (we're excluding entertainers here) lives in abject fear of appearing ridiculous. So why will "serious" singers, who would no more wear a crab costume on stage than hurl themselves under a speeding dump truck, entrust their voices to whichever microphone a club provides? You see, there is no guarantee that the house microphone is a good match for your melodious (or beautifully tormented) vocal cords. A bad match can turn your voice into an obnoxious instrument of pain, stripping it of all texture and nuance. In other words, your voice just showed up to work in a trash bag!

Roger Daltrey has swung Shure SM58s around for decades. Sometimes he even sings through them.

PHOTO BY STEVE JENNINGS

If you place any value on your vocal tone at all, it is essential that you seek out a microphone that reproduces the timbre of your voice with the utmost care. The live-sound community has certainly adopted some "industry standard" microphones—that's why the venerable Shure SM58 shows up on concert stages all over the world—but there is no such thing as an industry-standard voice.

But let's say that your voice *does* sound marvelous on an SM58. That doesn't mean that the house 58 you may be singing through tonight will be in fighting trim. Last night, that same mic could have been miking a screaming Howler Monkey for some industrial act. You wanna put your lips on that? You get the idea.

The hippest (and healthiest) way to ensure that your vocal sound is consistently excellent is to find a mic that enhances your particular voice, buy it, and carry it to every gig. The ultimate personal vocal microphone, however, should possess more attributes than the ability to gently caress that beatific voice of yours. In addition to the all-important tonal issues, here are some other factors you should consider when shopping for a vocal microphone.

Presence Peak. You can count on the fact that most vocal mics have a presence peak where midrange frequencies are boosted to help the mic cut through a dense stage mix. Of course, different manufacturers have different ideas about

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where in the frequency spectrum this peak should occur. The trick is to make sure the presence peak doesn't fall in a spot that adversely affects your vocal timbre. For example, I have a rather nasal tone, and a mic with a prominent boost at around 3 kHz makes me sound offensively shrill. This is (hopefully) a major "duh," but it's critical that you audition a prospective mic to ensure that its peak doesn't mess with your tone.

Proximity Effect. The closer a mic gets to your mouth, the more chance that its sound can become distorted or boomy. A suitable mic should produce ample gain and clear, clean tone at a comfortable distance from your lips.

"Proximity control is everything in a vocal mic," says Bruce Forbes, vice president of product and market development for CAD. "When you're on stage, the microphone should deliver everything you need without forcing you to eat it. Along with the tonal problems associated with proximity effect, if you have to eat the mic to be heard, as soon as you move your mouth away from the mic—boom!—there goes your mix."

You can test how a mic handles "up close and personal" use by simply eating the mic and singing a line or two in your typical dynamic range. If the mic freaks, that's a bad sign because you *will* occasionally be tempted to eat the mic when performance adrenalin kicks in. A good mic should be able to handle those moments of madness without deep-sixing your vocal timbre. Then, sing with the mic approximately four to six inches from your mouth and make sure that the tone is smooth and that you can hear yourself loud and clear.

Off-axis rejection. If a microphone exhibits poor rejection of ambient sound, the rush of noise from all directions of the typical stage will most definitely invite torrents of feedback. In order to ensure that the mic can produce an adequate amount of signal gain before feedback occurs, it should reject the majority of sound leaking into the mic from monitors and other sources. And, in addition to tempting the feedback fates, signal leakage can cloud vocal tone.

"The mic should isolate the sound of the voice as much as possible," says Cliff Castle, vice president of sales for Audix. "You don't want guitars, drums, or other sounds from the stage bleeding into the mic and covering up the vocal tone."

You don't need to be on stage to check the rejection capabilities of a mic. Simply count into the mic while rotating it from front to back. You should hear a rather substantial dip in level when speaking into the rear of the mic.

Handling noise. At some point, even the most static performer will usually snatch the mic out of its stand and attempt some fancy prancing. And nothing can spoil that climatic moment more than a loud "ka-boom" when the mic slips into your fingertips. On less robust mics, handling vibrations may get transmitted to the capsule and produce rumbling or squeaking sounds. Not cool. To test for handling-noise levels, simply slap the mic against your hand while it's plugged in and powered up.

The real world. Ultimately, you should rent or borrow a mic to see how it actually performs on a gig.

"In a music store, a $50 mic and a $200 mic can often sound very close in quality," says Castle. "But you need to get out to see whether the mic can cut above a band without producing a harshness in the high end. It's onstage where the tonal and rejection qualities of a fine microphone will really shine." ◆

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By Greg Pedersen

Get Great Acoustic-Electric Tones

Club gigs can be brutal trials for acoustic guitarists. Miking a prized dreadnaught, for example, often invites a wash of ambient sound and feedback that can trash the most exquisite tone. But even acoustic-electric guitars—those hybrid designs that are supposed to allow "unplugged" performers an easy transition to the live stage—can be victimized by feedback. In addition, the pickups employed by most acoustic-electrics often produce thin, midrange-heavy timbres that, if processed or EQ'd improperly, can sound rather offensive. Happily, there are ways to ensure that your acoustic-electric sound is a thing of beauty.

Chanteuse Eva Jay Fortune goes for a shimmering tone by plugging her Ibanez AG600E into a Roland SE-70 multi-effects processor.

PHOTO BY MARK C. DAVIS

Go direct. If you're plugging your guitar directly into the club's sound system, the first thing you should do is make sure an impedance mismatch isn't butchering your tone. The piezo-electric pickups that amplify most acoustic-electrics output an unbalanced, high-impedance signal, and house mixing consoles typically work best with balanced, low-impedance signals. If you attempt to force these mismatched signals to coexist, the reward for your hard-headed resolve can be a thin and wimpy guitar tone. The solution? Plug your guitar into a direct box, the keyboardist and bassist's long-standing weapon of choice for live work.

"Many of the acoustic-electric players I deal with haven't heard of a direct box or aren't even aware that they really need one, but they do!" says Kenneth Boutelle, a 20-year veteran of live sound crews who has worked with artists as diverse as Leo Kottke and the Tubes. "Simply put, a direct box guarantees that the impedance loading is correct between the guitar and the mixer. This will ensure, especially if you're using an active DI, that the natural sound of the guitar gets to the console with all the instrument's gain and low-end intact."

Savvy performers should purchase their own direct box and carry it to each gig. Sometimes, depending on the fortunes of the club, a house sound person will have only one or two direct boxes available, and arguing with your bassist

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or keyboardist to determine who gets to use the box isn't the most serene way to prepare for a show.

Change amps. The tone of an acoustic-electric can also be compromised by plugging the instrument into a guitar amp. Regrettably, club performers who switch between electric and acoustic guitars during a performance tend to simply plug their acoustic-electrics into the same rig they use to amplify their Strats and Les Pauls. Most dedicated acoustic-electric players will also use a guitar amp on stage because it's often the easiest, most readily available option for amplifying themselves above the band. Unfortunately, a conventional guitar amp—which is typically designed to accentuate ripping midrange frequencies—is not the most empathetic partner for acoustic timbres.

"A guitar amp will give your instrument the frequency-response characteristics of an electric guitar, which the acoustic player definitely does *not* want," says Carl Countryman, chief engineer for Countryman and Associates and a longtime live sound mixer."Your acoustic-electric will sound better if you try feeding it through a keyboard amplifier. Keyboard amps deliver a broader and flatter frequency response than the average guitar rig does, so they produce a more natural sound that is better suited to the tone of the acoustic-electric."

"You'd never plug a vocal mic into a guitar amp, would you? And yet, tonally, acoustic guitars have a lot in common with voices," adds Gary Brawer, owner of Stringed Instrument Repair in San Francisco, whose clients include Joe Satriani, Counting Crows, and Journey. "An acoustic guitar requires more overall fidelity, more bottom end, and more high end than most guitar amps can deliver."

Work those tone knobs. Proper equalization is another way to avoid trashing your tone. Piezo-electric pickups can contribute to a thin sound because they tend to be very present in the 500 Hz to 1 kHz range. This characteristic not only boosts those problematic midrange frequencies, it can leave low frequencies sounding a tad anemic.

"Try cutting the midrange frequencies that are causing the guitar to sound thin by approximately five or six dB," suggests veteran concert sound mixer Jeff Fogerty. (And, yes, he's related to John Fogerty of Creedence Clearwater Revival fame.) "This tweak will allow the low frequencies to be more pronounced. I also recommend using a parametric, rather than a graphic, equalizer if you really want to do some tone shaping. For example, if your problem frequency is at 73 Hz and you need to boost or cut that frequency, a graphic EQ will not let you dial in at exactly 73 Hz because it operates on fixed frequencies. On a graphic EQ, you'd have to go to 60 Hz or 80 Hz and try to fix the problem from there. With a parametric EQ, however, you can sweep the frequency right to the point where the problem exists. You can kill that one tonal problem and leave the rest of the guitar's sound intact."

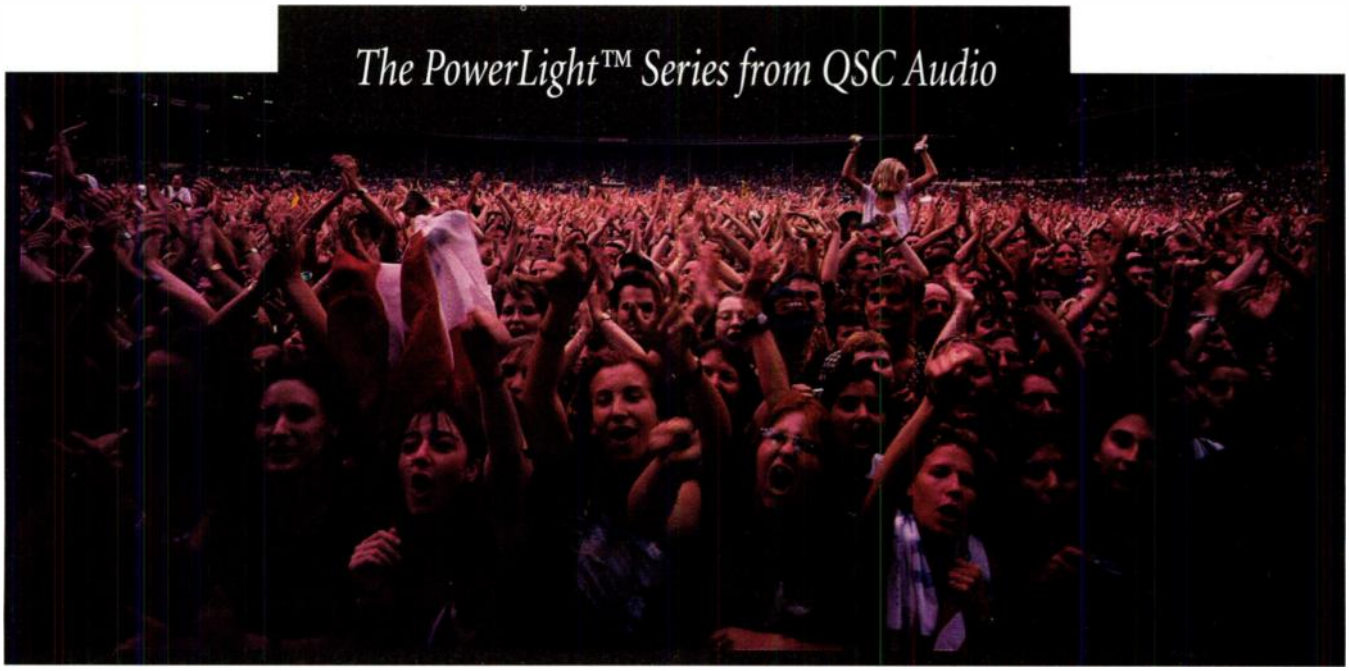
Fight feedback. Feedback is another sonic gremlin that often plagues acoustic-electric players. This problem is usually a tough one for club musicians becuase the stage is often a beehive of resonant frequencies that can cause the guitar to feedback practically nonstop. EQ tweaks and quick volume adjustments are sometimes not fast enough—or permanent enough—to keep you from dancing around the stage to find a "safe spot" where the squealing will stop.

"Perhaps the simplest way to combat feedback is to use a Kaman Feedback Buster," says Brawer. "It fits inside the guitar's sound hole and keeps air-flow and sound-pressure levels from causing problematic resonances."

Of course, feedback can occur at several different resonant frequencies, which can scotch the effectiveness of muffling options. In these instances, electronic feedback fighters should be called into the fray.

"The Sabine FBX-SOLO SL610 is great for eliminating feedback," says Boutelle. "It can be plugged into an insert jack on the mixing board, or you can run it between your guitar and your amplifier.

"If feedback pops up during your sound check," he continues, "the SOLO will seek out the first four feedback frequencies and remember them. Deep yet narrow filters are used to automatically attenuate problem frequencies. Then, as new feedback frequencies are apt to crop up during the evening's performance, you can program some of the filters to find those frequencies and block them. I typically use two filters to kill feedback at the sound check, and then I program two filters to seek out and prevent feedback during the show. That usually does the trick. But, hey, if you have more than four feedback frequencies going, you should probably be playing a different guitar!" ◆



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keys

BY LISA KLUBER

How to Hear Yourself On Stage

AS FAR AS MOST KEYBOARDISTS are concerned, the concert stage is a war zone. Volume gymnastics between performers (especially clueless guitar players who seem to turn up their amps every song), inadequate monitoring and amplification systems, and even the acoustics of the room, often make it difficult for keyboardists to hear themselves play.

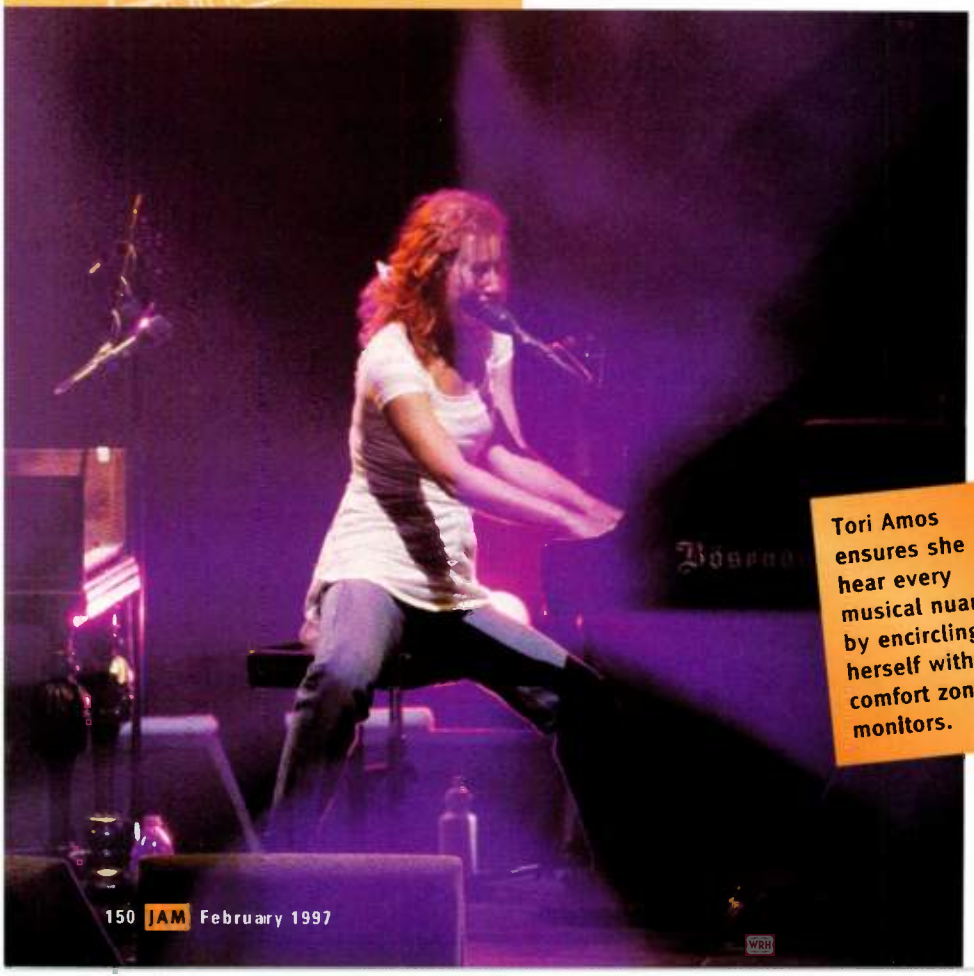
"Keyboardists are at a particular disadvantage hearing themselves onstage," says Vince Welnick, keyboardist for the Grateful Dead, the Tubes, and Todd Rundgren, who currently plays with Second Sight, "because what we require is a pretty large-sized amp and a big set of speakers—and those speakers must be able to handle the vast dynamic and frequency ranges of a keyboard. Otherwise the performer and the audience get treated to some very ugly distortion."

Despite the hassles, some keyboardists *do* cart large amps and speaker cabinets around. Harry Mello, a session keyboardist and piano man for modern rockabilly artist Dave Crimmen, rolls a monstrous, personal rack system to every club gig.

"I want control of my sound," says Mello. "I have a good amp with a lot of headroom, so I have enough power when I need it. I also use three speakers that are crossed-over so that one speaker handles the lows, one the mids, and one the very high frequencies. My system allows me to get a very clean and clear sound at almost any volume level, so I can blend my keyboards into practically any stage mix."

Adequate amplification, however, is just one remedy for the monitoring ills of the performing keyboardist. Although the situation often seems hopeless, there are a few other ways to ensure that you're not overwhelmed by the sonic power of your stage-bound compatriots.

Keep your "friends" close. "The great advantage of being a keyboard player is that you can keep your monitor right next to you," says Bob Bralove, keyboard technician for Stevie Wonder and sound director for the Grateful Dead. "Obviously, if you're not hearing yourself, the first inclination is to turn up the volume. But try to see what the



Tori Amos ensures she can hear every musical nuance by encircling herself within a comfort zone of monitors.

PHOTO BY STEVE JENNINGS

trouble is, first. You may be able to solve the problem by changing the position of your speakers."

Clean up your mix. If you're depending on a house monitor mix to hear what's going on, keep that mix as simple as possible. It should not be necessary to hear everything in order to feel comfortable on stage. In fact, throwing a dense, "sounds just like a record" music mix into the monitors can actually make it *more* difficult to hear all the things you're playing.

"What you should be looking for is an uncluttered mix that still gives you enough of a reference to let you rock a little bit," says Mark Alexander, who plays keys for Meat Loaf. "You may not hear every background vocal blend, but if you can hear the basic meat and potatoes of the mix, that should be all you need to relax and have yourself some fun up there."

Alexander is one of an increasing number of musicians who use ear monitors, instead of conventional wedges, to hear themselves on stage. As these systems also diminish ambient noise levels (because they are inserted into the ear canal, just like an ear plug), Alexander is less susceptible to stage-volume fluctuations from the other players. But although this affords Alexander the chance to have his keyboards blazing loud in his personal monitor mix, he prefers to be blended in with the other instruments.

"If I'm too loud in my monitors, I usually find myself playing ahead of the beat," he says. "I tend to groove better when I'm more tucked into the mix."

Find your space. Keyboards cover a lot of tonal ground, which can be a *bad* thing if you don't try and work with the myriad frequencies exploding from the typical bandstand. The full-spectrum sound that tickles your fancy in the recording studio—or at home—may clash with the other instruments until all you can hear is an indistinct, musical gumbo. If you want to hear yourself better, try to find a tone that pops you out of the mix.

"The piano and the synth can cover the very lows of the orchestra to the very highs, so a keyboardist can muddy up a stage mix real fast if he or she isn't careful," says Alexander. "Your low frequencies should not clash with the bass player's because all that low end will just clutter up the overall mix. Usually, I'll cut some of the bottom end from my piano sound and also boost some midrange frequencies. The resulting tone may not sound great on it's own, but in the context of the band's stage mix, it will probably sound just fine and it will also cut right through the instrumental wash.

"If you tweak your EQ right, you shouldn't have to keep raising your volume during the show," Alexander continues. "Don't be fooled by the great sound you get when you're sound checking all by yourself because when the band kicks in, that rich keyboard sound will get swallowed right up. Then, things can get crazy because if you can't hear yourself, you'll start turning up until the stage volume is totally out of control."

Stage volume, unfortunately, will probably continue to be the bane of club and concert keyboardists. After all, you can maximize your own monitoring situation, but it's extremely difficult to police the gain knobs of your bandmates.

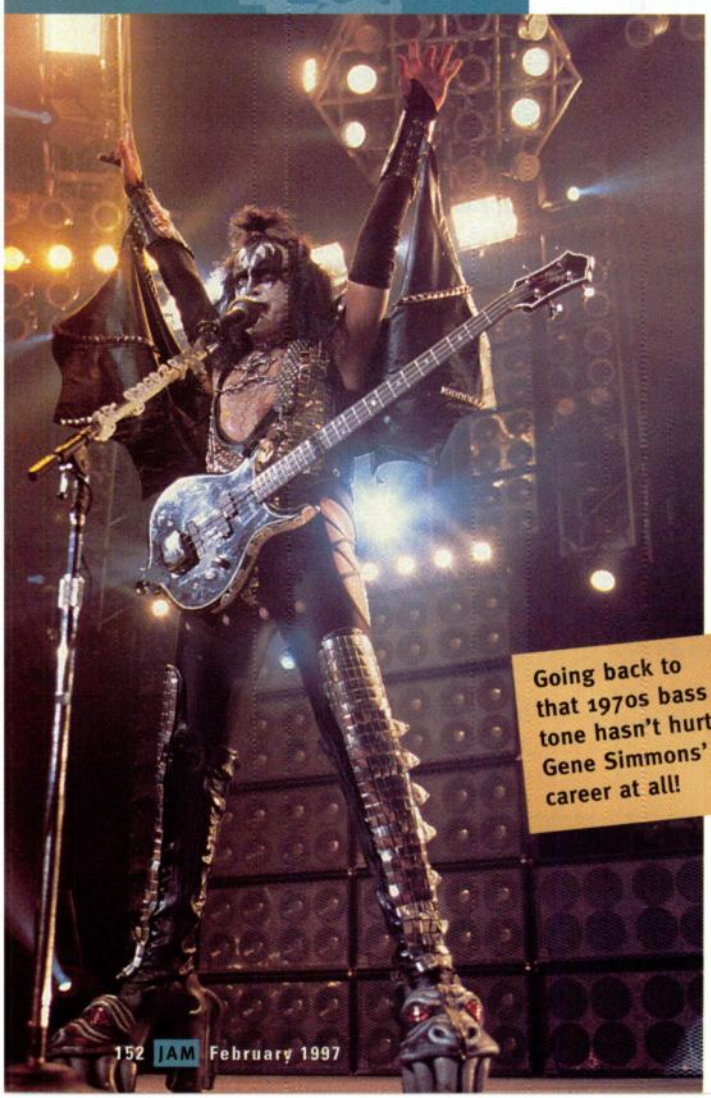
"Until someone builds the ideal personal keyboard amp, it will always be a challenge for a keyboardist to consistently hear what he or she is doing," laments Welnick, "because, currently, no amount of keyboard amplification can compete with a Stratocaster." ◆

bottom

By Brian A. Courtney

New Bass Amps with Vintage Boom

Bass players are such romantics. It seems we're always sitting around spilling our guts about how we miss the old days. Old friends, old loves, old cars, old basses, and old amps. Why this fascination with the past? Well, as far as the old amps and basses are concerned (I'll let others comment on the seductive nostalgia of past relationships and Detroit iron), it's all about sound. Old bass amps just tend to sound so hip! Much of this tonal affection has to do with the fact that most of these ancient amps were tube powered. Now, it's hardly news that tube amps sound wonderfully warm and fuzzy, but for bassists looking to pump out some fat bottom, that tube coloration is a real boon (or should I say "boom").



Going back to that 1970s bass tone hasn't hurt Gene Simmons' career at all!

Unfortunately, many old amps were relatively low wattage, which means that they are underpowered for the needs of today's gigging bass player. Believe it or not, those classic amplifiers actually produced their tremendous punch at low volumes. As a result, they still sound great in the recording studio where gain (from the amplifier itself) is not a concern. Onstage, however, the wimpy wattage makes it almost impossible to pump out a loud, rumbling tone. Bummer.

Thankfully, the trend towards retro sounds on records has pushed manufacturers to design many tubular (or similarly vintage sounding) delights for the studio and stage. Bassists can take advantage of this trend, too, as bass amps are now available that attempt to combine the best of the old with the best of the new. I recently auditioned three of these "new vintage" models: Fender's Rumble Bass ($1,999), Ampeg's reworking of the classic SVT ($1,750), and Crate's BX-220H ($500). Here's what these blasts from the past have to offer the hip, oh-so-modern bassist.

Fender Rumble Bass. Fender's Rumble Bass is the big cat daddy of the new vintage amps. The head looks like 1957 all over again with its camel-colored vinyl and large cream-colored, toaster-style knobs. (Also available are some groovy matching 4 x 10 and 1 x 18 speaker cabinets.) The Rumble Bass has power to burn, delivering up to 900 watts when its two input channels are linked in Combine mode.

What I liked most about the Rumble Bass is that it's easy to use. Each channel has Volume, Treble, Bass, and Mid knobs, and a normal/cut switch for the midrange frequencies. Because I'm not into channel switching or knob twisting, I just plugged into one channel, set all the knobs to twelve o'clock, put the mid switch on normal, and started playing. The sound was huge. Did I say "huge?" I should have said "the biggest sound you've heard in your life." If you are a volume freak in search of a great rock and roll bass amp, this is your ticket.

However, the amp doesn't sacrifice tone for pure volume. The Rumble Bass lives up to its name by giving you a tone

PHOTO BY STEVE JENNINGS

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that is big and warm, with some nice articulation on the mid frequencies. Think of John Paul Jones on stage with Led Zeppelin circa 1972. Admittedly the Rumble Bass isn't a funky, slap-and-pop type amp, but it wasn't designed to be. Fender has delivered a vintage-style amp that rocks. Period.

Ampeg SVT Classic Reissue. Ampeg is known for making great-sounding bass amps with classic styling, and the SVT Classic Reissue is another excellent amp in a long line of excellent amps. When auditioning this amp head, I used a Ampeg SVT215E cabinet that was covered in basic black vinyl with a sparkly blue, radiator-grille style front. The Classic Reissue has cool knobs, but what I liked most were the huge Power and Standby switches. These are wisely located on the front of the amp and produce a solid "click" when activated.

The tube amp is rated at 300 watts, which seems conservative because it had plenty of power to produce great tone at extreme volume levels. There was, however, a tendency towards audible distortion when the gain was pushed toward the upper limits. As a nod to modern times, High and Low input jacks are provided to service passive and active basses.

The Classic Reissue produces a wider variety of tones than the Rumble Bass, due to its 5-position midrange switch. I was able to get both a good funk tone with a nice high-end click and a warm, dark tone that would be perfect for a blues player.

Crate BX-220H. I'm not sure Crate would think of this amp as a new vintage design, but nonetheless, the BX-220H has a fun, 1960s look. Even its name—BX-220H—has a kind of *Lost in Space* ring to it. The faceplate has a gray-matte finish that evokes a circa 1967 "high tech" look, and the controls remind me of the radio knobs on an old Ford Falcon. I auditioned this head with a generic cabinet loaded with one 10-inch and one 15-inch speaker.

The BX-220H proves that you can get a pretty warm sound using solid-state electronics. The amp tone control is a graphic EQ that is effective and easy to use. Gain *and* Master volume knobs provide for a number of gain stages and can produce massive sound levels. I liked the fact that Crate put all the controls on the front of the amp. (I hate reaching behind an amp, on stage and in the dark, to make adjustments.)

Having never played a Crate amp, I must admit that I was a little surprised at the number of usable tones I discovered. The BX-220H produces a thinner sound than the way more expensive Rumble Bass and SVT Classic Reissue, but the slightly reduced low-end response can be an advantage for certain styles of music. For example, the BX-220H is a great amp if you're paddling into the surf music revival. The amps twangy quality is perfect for most instrumental styles, in fact. (Even that wacky cocktail music revolution!)

However, the hippest thing about the BX-220H is that it delivers its vintage tone at a relatively vintage price. At a list price of $500, this amp is a no-brainer for retro fiends on tight budgets.

Old souls with new hearts. New vintage amps such as the Rumble Bass, SVT Classic Reissue, and BX-220H are great options for bassists who want a retro tone but also want modern conveniences such as adequate power and enhanced tone controls. Now, if we could each find a completely original, blond 1962 J-Bass to go along with these amps, that would be truly wonderful! ◆

bang

By Scott Mathews

Defending Your Drum Sound

The hardest working musicians in show biz are drummers. There's simply no contest. We sweat it out rehearsing with our bands until we are blue in the face and deeply red in the blood blisters. And don't forget the king's ransom we spend on gear and supplies to ensure that we (and those we are propping up) will sound great when the band hits the planks.

Multipercussionist Joel Davel tunes up before an "unmiked" gig in the tiny upstairs room at San Francisco's Paradise Lounge.

PHOTO BY MARK C. DAVIS

Then, at the gig, the fruit of all our efforts is usually dropped into the hands of a house mixer who has the power to butcher our sound with a few careless tweaks at the console. Let's face it: unless you can afford to bring a personal sound crew along to every gig (not likely!), you're bound to run into master mixers *and* tin-eared sound clowns.

From club to club, the only sure defense against a disastrous drum sound is to take matters into your own hands. If you can make sure that your drums sound good acoustically, they'll probably sound just fine through the house system—even if a bozo is at the board. Here are some techniques that have saved *my* stage sound more times than I wish to recall.

Get in tune. First off, it is critically important to tune your drums well. There is little or no hope of getting a drum kit to rage majestically through a sound system until the drums are in tune and acoustically balanced. In addition, every drummer should strive for a "signature" sound that acts as a sonic calling card, whether he or she is performing unmiked in a tiny club or bashing it out for stadium crowds. This, of course, is easier said than done because a great deal of experimentation is the key to finding that individual sound. Use different drums and drum heads and audition various tunings until you discover something that is distinctly yours.

Get your kicks in. For a tight kick drum that has plenty of attack yet retains enough low end to whomp the crowd in their collective gut, cut an 8- to 10-inch circle into the front of a thick drum head. Inside the drum, a pillow (or some other absorptive material such as a

foam pad or a blanket) can be positioned to diminish resonance and increase punch. Now use a felt beater and lay into it with that foot like you mean it!

For a fuller tone and some rumbling subsonics, use a thin front head that is intact (*sans* hole). Make sure there are no ripples in the front and rear heads: these will flatten the sound and cause the drum to lose all the low end you're trying to gain. To take this approach even further, you can produce a very long, resonant note by removing all muffling materials from the drum. This is an extreme move that is not meant for the meek, but you'll be rewarded with a low-frequency boom that's as big as a house. Have mercy!

"Snare" apparent. It's a good idea to bring a variety of snare drums to the gig because you never know how a particular drum will sound in the room. As far as heads go, I find that a medium-to-thick coated batter head and a thin, clear bottom head works well. Keep a drum key nearby: the tension will become looser as you bash away throughout the show.

Resounding toms. Using both heads on each tom produces a mighty wallop. Medium-to-thick coated skins on top and thin coated skins down under typically deliver the most robust sound. Head tension should be like the tension between the lead guitarist and the singer—in other words, *plenty*.

Painless cymbals. Obviously, how you place your cymbals will determine how much they splatter the overall drum mix. If possible, keep these piercing discs of destruction away from tom mics. Signal bleed can send unmerciful amounts of high-end through the sound system.

Mobile mic cabinet. I strongly advise bringing a small, personal mic collection—which should be considered an integral component of your live sound—to every gig. This is a somewhat pricey move, but compared to what you have already invested in your kit, the cost of a few mics is well worth the quality control it will provide. A personal mic stash will keep you from worrying about unmatched house mics (an SM58 on this tom, a MD 421 on that tom, etc.) savaging your finely tuned drums.

I typically carry some tiny, clip-on AKG C 408s for my toms, a trusty Shure SM57 for the snare, and an AKG D 112 to front the mighty tiki god painted on my kick-drum head. Obviously, I want to ensure that no soundperson messes with the tiki's thunder. ◆

tech

By Meg Ryan

Survive Your PC's Stage Fright

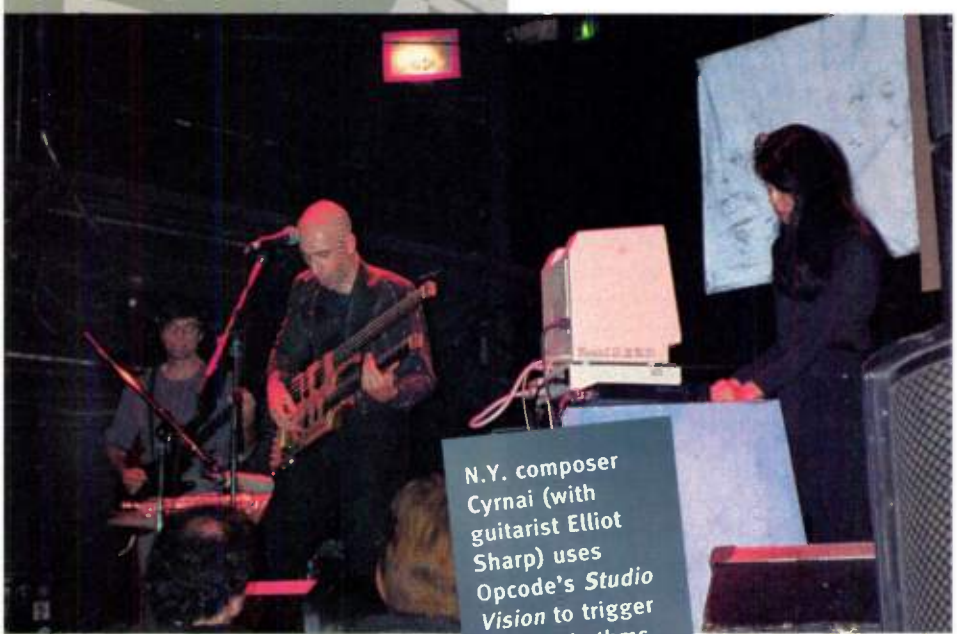
So you want to ask your Mac to join the band, huh? Well, computers can be extremely unreliable, and they can even have the sorry lack of manners to crash in the middle of a performance, causing you and your bandmates to consider reforming as an acoustic act. However, a little common sense and extra preparation can prevent a host of digital disasters. Here are six quick tips to help you get your music safely out of cyberspace and into a performance space.

Know your system. Familiarity with your system is absolutely critical. Get extremely intimate with your programs. Sleep with them if you have to. Even more important is that you understand how your programs and equipment interact.

"In the relative safety of your home, a computer may be a tool for creating music, but once it's on stage, it becomes an instrument, just like any other," says Jonathan Hoffberg, a guitarist and composer based in Berkeley, California, who uses Digidesign's Pro Tools and *Sound Designer* for sequencing and Opcode's *MAX* to program lighting and sound for his live performances. "A computer-using performer should be just as familiar with his or her equipment as a guitarist or keyboardist."

"Design for the best; prepare for the worst." That's the motto of Jeff Forehan, a San Francisco percussionist and vocalist who has toured as a keyboard technician for acts such as the artist formerly known as Prince and Bananarama. Forehan advises the computer-using performer to have many, many technical rehearsals. These tech run-throughs are essential for putting on a good show. Now, all the planning in the world will not guarantee a flawless performance, but the tech rehearsal is a great opportunity for fleshing out potential problems *before* you hit the stage.

A productive tech rehearsal involves running through the show step by step from beginning to end. Check and double-check your MIDI Program Change mapping to ensure that every program comes up correctly for each song and make sure your continuous controllers are programmed to taste. If you're using samples as a major component of your show, make sure all the samples are loaded in the right program locations in your sampler so there are no surprises when you're in the middle of a performance. Obviously, you should have done most of this programming at home, but it never hurts to confirm that everything works according to plan when the full band is rehearsing an actual set.



N.Y. composer Cyrnai (with guitarist Elliot Sharp) uses Opcode's *Studio Vision* to trigger techno rhythms.

PHOTO BY TIM NIEMI

You should also figure out how long it takes to reboot your equipment and reinitialize all your programs and data in case you ever have a catastrophic computer crash. (You also may want to have your guitarist prepare a few solo numbers—or convince your vocalist to try his or her hand at stand-up comedy—to cover those awful "dead" moments when you're frantically trying to revive a crashed system.)

In preparation for the inevitable, you should design your system and data flow with the intention of minimizing reinitializing and data-retrieval time. Reinitializing should take no more than one minute when you're prepared, so simulate a crash or two during rehearsals so you can practice the procedure.

Finally, documentation is an essential, often-forgotten component of the tech rehearsal. Label and document everything from signal routing to software versions. When Forehan was on the road with Prince, he kept a notebook that included a detailed physical diagram of the keyboard and computer setups, a map of all cable connections, and an inventory of the program libraries for every sampler and sound module—for all his show setups.

Avoid on-the-spot editing. Making a programming change on the fly without thorough rehearsal is an invitation to disaster because changing sequences can sometimes trigger a cascade of sequencing problems. If you are suddenly hit with a "brilliant" idea in the middle of a set, save it for the next show.

Go for stability. Because data and hardware can be fragile and temperamental, providing a stable environment for your equipment can often help you avert tragedy. Computers, after all, are testy little critters and can succumb easily to club environments rife with smoke, dust, and intense heat. They also do not take kindly to the trauma of being knocked about on the road.

A laptop, because it's so portable, typically works easier for live performance transport and set up than a desktop CPU. However, if you use a desktop CPU, it should always be housed in a rack and have plenty of room to breathe. A good CPU rack should provide enough rear air ventilation for the computer to breathe and offer enough space for a small cooling fan to be mounted.

Although rack-mounting a computer system is often a DIY proposition, there *are* some manufacturers who produce rack-mounted PCs. You'll have to do some searching to find these companies, as they often cater specifically to the music community. (Of course, we'll attempt to keep **EM** readers informed of any such products we discover.)

Forehan, however, often circumvents the "CPU cartage" problem entirely. He uses an old Alesis DataDisk, a rack-mountable MIDI file player that lets him play back sequences live without having to schlep around a CPU.

Be redundant. Always plan several different ways to do the same thing, just in case your primary resource fails. Like a guitarist with extra strings handy or a wind player with a spare box of reeds, the savvy computer musician should always carry two of everything: data disks, CPUs, drives, and samplers. And everything should be programmed and ready to go in the event of a catastrophic crash.

If you can't afford to duplicate your entire stage setup, at least have copies of the most fragile elements, such as data. You just can't be too careful. In fact, when Forehan was the keyboard technician for Prince's *Love Sexy* album and tour, the technical crew had a "grandfather-father-son" backup system, in which there were *three* copies of everything. "Even if one doesn't have the financial resources to have two samplers available, one should always be sure to have the resources for three safety copies of data."

Tell the time. If you use sequences as a component of your live performances and you expect your musicians to play along with them, your drummer will need some type of click track. Forehan had to learn this lesson the hard way when he was keyboard technician for Bananarama's world tour in 1989. The band played to sequences, but there were no click tracks to cue the drummer. As a result, the band often carreened into musical train wrecks during big concerts in New York, London, and Detroit. Needless to say, these moments were huge embarrassments. The moral of the story is, technology means very little if the band can't groove.

Practice makes perfect. The real key to a relatively problem-free performance of computer-based—or computer assisted—music lies in preproduction.

"Nothing feels worse than being unprepared when old Murphy rears his ugly head," says Forehan. "But nothing feels better than being ready for him and kicking his little butt." ◆

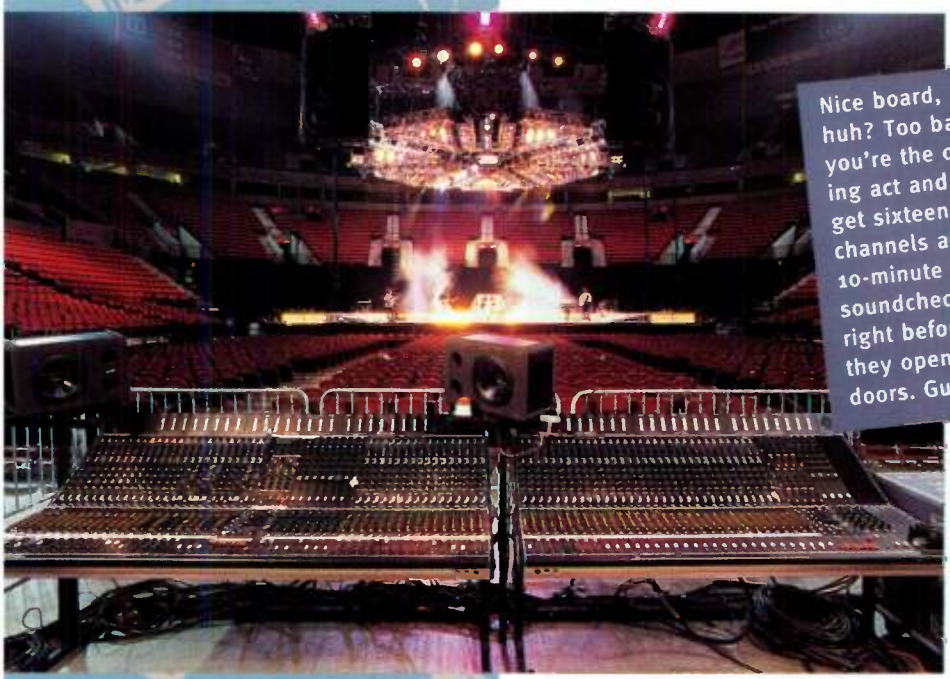
house

By Brett Ratner

Performing a 15-Minute Panic Mix

It's the most important gig in the band's career. A&R dudes have come in droves, congregating to form a herd of power ties and Italian shoes. Dozens of acts will showcase at this musical equivalent of a Hollywood cattle call, and the aforementioned band's set is wedged between some thrasher/punk outfit and yet another Nirvana clone. The pressure is on, especially on you: the sound person.

You've got fifteen whole minutes to make these musicians sound like future rock stars, in hopes that they can join the ranks of those privileged few with the luxury of a *real* sound check. It won't be easy. Unlike the other bands, this group has two backing vocalists and a keyboard player, not to mention a lead singer toting an acoustic guitar that is prone to feedback. What do you do?



Nice board, huh? Too bad you're the opening act and only get sixteen channels and a 10-minute soundcheck—right before they open the doors. Gulp!

PHOTO BY STEVE JENNINGS

"The first thing, most important above all else, is line check, line check, line check," says Jonathan Laster, sound engineer at Nashville's Tennessee Performing Arts Center. "If you've only got fifteen minutes to sound check, you've got to know that you have signal coming from the stage before you have the luxury of worrying about anything else."

Considering that a line check will eat up at least five minutes before it's time to play (and that's only if you don't discover any problems), you're dead in the water right? Wrong. Even if the performance has started, you can still ensure that the band sounds okay.

"First, make sure that your vocals are hot," says Laster. "That's the most important thing out of the gate. Most clubs are small enough that the stage volume will carry you for a little while, so you can afford to let the instrumental mix take second priority. Once you are sure the vocals are happening, then you can start mixing in some foundation—the kick drum and the bass guitar. When you have that going, just work in the other elements of the music in and around the vocals."

So how do you make sure the vocals sound good three measures into the opening tune? "Often, the vocalist is going to be singing through a Shure SM58 because it's an industry standard," explains Laster. "The SM58 has a presence peak that accentuates the midrange frequencies, and that's nice because you can just throw the fader up and know the vocal sound will be acceptable.

"A 'ready to go' sound can be a lifesaver because live mixing throws so many variables at you that you can't depend

on stock EQ settings to help you out," he continues. "The sound you hear depends upon the artist, the house sound system, and the room, so everything changes from show to show. To be safe, I generally approach EQ adjustments from a subtractive standpoint. I take out stuff I don't like. Then I'm left, theoretically, with what works."

According to Neil Fagan, sound engineer at Nashville's legendary Bluebird Cafe, a lot of the tones you wind up subtracting reside in the middle frequencies. "I like to have a warmth on the low end and a crispness on the high end," he says. "The midrange is always everyone's trouble spot. It's the frequency range that gives you the most perceived volume, but it's also the range that produces the most honkiness."

Regarding the potentially problematic acoustic guitar in our showcase band, Fagan recommends zeroing in on those troublesome mid frequencies to ensure a quick tonal fix. "The first thing I'll do with them, if they'll let me, is completely pull the mids right out of the guitar sound," says Fagan. "I'm not a purist about acoustic guitars. For live sound, I don't go for that pure, organic sound. I like to have some thump on the low end and some sheen on the top. Obviously, the first thing I'll do is listen. I won't immediately do anything to anybody until I've heard them play their instrument."

Like Laster, Jack Irwin, the sound engineer at one of Nashville's busiest clubs, 12th & Porter, focuses on the vocal first. "The people coming to see these shows are not really coming to hear a lead guitar or a cool reverb on the snare," says Irwin says. "They're coming to hear the singer and the songs. So I try to keep that in perspective and try to keep the vocals front and center."

Irwin is a master at getting results from a short sound check, and has developed a method to build a good mix around the vocals—*fast*. "I've gotten in the habit of quickly adjusting the house EQ to the singer or vocalists who are performing that show," he says. "I'll usually get the band up there and get the instrument levels in the ballpark without doing any EQ adjustments. Then, I'll have the vocalist sing a cappella. If there are background singers, I'll have them all sing, too. I'll very quickly adjust the EQ according to the tonal idiosyncrasies I'm hearing in the vocals. This means I will notch out anything that sounds like it's booming out there in the club or that sounds strident. Usually, if I can get the vocals set, and if the band is able to maintain a reasonable stage volume, all I have to do is bring the faders up on the rest of the instruments, and the general mix is done."

Only after the band is actually playing does Irwin concentrate on the tone of individual instruments. "I'll solo each instrument in my headphones, leaving one earpiece off so I can still hear the sound of the instrument in the room," he says. "Usually I'll sweep a low mid or mid frequency until I find something that sticks out as unpleasant or something that might be causing low-end rumble. If I discover a problem frequency, I just notch it out real quick."

To save time—especially when you don't have any time to save—Irwin recommends having a consistent setup at the mixing board. "What I do to save time is to leave all of the mixer channels in a basic, default setting," he explains. "I always bring up the drums, the bass, and the rest of the instruments on certain channels. The vocals always get routed to the same compressors and are grouped in a logical fashion. This arrangement cuts down my set up time considerably."

Irwin, Fagan, and Laster all agree that when attempting a brutally quick mix, effects (such as reverb, delay, etc.) should be considered a very low priority. They also agree that the biggest time-saver of all is letting the person who knows the room do his or her job.

"The thing that really slows down a sound check is when bands bring their own sound person, and instead of working with the standard house setup, he or she flatlines all the monitor EQ settings and the house EQ settings and starts from scratch," says Irwin. "Because I know the room, the house sound system is like my home stereo. It literally takes me as much time to get a band up and going as it does to play a CD at home. And I'm not saying this because I think I'm such a great and wonderful sound person, I'm saying it because I'm working in that room every day.

"If someone brings in a supposed 'hot shot' engineer," he continues, "and he or she feels the standardized setup just isn't good enough for them, *that's* when a 15-minute sound check can turn into a 2-hour sound check." ◆

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SERVICE CLINIC

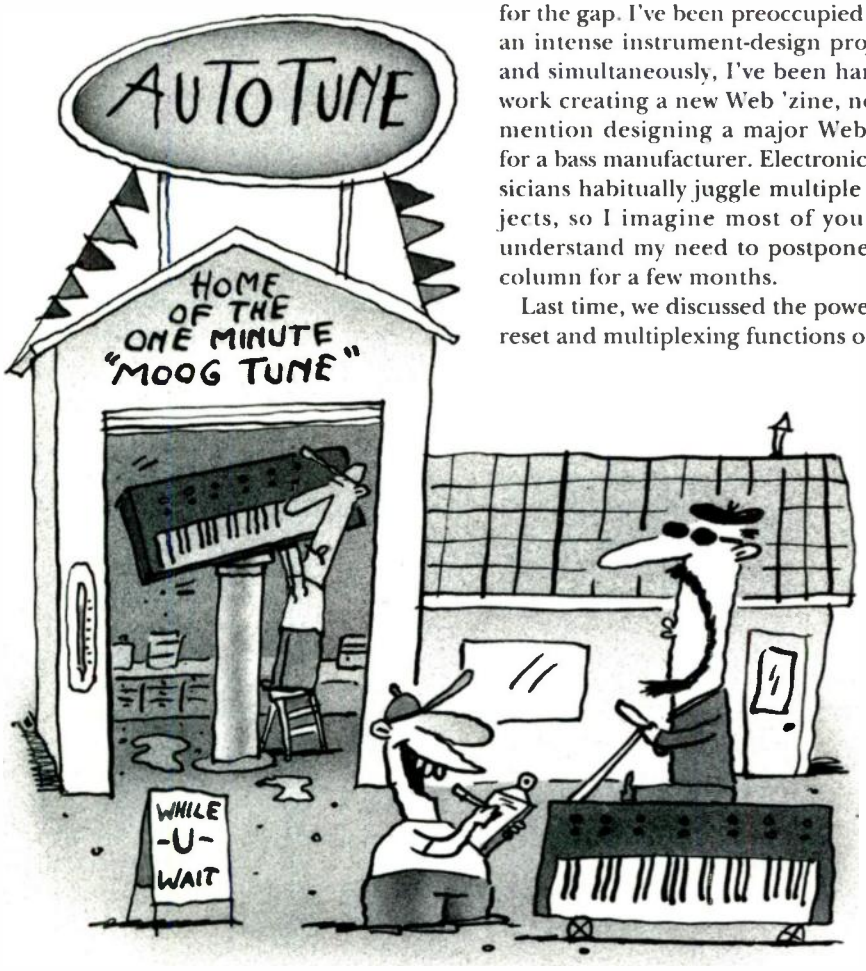
Memorymoog & Hybrid Design

After a brief hiatus, the synth doctor is back with a vengeance.

By Alan Gary Campbell

As you probably noticed, it has been a while since "Service Clinic" has appeared in these pages (November 1996 **EM**). I apologize for the gap. I've been preoccupied with an intense instrument-design project, and simultaneously, I've been hard at work creating a new Web 'zine, not to mention designing a major Web site for a bass manufacturer. Electronic musicians habitually juggle multiple projects, so I imagine most of you will understand my need to postpone the column for a few months.

Last time, we discussed the power-up reset and multiplexing functions of the hybrid, polyphonic synth. This month, we take it from there.



JACK DESROCHER

PANEL-CONTROL DETECTION

After the hybrid synth's power-up cycle is completed, the panel-function latches and sample/hold circuits are loaded with the default program data. Typically this is the data for Program 1 or Program 0; most hybrid instruments don't "remember" the final edit condition at power-down.

Before any edits to the current program can be detected and sent, however, a referent for the initial control positions must be established. Manufacturers have employed various schemes to provide a referent for each panel-control value and to detect panel-control changes. Because the Moog Memorymoog remains the vintage polyphonic synth about which **EM** readers most frequently inquire, it seems reasonable to continue to use it as an example.

In the Memorymoog, the values for the panel-control positions at power-up are stored in a table (labeled "POT-POS" in the object code). The panel controls are continuously scanned and decoded, and the output values are compared to the table. When a change in value is detected, the new data is sent to update the affected sample/hold. In fact, when you perform a motion as simple as rotating the filter-cutoff pot to tweak the brightness of a patch, you've

actually updated the table value dozens of times! All of this happens so quickly, though, that the process seems smooth and continuous.

It may be instructive to compare this scheme with the less intuitive approach of the Sequential Circuits Prophet 5. To detect and update the table values in this earlier design, the user has to take a panel-control pot to its maximum setting and then back off to the desired edit position.

Interestingly, the Memorymoog uses a simple, analog comparator as the basis of the detection circuit. In this scheme, the instantaneous table value is sent to the system D/A converter, and the converter output yields an appropriately scaled voltage that drives one input of the comparator. Each panel potentiometer is wired as a voltage divider based on the reference voltage used by the D/A converter.

The wiper output of the pot associated with the table value drives the other input of the comparator. If there is a difference in the two voltages—in other words, if the scanned panel-pot position is different from the position value stored in the table—the comparator generates a high output. This signals the circuit to send the pot voltage to the A/D converter and to update the table value and the associated sample/hold.

The comparator employs some hysteresis to eliminate spurious editing caused by component tolerance variations, pot "jitter," and mechanical vibration. (The "self-edit" problems that plague early Memorymoogs are caused by a PC-board layout error, not by a poor detection circuit.)

Of course, the A/D and D/A converters and the detection circuit are multiplexed and demultiplexed among the buffered pot outputs and the associated sample/hold inputs. This process occurs very rapidly and is intended to give the user the impression of continuous control. In practice, however, the Memorymoog's 8-bit Z80 microprocessor—which must also scan the keyboard and control numerous other functions—is heavily taxed. Some of the multiplexing is evident as LED flicker and a slight keyboard-response delay.

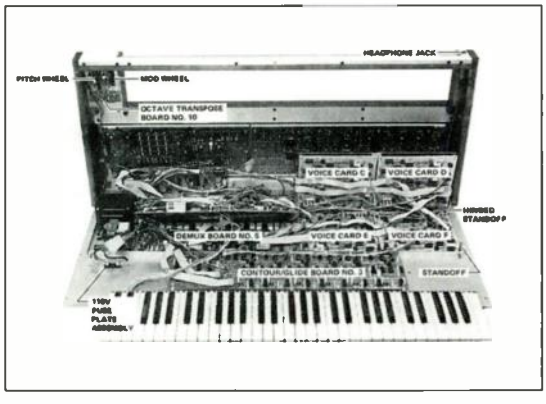
Memorymoogs with the standard Moog MIDI retrofit truly overtax the Z80, and the keyboard response is somnambulistic. That's why many players prefer the non-MIDI version for real-time performance.

THE AUTOTUNE FUNCTION

The Memorymoog also provides an excellent example of the requirements for and limitations of a hybrid Autotune function. Each Memorymoog voice board is roughly equivalent to a Minimoog in miniature, with three oscillators per voice. In a 6-voice instrument, that's eighteen oscillators to keep in tune at one time. So a well-behaved Autotune is essential.

The Autotune function in early Memorymoogs is a boon and a bane. It uses a simple, iterative scheme based on a CTC chip and a flip-flop. Each oscillator is calibrated in sequence by timing the oscillator period over a given range and then updating the oscillator tuning offset via the D/A converter. This is a fairly rapid Autotune scheme, but one that leaves the instrument somewhat prone to drift. These slight inaccuracies give the early Memorymoog a distinctly organic sound quality, but unfortunately, they also make it hard to predict just how in or out of tune the instrument will be at any moment. Learning how long to let the early Memorymoog warm up and how often to press the Autotune button are performance techniques as important as scales and trills. (Gross tuning drift in early Memorymoogs is the result of minor manufacturing defects. More on this shortly.)

Later, Moog Music updated the Autotune function at both the hardware and software level using a robust scheme that tames the tuning to an impressive degree. Unfortunately, this Autotune Update feature tunes the instrument so accurately that the organic quality is diminished. (The Autotune Update is required on units with the factory MIDI kit.) I left the old Autotune routine in my Memorymoog and put up with the tuning vagaries.



The Moog Memorymoog. The upper cab is open in service position, prior to calibration.

CALIBRATION

By now, readers who have been following this series will no doubt have an impression of how complex hybrid synths can be. With so many variable components, both analog and digital, it follows that hybrid-synth calibration is both critical and somewhat daunting.

Fortunately, as this class of instruments evolved, its internal software diagnostics evolved, as well. The Memorymoog is fairly advanced in this respect. To calibrate the oscillators, for example, an internal diagnostic routine can simultaneously display an oscillator's hexadecimal Range, Scale, and Hi Frequency values. Theoretically, calibration is as simple as bringing each hex value as close as possible to midrange, or "7F," via the appropriate trim pot.

In practice, the relative difficulty depends on component aging, the state of repair of the instrument in question, the temperature, the humidity, and the acrobatic skills of the technician. I look back with a mixture of humor and horror on the countless times I have found myself balanced precariously on the top step of a stool as I leaned over an open Memorymoog and tried to see the display while I tweaked the last voice card.

Do-it-yourselfers who are considering calibrating a hybrid synth should be forewarned that a 4.5-digit, 0.05 percent accuracy DVM and a high-quality strobe tuner (or its modern equivalent) are absolutely required. It is mandatory to check and, if necessary, adjust the power supply and converter reference voltages before attempting any other calibration. If you try to make these adjustments with a common, 3.5-digit DVM, you'll get the synth so far out of calibration that it will be nearly unplayable. Don't even *think* about it. Moreover, if you attempt calibration without access to complete service documentation that you have studied *exhaustively*, you are just plain nuts.

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MEMORYMOOG MISCELLANY

Numerous readers have inquired about various aspects of Memorymoog repair and service, and it seems reasonable to address some of them here. Most important to the player who wants to restore an early Memorymoog is information on the various updates that are required to make the thing stable. Many of these affect tuning and some keep the unit from working at all. Most engineering changes are described in Factory Service Bulletins published by Moog-Norlin. (A few are not, however. I discovered a second layout error on the RSC board that contributes to self-edit problems, and although this fix was known among techs, to my knowledge it was never published.)

Unfortunately, the documents themselves exist in various revisions, which complicates matters. On the positive side, most of the changes can be accomplished with common parts, and I am determined to get this information on the Web in some form as soon as I can find out who holds the copyrights to the Bulletins.

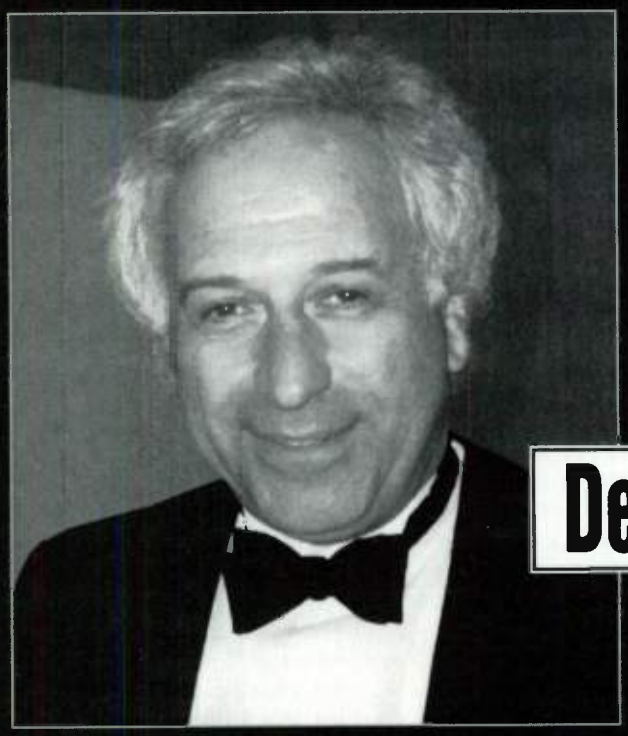
Several readers wanted to know whether the 3-volt lithium backup battery, which is hard to obtain, can be replaced with conventional batteries. I have seen this done with two alkaline AA cells (Duracell MN1500 or equivalent), wired in series in a small, 2-cell holder. This is a workable solution, but it is unlikely that the alkaline batteries will last more than a year or so in this application. Moreover, they should be replaced at eighteen months (maximum) to preclude leakage.

If you take this approach, make certain that the battery-holder leads do not inadvertently contact other components, especially when the instrument case is closed. Obviously, with this battery scheme you should be even more vigilant with regard to program-backup on tape.

Finally, it's a little-known characteristic, but Memorymoogs can easily transfer program data to one another directly via the cassette jacks without going to tape first.

EM *Contributing Editor* **Alan Gary Campbell** *is the publisher and editor of the* New Music Journal *and the owner of Musitech, a consulting firm specializing in electronic music product support.*

Legendary Composers



In addition to receiving an Emmy nomination for scoring all of the episodes of NBC's "Fame" series, William Goldstein has received numerous Emmy nominations and awards for his film and concert scores. So what composition software does he use? JAMMER of course. JAMMER instantly creates rock solid drums, powerful bass lines and beautiful rhythm tracks, all based on the chords you type in. It's like having a full band and orchestra ready to jam with you 24 hours a day.

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Reviews

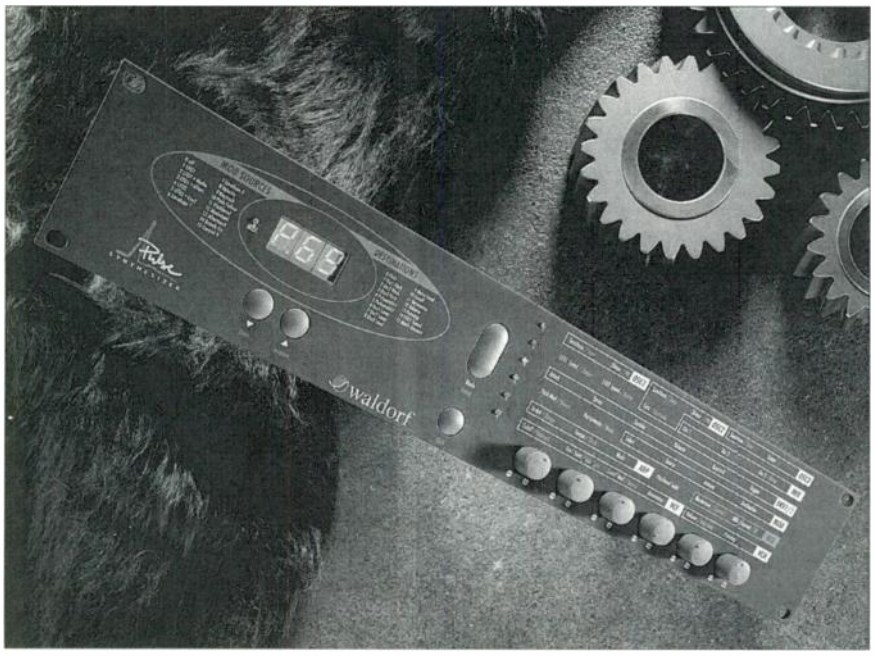
Waldorf Pulse

By Bill Heagy

Analog-synth lovers of the world rejoice!

Curiously space-age in appearance, the Waldorf Pulse analog synthesizer is a definite change from the massive array of knobs, wood, and patch cords that is typical of dinosaur analog synths. In fact, the Pulse's modern appearance comes as a mild shock to those who have grown accustomed to massive Moogs, spaghetti-draped ARP 2600s, and various other larger-than-life analog behemoths.

Rest assured, however, that this synth packs a full day's supply of magical, fat sounds and shimmering filter sweeps.



Waldorf scored big with the Pulse analog synth module, which offers superb sound, complete MIDI control, and wickedly fast envelopes.

Yet the Pulse is indeed a modern instrument with a solid MIDI implementation, reliable performance, and plenty of programmable parameters.

A MODERN FACE

Flat blue gray with muted yellow, white, and blue accents, the Pulse is just two rack spaces high and only a few inches deep. A glowing, moderate-sized, 3-character, 7-segment LED display reveals patch numbers and parameter values. Programs are recalled using a pair of scrolling increment/decrement keys that double as Store and Compare buttons.

Direct parameter control is implemented in an interesting way. A 6-line × 6-function matrix of parameter names is printed on the front panel. You select one of the six lines by repeatedly pressing a flat, pill-shaped Mode switch that doubles as a trigger for SysEx bulk dumps. A vertical row of six LEDs to the left of the matrix indicates which line of functions is currently selected. The knobs directly beneath the matrix control the parameters identified in the selected row. A Shift key accesses alternate functions for the knobs, which are marked in orange on the matrix.

The user interface is a logical affair. The knobs and switches have a solid, smooth feel, and the quality of the Pulse's construction in general instills confidence. Props to the people who figured out how to cram all that solid-state circuitry into such a small, neat package.

My main problem with the Pulse's interface is its display. Yes, I know it's a simple, monophonic synth, so it's fair to ask how much display area and detail you really need. But a 3-digit display isn't much no matter how you slice it, and although the row of knobs and printed matrix above them are well thought out, more visual information would be a definite plus. Otherwise, I want a knob for every single parameter in the box. The architecture of the Pulse is fairly traditional and straightforward, though, so the small display is not a fatal flaw.

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The rear panel is about as simple as you could want. It has a stereo pair of audio outputs on unbalanced ¼-inch connectors; MIDI In, Out, and Thru ports; and a socket for the external 12 VDC power supply. The expanded Pulse Plus version (see the sidebar "A Major Plus") also sports a control-voltage (CV) input, two CV outputs, a gate input, a gate output, and an audio input, all on unbalanced ¼-inch jacks.

BLESSED MEMORY

The Pulse has 99 memory locations, the first 40 of which are user-programmable. This seems a little stingy to me; program RAM is inexpensive, and I'd like to have a lot more of it. Fortunately, you can save patches via SysEx dumps.

"Range" seems the appropriate word for the factory presets. There aren't many usable stock basses or groovy lead lines here, but you do get an awful lot of programs that demonstrate that monophonic synths don't have to be boring. Hold down a note, and a gnarly, pulsing filter is likely to remind you how nice it can be to play with harmonics even if you aren't a guitar player.

Dirty, overdriven drones; pure, reedy tones; and soupy portamento extensions are among the offerings dished up from the Pulse's preset memory bank. Clanging, Gothic constructs; arpeggiated space bleeps; and a few analog drum sounds provide additional points of departure. Some of these sounds are complex enough to remind me of modular synthesizers; others make me think of an old ARP or even a metallic PPG.

There are a few more traditional sounds, such as a punchy analog bass or two; a Parliament-style, stinky funk lead; and a nice clavinet that still needs a little tweaking. But by and large, this box begs you to read the manual and program your own library of sounds. Don't worry if you need those old Roland and Minimoog bass sounds; this synth can come quite close to them. If you do get stuck, a nice random patch generator lets you unfreeze your "programmer's block" and get back to business.

MAKING WAVES

The guts of the Pulse include everything I expected and more. The unit's master tuning can be globally set between 430 and 450 Hz, with the default set at 440 (MIDI note A3). The signal

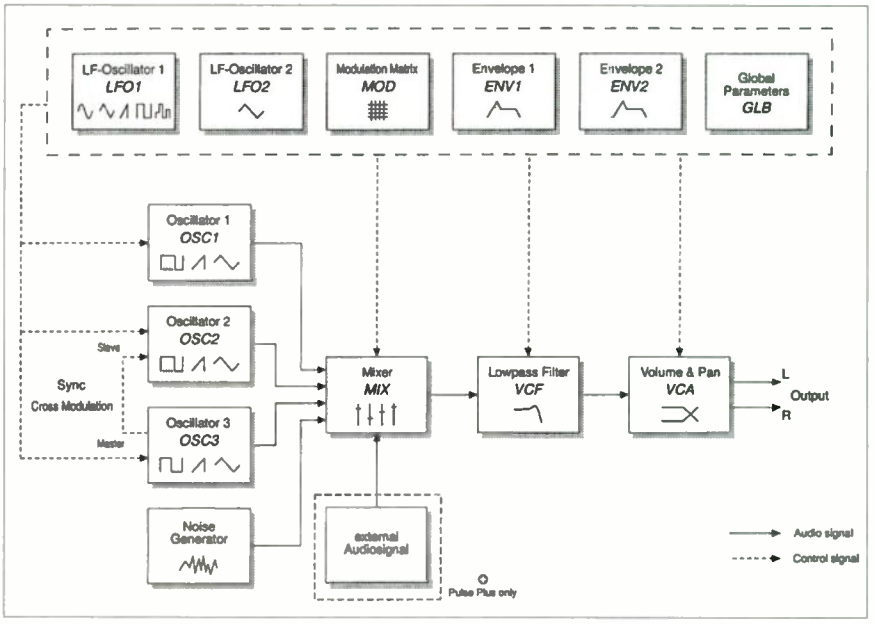

FIG. 1: The Pulse's voice architecture will be familiar to veteran analog-synth users. Components enclosed within the dotted box at the top are modulators. The optional Pulse Plus audio input is shown in the dotted box at the bottom. All other components are in the regular audio path. (Courtesy Waldorf)

and modulation paths are fairly standard stuff (see **Fig. 1**). Three analog oscillators deliver sawtooth, pulse/square, and triangle waveforms and include coarse semitone tuning and fine tuning. You can modulate the pulse width of oscillators 1 and 2 but not that of oscillator 3.

Oscillators 2 and 3 can be cross-modulated, which produces an effect similar to ring modulation. In this situation, oscillator 2 generates a variable pulse wave, and oscillator 3 generates a nonvariable pulse, sawtooth, or triangle wave. These signals are combined to produce a waveform that includes the sum and difference frequencies of the two oscillator outputs. This is great for generating complex timbres.

Oscillator 2 also can hard-sync to oscillator 3, which becomes the master in this application. Each time oscillator 3 starts a new cycle, oscillator 2 restarts its cycle, too, even if the two are producing different waveforms and frequencies. Hard sync is great for bass sounds with impact and cutting lead lines.

Combine these features with the ability to overdrive the signal through the mixer to the filters, and you get a lot of waveform possibilities with just three oscillators. The results are rich, saturated sounds drenched in harmonics.

To top it off, there's also a pink-noise generator, which is routed straight to the mixer. Two flavors of portamento (glide) with duration control are included, as well. With normal portamento, the pitch glides from one note to the next regardless of how you attack the note. Fingered portamento only works with sustained, legato notes.

HOW LOW CAN YOU GO?

The Pulse has two low-frequency oscillators (LFOs). Their frequencies range from 0.0008 Hz to 261.6 Hz, which is the frequency of middle C. LFO 1 features sine, triangle, sawtooth, pulse, and sample-and-hold waveforms. The sample-and-hold form selects a random value every time the LFO cycle restarts.

LFO 2 is limited to a triangle waveform, a frequency control, and a delay parameter. The amount of delay before oscillation begins depends on the parameter setting of the filter envelope (EG 1) trigger mode (discussed shortly).

If you use the triangle, sawtooth, or pulse waveform, you can synchronize LFO 1's frequency to MIDI Clock. In this case, however, you lose the ability to modulate the LFO speed with the Pulse's extensive modulation matrix. When synched to MIDI Clock, the LFO frequency is determined by a rhythmic-duration parameter, which can range from a 32nd note all the way up to eight bars of 4/4 time. If you set this parameter to a short duration, the LFO

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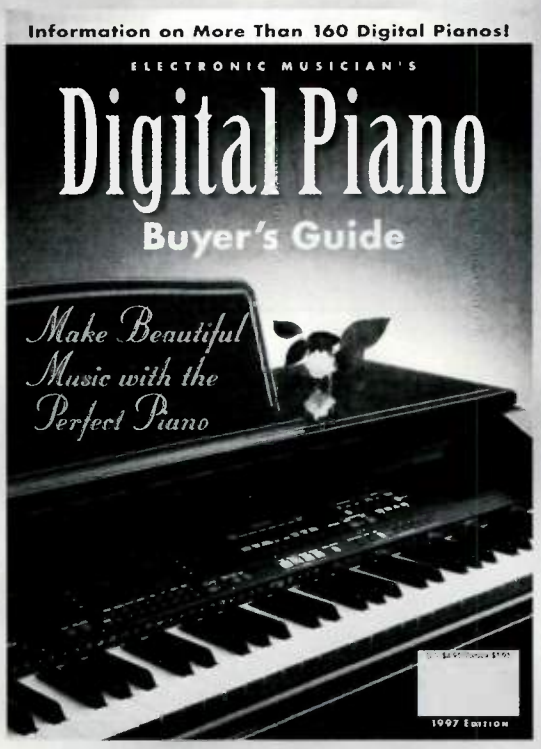

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frequency will be fast; if you set it to a long duration, the LFO frequency will be slow. In any case, the LFO is synched to your sequence.

It's difficult to describe how cool this feature was on a recent remix I did: the synth was pulsing away (excuse the pun) in tempo, with analog bleeps cooking up a groovy little rhythmic stew. It was tasty, to say the least.

THE ENVELOPE, PLEASE

The synth's two envelope generators are functionally identical ADSR types with key tracking. The two EGs can operate in four different trigger modes. These modes provide some nice options that control when envelopes are triggered, retriggered, and/or released as a function of incoming Note On and Note Off messages. Envelope 1 defaults to the VCF, but it can be applied to other components. EG 2 is meant to control the VCA but also can be applied elsewhere.

In Single Trigger 1 mode, the first Note On message starts the envelope's attack stage, and the release stage isn't initiated until all keys are released. Single Trigger 2 starts with the same basic idea, but when the envelope retriggers, it starts at its current value instead of starting at zero every time. In Retrigger 1 and 2 modes, the EG restarts with every incoming Note On. As with Single Triggering, the difference between these two modes is that in Retrigger 1 mode, the envelope restarts at zero, and in Retrigger 2, the EG starts at its current value.

Product Summary

PRODUCT:
Waldorf Pulse analog synthesizer
PRICE:
$999
DISTRIBUTOR:
GSF Agency
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AUDIO QUALITY	● ● ● ● ◖
VALUE	● ● ● ● ◖

The Keytrack parameter (with values of ±64) uses Note On values to control the duration of the envelope stages. If the Keytrack value is positive and you play a note above E4 (MIDI note 64), the duration of the EG stages is lengthened in proportion to the note number, producing a longer envelope; notes below E4 decrease the EG time, shortening the envelope. With negative Keytrack values, the process is reversed, so that notes below E4 increase the EG time.

Many analog synths don't have fast enough envelope response when push comes to shove. In contrast, one of the great things about my old Minimoog is that it has a fast, perfectly filtered attack transient right at the beginning of the sound to make it punch, especially when I am trying to lock the synth with a kick drum. Thankfully, the response of the Pulse EGs is exceptionally fast, so precise, quick attacks are a breeze. I applaud Waldorf here.

CUT OFF AT THE LOWPASS

Other than the oscillators, probably the biggest make-or-break point on an analog synth is the filter, and here again the Pulse shines. A resonant, 24 dB/octave, lowpass filter provides warmth and fatness in abundance, and it can be driven into self-oscillation.

You can set the filter's cutoff frequency and modulate it from a variety of sources. The Cutoff Modulation parameter determines the basic depth of modulation from any source. You can modulate the cutoff by up to ±200 percent via key tracking, and the Envelope 1 Sensitivity parameter determines the amount by which EG 1 affects the cutoff frequency. In addition, the Velocity Sensitivity parameter uses incoming Velocity values to determine the amount by which EG 1 modulates the cutoff frequency. For instance, with a higher Velocity Sensitivity setting, Velocity values have more effect on EG 1 depth, which in turn modulates the cutoff frequency.

LEST WE FORGET

The last major component in the Pulse's audio signal path is the voltage-controlled amplifier. VCAs tend to be pretty mundane affairs, but this one does have a nice extra touch: although the synth is monophonic, its VCA output is stereo. The Pulse offers both a modulatable pan-position parameter and panning via MIDI (CC 10), so you can dynamically move the signal around in the stereo field.

The VCA's basic output level is set with a master Volume parameter. You can modulate the VCA output level with Velocity and EG 2 as well as via MIDI Control Changes.

TRICKS DINO CAN'T LEARN

I admit my affection for retro, fat-sounding equipment is pretty strong. But there are definitely areas in which contemporary designs can smoke those old beloved ARPs and Moogs. On the Pulse, the Modulation Matrix is one such area.

Four modulation paths allow a choice of sources, destinations, and modulation depth, with positive and negative depth values in most cases. Two additional fixed modulation paths for oscillator frequency and filter cutoff are also available, and you get several other important modulation features, including key tracking and pitch following.

This is a very flexible setup. Almost everything can be modulated and addressed via MIDI. All of the parameters are assigned to MIDI Control

Waldorf Pulse Specifications

Program Locations (preset/user)	59/40
Polyphony	1 note
Audio Oscillators/LFOs	3/2
Audio Waveforms	sawtooth, pulse/square, triangle
VCFs/VCAs	1 (lowpass)/1 (stereo)
Envelope Generators	2 (ADSR)
Audio Output Level	+4 dBV, nominal
Signal-to-Noise Ratio	approx. 80 dB
Dimensions	2U x 3.27 inches (D)
Weight	6 lbs.

Change messages, which provides incredible options for real-time control. Setting up a simple modulation template is straightforward, and manipulating, recording, editing, and playing various parameter changes in real time really expanded my appreciation of this little dandy.

I detected no glitches or modulation lag time. I went right into "having fun" mode, playing with harmonics, doing crazy panning, modulating filter resonance into self-oscillation, and creating dynamic performance bits from Pluto. Changing pulse widths, overdriving oscillators, and sweeping pitches all over the aural landscape are just a few of the possibilities. More than just being interesting, the ability to dynamically change the sounds (especially sustained timbres) really expands the notion of a monophonic instrument.

UP THE DOWN STAIRSTEP

The Pulse's arpeggiator is an interesting retro addition that proved pretty fun despite its limitations (chiefly, the limit of sixteen preset rhythmic patterns). Basically, the arpeggiator breaks down MIDI chords into their component notes and triggers the notes according to the rhythmic pattern you've selected. You can select the arpeggiation range (up to ten octaves), the note value (whole notes to 32nd notes, including triplets and dotted values), and tempo (48 to 300 bpm).

The arpeggiator can play up, down, alternating up and down, or random patterns. In Assign Mode Up, as many as ten notes are arpeggiated upward in the order in which they are played. Assign Mode Down is the reverse of Assign Mode Up, and Assign Mode Alternate, as you would expect, alternates between up and down arpeggiation. In Hold mode, the arpeggiator plays on endlessly until you play a new chord, turn off the parameter, or send an All Notes Off command.

The arpeggiator can sync to MIDI Clock (in which case it recognizes Song Position Pointer), or it can act as the master. When the arpeggiator is the master clock, you set its speed with the Tempo parameter. The synth will send both MIDI Clock messages and the notes pumped out by the arpeggiator to the MIDI Out port.

The Pulse's arpeggiator is as much fun as the one in the old Oberheim 4-Voice. I particularly enjoyed setting up a cool, repeating pattern while programming sustained sounds into a more textural pad. (The arpeggiator provides a way to get around monophonic limitations.)

MIDI MIGHT

The Pulse's receives on any MIDI channel and responds to MIDI Program Changes 1 to 100; however, the unit has memory for only 99 patches, so Program Change 100 calls up a randomly selected program. You can set the Pitch Bend depth (scaling) and dump and load patches via SysEx.

Because the Pulse arpeggiator can control other MIDI synths, a simple Panic feature is included to deal with stuck notes. It sends All Notes Off on the base channel, but it does not send individual Note Off messages for synths that don't recognize All Notes Off.

Mod Wheel and Breath Controller messages are always used as modulation sources. In addition, you can select one MIDI Control Change message (or Channel Aftertouch) to act as an assignable modulation source, which is called Control X. These controllers can be routed to filter cutoff, oscillator frequency, or any of the four assignable modulation paths in the Modulation Matrix. In addition, every parameter is permanently premapped to a MIDI Control Change message for real-time modulation.

HEAVEN IS ANALOG

In my work, I often stack patches or use so-called "complex" patches on modern digital synths to create musical parts with a lot of life and movement, sounds that go somewhere interesting. On analog synths in general, and the Pulse in particular, undertaking these little sonic journeys is more tactile and enjoyable, and the results are much more satisfying. Synth programming can still be fun! When it comes to ease of use, I would prefer a better display, but I can't say the Pulse's display is a joy killer, because the machine is user-friendly otherwise.

With the Pulse, you get real oscillators with hormones, rich filters, fast envelopes, sufficient modulation paths, and sounds that have immediacy and impact. Of course, you still have to supply external processing for some of the effects we've grown fond of, but I'd rather add quality outboard effects to a

A MAJOR PLUS

Toward the end of the review period, I was fortunate enough to get the Pulse Plus, a version of the Pulse synth that has additional features analog lovers will find addicting. For only $100 more ($1,099 total), the Plus gives you control-voltage (CV) and gate inputs and outputs along with an audio input.

What does that mean to the average Jane? To begin with, you can use the Pulse Plus as a CV-to-MIDI converter (and vice versa) for analog synths that have CV and gate inputs and outputs. In addition, you can interface the Plus with older analog keyboard synths, which you can't do with the regular Pulse.

You can adjust the CV input and output levels, transpose notes received via the CV input, set a DC offset for each CV output, and scale the CV outputs by applying a logarithmic or linear curve. (The CV input only works with linear voltages of 1V/octave.) The Gate In polarity can be set positive or negative.

These features let you set up the Pulse to properly accommodate the diverse idiosyncrasies of your old analog beasts. It's a pretty flexible and practical design. The CV and external signal levels are also accessible via MIDI controllers, providing some interesting automation possibilities.

The Plus audio input is a must when you have such a nice filter. I pumped some sampled bass loops through the filter and loved being able to sweep the resonance and add a sub-bass that totally transformed my original source material. Don't hesitate to get the Plus version because of the few extra dollars it costs. I don't consider the Plus an option; it's a necessity.

great analog-synth sound than use the mediocre onboard effects found in some digital synths. With quality effects, my philosophy is that a good source sound leads to a good processed sound, and the Pulse's audio quality is excellent. The Waldorf synth worked extremely well with a multitude of tap delays, harmonized choral programs, lush reverb programs, and so on.

I was lucky to have a Minimoog, ARP Odyssey, Roland MKS-70, and a few other vintage pieces to A/B with the Pulse, and I found the Waldorf box to be a real gamer. Although it doesn't replace my old beasts, it does a very good job of striking a balance between an authentic analog vibe and MIDI features we've come to expect in modern synths. It also causes fewer headaches than you get with the old dinosaurs.

▼

Some presets are complex enough to remind me of modular synths.

The Pulse covers the gamut of analog synth applications, albeit with its own strong personality. And although it may seem pricey for a monophonic synthesizer, I feel it represents a solid value. It doesn't have absolutely every desirable feature found in mono analogs—for instance, you don't get an inverter, ring modulator, envelope follower, multimode filter, or white noise generator—but if it had all that, the price would go through the roof.

This isn't the kind of synth you will want to sell off in two years; it's not just a trendy, flavor-of-the-month instrument. If you love analog synthesis, it's a great purchase that will pay back your investment over and over again for many years.

Bill Heagy *is a New York City–based producer and songwriter who is currently producing two singles for the soundtrack to the upcoming film* Brothers Kiss. *He would like to send a big shout to heaven for his late friend, analog-synth king, and fellow* **EM** *author Chuck Fischer, who originally was to have written this review.*

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Alesis NanoVerb

By Barry Cleveland

A pint-sized wonder from the makers of the MicroVerb.

The prefix *nano* is derived from a Greek word meaning "dwarf," and according to my *Reader's Digest Great Encyclopedia Dictionary*, its primary definition is "exceedingly or abnormally small." That certainly describes the Alesis NanoVerb: this ½-rack-space, digital effects processor weighs a little over a pound and almost fits in the palm of your hand.

But don't be fooled: although its case may be exceedingly small, the NanoVerb's sound and specs are anything but. Alesis' new baby boasts 18-bit A/D and D/A converters, 24-bit internal processing, and algorithms derived from the outstanding (and considerably more expensive) QuadraVerb 2 and MidiVerb 4 effects processors.

The unit's appearance and Alesis' promotional materials give the impression that the NanoVerb is aimed at hobbyists and semipro users rather than pros. A quick look at the 30-page manual reinforces that impression: it covers everything from how to unpack and plug in the unit to the basic concepts of effects processing.

But the NanoVerb has professional specs, performs remarkably well in a variety of applications, and is quieter than some popular high-end processors that cost far more. The main reason it is not marketed as a "pro" product is simply that it has very few editing features. You just dial up a preset that sounds good for your application, and away you go. If you want lots of editable parameters, this isn't the box for you.

The unit's sixteen presets include ten reverbs: three halls, three rooms, three plates, and a Nonlinear (gated) reverb. You also get discrete stereo chorus, rotating-speaker simulation, flanging, and mono delay. The final two presets are simple multi-effects programs, one of which combines chorus and reverb, and the other offers chorus, reverb, and delay.

WHAT YOU SEE

The front panel features just five controls: input level, wet/dry mix, output level, detented preset selector, and an Adjust knob for editing certain parameters. The unit's two LEDs indicate power on/off and input signal/overload. Those five controls are all you get; as I said, the NanoVerb is not intended for inveterate program tweakers.

On the rear panel are five unbalanced ¼-inch jacks for the left and right inputs and outputs and the Bypass footswitch. The left input is normalled to the right input for creating stereo output from a mono source; plugging into the right input breaks the normal for discrete stereo operation. There is also a jack for the dreaded wall-wart power supply.

Again, keep in mind that, except for the Adjust knob, this is a preset-only machine. It has no user memory and no MIDI features. What you see on the front panel is what you get.

THERE'S ONLY ONE

The Adjust knob controls one predetermined parameter for each preset. For the reverb presets, it adjusts reverb decay time; for delays, it handles the delay time. In the other single effects, you can adjust the chorus depth/rate, flange rate, and rotation speed (for the rotating-speaker simulation). On the first multi-effects preset, the knob controls reverb-decay time, and on the second it adjusts delay time.

Although there are significant limitations to having only one adjustable parameter, you can adjust the values over a wide range, which allows you to make some pretty drastic changes. For example, the decay time for reverb presets can be short enough to create gated and doubling/stereo-imaging effects, even on hall reverbs, but you can also choose nearly infinite decay, even on plate reverbs.

These might seem like absurd extremes for some users, but I found them quite useful. In fact, I got some of the best "gated" reverb effects by using presets other than the Nonlinear reverb preset, which is the designated gated effect. I found that by turning the reverb decay almost all the way up I could produce shimmering blankets of layered textures without overloading the input or getting runaway feedback. Its ability to handle layers of regeneration smoothly and musically makes the NanoVerb a good choice for ambient-music producers.

One drawback to having such a wide range of variation on a small knob, however, is that it can often be difficult to visually find a previous setting. This was also a problem with the preset-selector switch, which did not always line up with the preset number, especially when viewed from above.

GUITAR MAN

I mostly used the NanoVerb on my guitar rig. After consulting the manual, I tried connecting it to the mono effects loop on my preamplifier, putting it inline between the preamp's stereo outputs and a stereo amp, and plugging my guitar directly into it with a mono output to a mono amp.

When I connected it to the mono effects loop on my guitar preamplifier,

Alesis' NanoVerb proves that good things really do come in small packages, delivering big sound in a little box at a low price.

it sounded okay, but many of the effects were compromised. As might be expected, it worked much better when I connected it in stereo between the preamp and the amplifier.

The unit introduced relatively little noise into the chain, even when using effects with a lot of regeneration. There was definitely some hiss, but it wasn't enough to be unworkable. When I plugged directly into the NanoVerb, there was plenty of gain, and the mono effects sounded better than when I was patched in through the effects loop.

Most of the presets could be tweaked to produce good results with electric guitar, but the flanger was problematic because of its overemphasized low-frequency sweep. (The Nonlinear reverb also didn't work well on guitars, but it clearly wasn't intended for this application.) On the other hand, the rotating-speaker simulator, "Room 3," and the plate reverbs were outstanding in this application.

FOR THE RECORD

Next, I used the NanoVerb at several mixing sessions. After trying the unit on a fairly wide variety of instruments, I concluded that the Concert Hall presets are all useful, particularly with short to medium decay times. However, some instruments tend to make the halls sound slightly metallic. The metallic sound could probably be eliminated if it were possible to adjust the reflections, but that's not an option.

"Hall 1" is bright and has a quick, smooth decay curve. I found it good for vocals, some electric guitar parts, and hand percussion. It is one of the presets that works very well as a gated reverb. It did *not* work well with electric bass, certain synthesizers, and woodwinds. "Hall 2" is similar to "Hall 1," but it is darker and worked best with bright sounds, such as acoustic guitars, bright Strat-type guitars, and hand percussion. On woodwinds it tended to hiss and accentuate breath sounds, and on low percussion it sounded boomy. "Hall 3" has 12 ms of predelay and sounded best with longer delay times. It worked well on snares, big toms, bass, and even some vocals.

The room presets are all quite good and work nicely with a wide range of decay times. I also used them with the decay turned off in order to create nice stereo imaging and doubling effects. "Room 3," which is very warm, was particularly good for this purpose.

"Room 1" does a nice job of emulating a hardwood space with lots of early, bright reflections. I found it useful for everything from guitars and basses to percussion. It has a slap echo that livens up any sound that has a quick attack. "Room 2" is small and relatively flat. It worked well when just a bit of ambience was needed to give an instrument some presence. I liked it on bass, woodwinds, and brass. "Room 3" is a warm studio room that made just about everything sound good, particularly strings and other acoustic instruments.

The plate reverbs are exceptional. "Plate 1" is a bright plate that really sounded great on vocals and also worked well on the larger woodwinds, such as bass flute and bass clarinet. "Plate 2" is a warmer plate designed for acoustic instruments and strings. It was wonderful on cellos, pianos, and acoustic guitars. "Plate 3" is a thin, vintage tube-style plate with very little low

end and a quick diffusion. Although it is somewhat subtle, it enhanced almost everything I tried it on.

Alesis refers to the stereo chorus as "true stereo." In fact, it is discrete stereo, with independent left and right signal paths and a dual-mono algorithm. Therefore, it retains the input signal's original stereo image at the outputs. However, the algorithm does not emulate a real stereo space. (For a more in-depth discussion of the distinction between "true" and "discrete" stereo processing, see the review of Alesis' QuadraVerb 2 in the June 1995 **EM**.)

I found the stereo chorus pleasant-sounding and useful in lots of applications. The left and right channels have independent LFOs that can be set at different rates. However, if you are looking for that ultrathick, Andy Summers sound, this preset won't do the trick.

DOWNS AND UPS

The flanger is by far the most disappointing and one-dimensional preset in the NanoVerb. It is overbearing, has an annoying sweep, and is not very musical. It also boosts low-mid frequencies at the end of its sweep cycle, resulting in an unpleasant boom that tends to overload the input. Nevertheless, when I used it on certain percussion instruments with the rate set low and on vocals with the rate set high, I was able to get some interesting and usable sounds.

The rotating speaker, or "Lezlie," effect sounded great on guitars, keyboards, and even vocals. For this preset,

Product Summary

PRODUCT:
NanoVerb effects processor
PRICE:
$179
MANUFACTURER:
Alesis Corporation
tel. (800) 525-3747
or (310) 841-2272
fax (310) 836-9192
e-mail alecorp@alesis1.usa.com
Web www.alesis.com
CIRCLE #438 ON READER SERVICE CARD

EM METERS	RATING PRODUCTS FROM 1 TO 5
FEATURES	● ● ● ●
EASE OF USE	● ● ● ●
AUDIO QUALITY	● ● ● ● ◖
VALUE	● ● ● ● ●

the Adjust knob toggles between fast and slow speeds. The transition between them is gradual and very similar to the real thing. If you feed just enough signal into the input to trigger the effect, it doubles as a dandy auto-panner. In my opinion, this one effect may be worth the price of the entire unit.

The clean mono delay has three quickly fading repeats. Delay time is adjustable to 1,270 ms in increments of 10 ms. It sounds good on almost anything, and within the limitations of a single-delay preset, it is useful in a variety of applications.

The Nonlinear reverb preset sounds great on some drums and percussion, but it tends to glitch on others, particularly when the gate length is long. As I mentioned earlier, I found that I could often get better "gated" sounds by shortening the decay times on some of the other reverb presets. I did manage to get a strange bagpipe-type sound while using this preset to process a bass clarinet, though. Go figure!

The "Chorus/Room 1" multi-effect preset combines the true-stereo chorus with a large room, and it works well on most instruments. The reverb decay time is adjustable. "Chorus/Room 2" is actually a chorus/room/delay program with an adjustable delay time. It is also a fine general-purpose preset.

NanoVerb Specifications

Number of Presets	16
Audio Inputs/Outputs	2/2
Dynamic Range	>90 dB
THD	0.02% (1 kHz, 0 dBV)
Frequency Response	20 Hz – 20 kHz (+1/-3 dB)
Dimensions	5.25 x 1.75 x 4.5 inches
Weight	1.25 lbs.
A/D and D/A converters	18-bit Delta-Sigma
Sampling Frequency	46.875 kHz

JUST DO IT

The NanoVerb is an outstanding value. With its sound quality, convenient size, simplicity of design, and bargain price, it's pretty much a can't-miss product. My few complaints are trivial when you look at the big picture.

True, the NanoVerb may not be the first choice for Michael Jackson's vocal tracks or John Williams' solo guitar project. But it will certainly be a welcome addition to many serious project studios, guitar rigs, and sound-reinforcement racks. For that matter, I wouldn't be surprised to see NanoVerbs in a major studio or two. Clearly this will be another great Alesis success story.

Barry Cleveland *is a San Francisco–based guitarist, composer, engineer, and producer.*

TASCAM M-1600

By Brian Knave

This low-priced 8-bus console offers a generous feature set.

TASCAM's advertisement for the M-1600 recording console asserts that you'll love the board for what it does but that you'll buy it for what it costs. The ad could well be right. I recorded and mixed with the M-1600 for a couple of months and found it to be a versatile, reliable, and good-sounding board. Certainly, if you're looking to spend the lowest-possible dollar amount for an 8-bus recording console, you should take a good look at this board.

Of course, it's not easy to design a high-quality yet inexpensive mixer. As feature-filled as the M-1600 is, TASCAM couldn't help but make the occasional sacrifice here and there to stay within budget. The question is, did they make the right decisions regarding what to include and what to leave out? Furthermore, do the various components offer the quality you need to make premium home recordings? A close look should help you determine whether or not the M-1600 is the recording console for you.

FACE OF THINGS

The M-1600 comes in two functionally identical configurations: 16-channel (which I tested) and 24-channel. An optional meter bridge, the MU-1616 ($500) or MU-1624 ($650), can be added to either board. (I tested the MU-1616 meter bridge.) Another optional accessory is the MA-8 mic preamp ($450), which provides an additional eight XLR mic inputs. The stock eight XLR inputs, however, should suffice for most applications; the average home-studio owner isn't likely to have more than eight mics anyway.

Considering that it's internally powered, the M-1600 weighs relatively little: the 16-channel model weighs 44 pounds, and the 24-channel board, 57 pounds. Durably constructed of slate-colored, sheet-metal plates screwed onto a metal chassis, the body is compact and cleanly designed. The mixing surface is conveniently tilted toward the user, and the top-mounted jackfield angles slightly away.

Controls and jacks are logically laid out and clearly delineated with white silk-screened lettering and numbering. Navigation is further simplified by color-coordinated knobs and switches. The M-1600 was easy enough to figure out; I used it for several sessions before even opening the manual.

The knobs and 100 mm faders are evidently one of the places where TASCAM saved a few dollars. Except for the pan pots, which are secured to the top panel by brass nuts, the rotary pots are not reinforced and can be readily jiggled back and forth. Curiously, the channel faders are relatively smooth and noise-free whereas the main and group faders stick a bit and produce a slight grating sound when slid. Luxurious controls these are not; however, despite a somewhat loose feel, the pots and faders work fine.

Two 12-segment, bar-graph LED meters monitor signal levels for the control-room mix. Eight 7-segment LEDs show group output levels. The main L/R meters also feature two overload LED lights that come on when the signal level reaches +21 dB over the nominal level (-2 dBu). This proved sufficient headroom for my recordings.

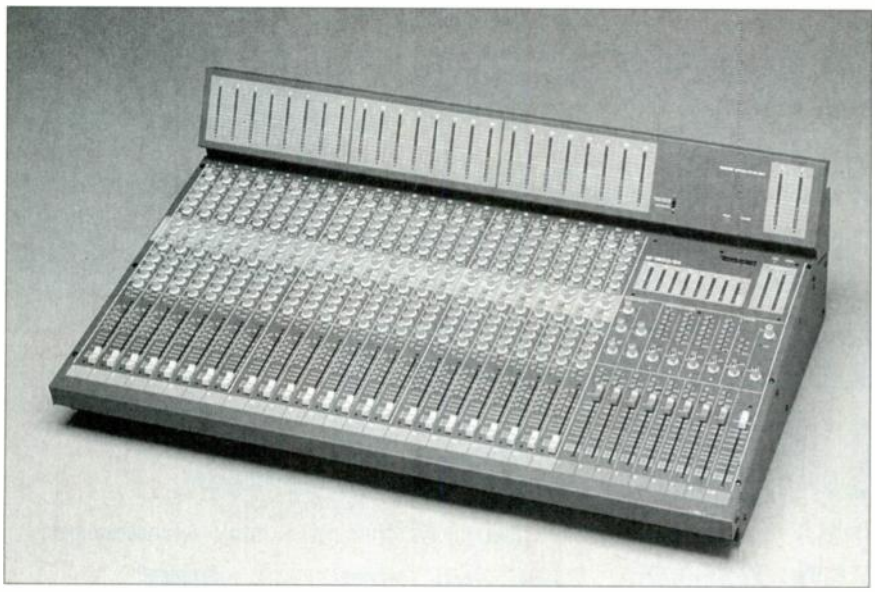
AT PLAY IN THE JACKFIELD

The M-1600 offers balanced XLR connectors on the first eight channels and balanced ¼-inch TRS line-in jacks on all channels. Strangely, on channels 1 through 8, there is no way to switch between the XLR and ¼-inch jacks. In fact, according to the manual, you should disconnect the XLR input while using the line input and vice versa—a considerable hassle. (Despite the warning, however, I left my sound modules plugged in while using microphones on the same channels and experienced no problems.)

Each channel includes ¼-inch TRS insert jacks. However, there are no insert jacks for the main mix outputs or group outputs, so if you need to, say, compress a subgroup or complete mix, you will have to configure a patch bay. (Of course, you could compress the complete mix by running the M-1600's stereo outputs directly through the compressor.)

The jackfield also offers six auxiliary-send jacks, four stereo effects returns (eight jacks), balanced monitor outputs, and stereo sub inputs (for connection of signals from an external mixer or other source directly to the stereo bus). Six Aux Sub jacks provide a direct route into the aux buses. These are useful for cascading effects across multiple mixers. Both the stereo outputs and 2-track inputs are balanced and can be switched between -10 dBV and +4 dBm.

A tiny switch that delivers 48V phantom power to the eight XLR channel inputs (globally) is inconveniently located on the lower right-hand side of the M-1600's back panel. Fortunately,



TASCAM's M-1600 8-bus recording console is designed primarily for users of MDM recorders such as the DA-88 and DA-38. It offers clean design and a generous feature set at rock-bottom prices but requires special D-sub cables for interfacing with multitrack recorders.

an LED indicator light on the top right of the unit's mixing surface shows you when phantom power is on. Also located on the back panel is the M-1600's power switch.

Another inconvenience worth noting is the M-1600's power cord, which is the 2-prong type. I would have preferred a 3-prong, IEC power cord, which would provide a better ground for audio.

MISSING IN ACTION

Perhaps you noticed that I didn't mention the location of the M-1600's ¼-inch group outputs and tape returns. That's because there are none. To use the M-1600, you'll have to have cables that terminate in 25-pin D-sub connectors. From TASCAM's perspective, this is one of the console's strong points because the D-sub connectors provide a more secure connection, not to mention less clutter in the studio.

It's true that once you have the right cables, the D-sub connectors make for a quick and easy setup (or breakdown). And if you already own or plan to purchase a TASCAM DA-88 or DA-38 or Fostex RD-8 tape deck, you can enjoy the full advantage of using cables that terminate in D-subs on both ends. But if you own any other type of recording device, prepare to spend some extra dollars getting properly cabled.

For this review, TASCAM loaned me two 8-channel, D-sub snake cables. These worked just fine while I used the M-1600 with the TASCAM DA-38 (reviewed in the November 1996 **EM**). But to use the board with an ADAT, I had to look elsewhere. Thankfully, Whirlwind was kind enough to lend me three breakout cables built specifically for MDMs. Why did the connection require three cables? Because I used the Elco connector on the ADAT, and cable companies simply do not stock cables for connecting D-sub to Elco connectors. Fortunately, most cable companies will gladly make custom cables. If you intend to use the M-1600 with any recorder other than a DA-88, DA-38, or RD-8, ordering custom cables should be first on your list of things to do.

Between the group-output and tape-return D-sub connectors on the rear panel of the M-1600 are tiny switches for changing the input/output reference levels between -10 dBV and +4 dBm. Generally, if you connect to another D-sub connector, these should be set at +4 dBm; if you use a D-sub that breaks out into RCA or ¼-inch connectors, you'll probably use the -10 dBV level. To be certain, refer to the multitrack recorder's manual.

DOWN THE STRIP

A tour down an M-1600 channel strip shows where the signal flows and how it can be manipulated. First up is the Direct Out switch. When pushed in, this button reroutes the channel's postfader signal directly to a recorder track without first passing through a group output. This is useful—in fact, required—if you want to record more than eight tracks at once, because the first eight signals can go through the group outputs and extra signals can be sent direct.

Just below the Direct Out switch is the Tape switch. This is the button to push when you want to return tape signals through the main channel path during mixdown. While the Tape button is engaged, any signal coming from the corresponding microphone or line input is automatically rerouted to that channel's monitor path. Conversely,

M-1600 Specifications

Equivalent Input Noise	-127 dB
Total Harmonic Distortion (20 Hz–20 kHz @ +14 dB)	0.006%
Channel Crosstalk (@ 1 kHz)	>90 dB
Frequency Response	20 Hz–25 kHz
EQ	3-band with sweepable mid
Simultaneous Aux Sends	6
Dedicated Stereo Returns	4
Channel I/O	XLR (8 channels), ¼-inch balanced (all channels), insert points (all channels)
Talkback/Onboard Oscillator	no/no
Faders	100 mm

disengaging the Tape button directs the signal from tape into the monitor path and the mic or line signal into the channel path—the typical setup for recording overdubs.

Next is the Trim control, which is used to set the initial gain on both mic and line signals. An adjacent, 2-color signal-present/overload LED glows green when a signal higher than -12 dBu is in the channel path and red when the signal reaches 21 dB over nominal level, at which point the channel circuitry is overloaded.

The Trim pots work fine, but their gain curve seems inordinately steep (almost logarithmically so) to my ear. Most of the gain increase happens in the last quarter of the knob's revolution and very little of it in the first half. The technical support staff at TASCAM informed me that the curve is designed to approximate the analogous (and natural) gain curve in human hearing. Of course, the gain curve on trim pots should not be completely linear, but the sharp curve on the M-1600's makes them a bit difficult to adjust.

Below the Trim control are four knobs for the M-1600's 3-band EQ. The high-frequency shelving filter's frequency is fixed at 12 kHz, and the low-frequency shelving band is set at 80 Hz. The midrange band's center frequency is sweepable from 100 Hz to 10 kHz. According to the numbers silk-screened beside the EQ knobs, each level knob offers 16 dB of cut or gain. However, I think it sounds more like 10 or 12 dB either way—enough for most applications, yes, but decidedly not 16 dB.

More challenging is that the midrange sweep does not evenly address the broad range of frequencies. The first half of the knob's revolution affects from 100 to 500 Hz, and the second half covers (presumably) 500 Hz to 10 kHz. That's a huge sweep for a half-twist of the knob. As a result, control of the high mids is less precise than control of the low mids. Personally, I'd rather have it the other way around to more accurately isolate critical midrange frequencies of vocals, guitars, horns, and so on. To be fair, any midrange sweep that attempts to cover so much range with one little knob is probably going to favor certain frequencies while shorting others, but the M-1600's midrange EQ is a little less exacting than it might have been.

You can switch the entire EQ section to the monitor path, a great feature that most budget recording consoles don't offer. I've worked with a lot of persnickety musicians who have griped about their monitor mix: "That's not how my guitar sounded last night!" With the M-1600, you can dial in compensatory EQ while tracking, and then, with the push of a button, redirect those same settings to the tape-return signals for a quick mixdown. The switch also can be used to defeat the EQ while mixing or tracking, which is pretty cool.

AUX CHASE

After the EQ section comes the versatile Selectable Monitor section, one of the M-1600's strong suits. Basically, it's an in-line monitor system that allows Aux 1-2 to be used as a monitor return during tracking or overdubbing. This allows you to monitor the signals coming from tape while simultaneously using the same channel path for recording more signals. Then, by pressing the Tape button during mixdown, you can add mic or line signals to the main mix (again through the Aux 1-2 bus) while returning tape signals through the main channel paths. (This feature is called Flip or Mix B on some boards.)

The Selectable Monitor section consists of a pan pot, an Aux 1-2/Mon level pot, a Mon To L-R switch, a pan control, and a Post switch. The Mon To L-R button is the source-select switch. When overdubbing, you depress the Mon To L-R button to place the tape-return signal into the monitor path. You can now monitor tape level and pan position using the Selectable Monitor.

When Mon To L-R is switched off, the source signal goes to the channel path, from which it is trimmed, panned, and sent to the main mix. That's the setup to use while mixing down so that you can use Aux 1-2 for effects. The Post button selects whether the effects (or other signals) going to the Aux 1-2 bus are pre- or postfader.

Next down the channel strip is the Aux 3 knob, which is accompanied by a switch labeled From Aux 1-2/Mon. When this switch is on, the signal from the Aux 1-2/Mon level control is sent to the Aux 3 send-level control. This allows you to put effects in your monitor or headphone mix without printing them to tape—an important application that's useful while tracking. Furthermore, Aux 3 can be assigned to the microphone/line inputs during mixdown, which is helpful if you want, say,

an outboard reverb on a snare-drum signal coming from a sound module. When the switch is off, Aux 3 reverts to a regular postfader send. Auxes 4, 5, and 6, which are postfader, complete the Aux Send section.

TASCAM's in-line monitor system provides the M-1600/16 with 32 inputs at mixdown (48 for the M-1600/24). But because the mic/line inputs must return through the Aux 1/2 bus, the maximum number of effects on a full 32-input mix is four. Of course, if you then assign Aux send 3 to the line inputs returning through the Aux 1/2 bus (as in the snare-reverb example above), that leaves three effects sends for each tape return.

Fortunately, though, another feature, Sum 3+4, allows you to place a single signal processor in either or both of the Aux 3 and 4 signal paths simultaneously. This feature, which to my knowledge is unique to TASCAM consoles, helps give the M-1600 notably more effects-routing options than other boards in the same price range.

YOUR ASSIGNMENT

Directly below the Aux section is the channel pan control. The pan pot, of course, is used not only for panning during mixdown, but for assigning signals to odd or even-numbered groups while tracking. Next down is the Solo switch, which solos the channel signal "in place" (another rare feature for a board in this particular price range), followed by a mute switch.

Channel-assign switches are located to the left of the faders. The input faders are linear rather than logarithmic, and they offer an extra 10 dB of gain above the unity-gain setting. Each group fader is accompanied by three switches: solo, mute, and a left or right main-mix assignment switch. However, there is no stereo pan control or mono button for the group faders, which means that the group outputs can only enter the main mix either hard left or right. Again, you still get more control of group-fader signal routing than you do with comparable boards in this price category.

MASTER BLASTER

The M-1600's master section has five Aux Send Master level controls. As noted earlier, the Sum 3+4 switch combines the signals from Auxes 3 and 4, further increasing effects-routing options.

Controls for the four stereo effects returns include individual level controls, solo switches, and switches for assigning the returns to the L-R mix, Aux 1-2 bus, and group buses. The monitor section offers simultaneous monitoring of the L-R mix, Aux 1-2 bus, and 2-track inputs. Another switch allows you to check the mix in mono: this important feature lets you easily check for phase discrepancies. Level controls for the studio monitors, soloed elements, and headphones are also located here.

I was glad to see two headphone jacks on the M-1600, something else you don't get on all boards in this price range. In my studio, I typically use one jack for feeding a headphone distribution amp (located near the musicians) and the other for my headphone monitor while tracking.

SOUND JUDGMENT

When the M-1600 arrived, I was in the middle of working on a complicated mix that required multiple instances of track sharing and many fader, pan, and effects-level moves—and thus, it took many passes to learn. The arrangement included five vocal tracks and several other audio tracks (all heavily equalized) plus numerous MIDI tracks and accompanying sound modules.

The upside was that after running the mix literally dozens of times and finally completing it on a Mackie 8•Bus console, I knew it by heart. This afforded me a good chance to "hear" the M-1600. To do so, I set up the M-1600 in place of the Mackie and attempted to re-create the same difficult mix.

Granted, the Mackie 8•Bus board has 4-band EQ with two bands of sweepable mids, one of which is fully parametric. (Keep in mind that the 16-channel Mackie board costs about $800 more than the TASCAM M-1600.) To my surprise, though, I was able to produce a comparable mix with the M-1600. In fact, were it not for the EQ advantages afforded by the Mackie, I think the quality of the two mixes would be nearly indistinguishable.

I did similar tests on the mic preamps, again comparing them to the Mackie's. I set up both boards simultaneously and used a Y-cord to send a single vocal performance through both consoles and then to adjacent ADAT tracks. I then panned the two tracks hard left and right and A/B'd them on different playback systems. Again, the M-1600 fared well. I could hear a slight difference in

sound—the Mackie track sounded warmer by a hair—but it would be hard to say which track was more true to the source material. The M-1600 may be a budget mixer, but one component TASCAM didn't scrimp on was the mic preamps.

BRIDGE BEEF

As mentioned earlier, I also tested the MU-1616 meter bridge. The MU-1616 and MU-1624 meter bridges offer 12-segment bar-graph LED meters that can be switched to display either the tape-return or group-output levels (but not mic/line inputs) for each channel plus 12-segment LEDs and overload indicators for the main-mix levels.

The MU-1616 worked fine, but it was a bear to install. For one thing, the screws that secure the unit to the board are tiny, and I repeatedly dropped them into the insert-point jacks as I attempted to install the meter bridge. (I had to turn the board upside down to retrieve the screws.) Also, rather than providing a single set of wires for connecting the meter bridge to the board, TASCAM terminated each meter with its own connector. The wires are short and the plastic connectors are minuscule, so hooking the thing up proved to be a real headache.

It should also be noted that the meter bridge, instead of being swivel mounted, is fixed to the body of the board. That is, once screwed into place, it's immovable. This is not an uncommon design, but it does mean that the meter bridge somewhat obstructs access to the jackfield. Of course, this should prove no problem as long as you don't anticipate much cable repatching in the jackfield.

Product Summary

PRODUCT:
M-1600 recording mixer
PRICE:
M-1600/16: $1,699
M-1600/24: $2,199
MANUFACTURER:
TASCAM
tel. (213) 726-0303
fax (213) 727-7656
CIRCLE #439 ON READER SERVICE CARD

EM METERS	RATING PRODUCTS FROM 1 TO 5
FEATURES	● ● ● ◖
EASE OF USE	● ● ● ●
DOCUMENTATION	● ● ● ◖
VALUE	● ● ● ●

But if you're constantly making new connections and rearranging cables—and especially if you don't use a patch bay—you may find that this meter bridge gets in your way. (Note, too, that the meter bridge requires its own wall-wart power connector.)

On the other hand, TASCAM is to be commended for at least offering a meter bridge for the M-1600—an option you don't get with other consoles in this price category. If you do opt for the meter bridge, you'll find that once you install the unit, it looks really cool and functions beautifully.

CONCLUSIONS

For the money, TASCAM's M-1600 is one heck of a deal. Dollar for dollar, the M-1600 offers more features than any other mixer in its class. It provides plenty of aux sends, an in-line monitor section (with EQ), in-place soloing, and good-sounding mic preamps.

▼

One component TASCAM didn't scrimp on was mic preamps.

My main gripe is about the EQ, which, although flexible, is somewhat compromised in the midrange section. Also, it would have been nice for people who already own (or intend to buy) a recorder other than a DA-88, DA-38, or Fostex RD-8 if TASCAM had included standard ¼-inch or RCA group-output and tape-return jacks in addition to D-sub connectors. Finally, I don't think a mixing console that in so many other ways qualifies as being "professional" should be outfitted with a 2-prong power cord.

But again, for the money, the TASCAM M-1600 is hard to beat. Personal- and project-studio owners who are looking to spend the lowest dollar for the most amount of mixer should definitely check this unit out, especially if they intend to match it with a TASCAM or Fostex MDM. My guess is that they'll love it for what it does but they'll buy it for what it costs. ●

Gracenote Musician's Manage (Win)

By Cat Taylor

Painless organization for musicians. No, really!

You're a working musician, and you're pretty smart, but you have neither the time nor the money to spend on money-management software, databases, and word processors. But you keep hearing horror stories about IRS audits, and this voice in the back of your head is saying, "You have to get it together!"

Musician's Manage was created specifically for musicians by the musicians at Gracenote Software to address these problems in a simple and effective way. The program consists of four separate modules: a money manager, a database for addresses and phone numbers, a database for repertoire, and a simple word-processor.

Each database module includes reference fields to help you organize your data, but the tables can't be customized. You can't, for instance, add new fields to the databases or design your own reports. In return for giving up some of the power of a full-fledged database, however, you get a program you can use right out of the box without spending hours poring over a manual.

I evaluated *Musician's Manage* on a 100 MHz 80486 PC with 16 MB of RAM running Windows 95. The 16-bit program also runs under Windows 3.1 or Windows for Workgroups, and a DOS version is available. A 32-bit version of *Musician's Manage,* written specifically for Windows 95 and Windows NT, is planned.

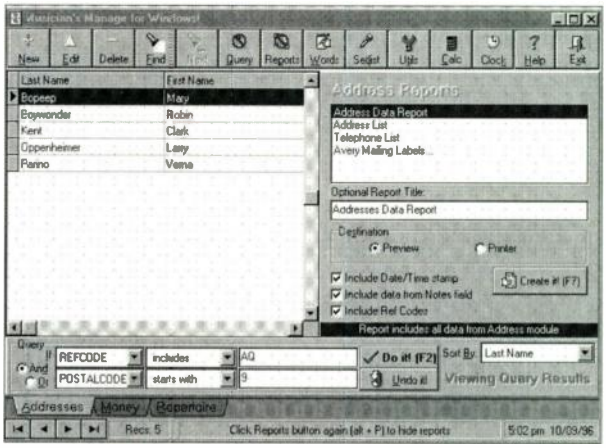


FIG. 1: The main Address screen appears when you open *Musician's Manage*. Note that the program's Query and Reports functions have been engaged by pressing their toggle buttons at the top of the screen.

DATABASE MODULES

When you open the program, the Address module comes up (see **Fig. 1**). The other two database modules (Money and Repertoire) can be accessed by clicking on tabs near the bottom of the screen. The main functions are accessed with a series of buttons across the top of the

screen; there are no pull-down menus. This button bar is the same for all three database modules.

The Query and Reports functions (explained later) appear at the bottom and right of the screen, respectively, and are accessed with toggle on/off buttons on the button bar. There are also a few simple utilities, including a calculator, digital and analog clocks, data backup for each module, and data-indexing and packing functions.

The databases display your data in rows and columns, as a spreadsheet does. You can sort the data onscreen by certain fields, such as last name and postal code, by clicking on a drop-down list in the lower right corner of the screen. Only some of the fields are shown in the grid, and you cannot choose which ones to show, though you can change their order on the screen by dragging the headers. Clicking on any of the rows brings up the detail/editing view in which you can change or add data. It is possible to enter data into all of the fields through keyboard commands, without ever having to use the mouse, which is a great time-saving feature.

The Query function lets the user display only the records that meet chosen criteria. For example, you can query for people whose first name starts with J and whose last name is not Doe. The Report function puts your data into different forms, including Income and Expense Reports, and you can print on 25 types of Avery mailing labels for both pin-feed and plain-paper printers. If your favorite type of label is not listed, you may be able to work around that by adjusting the way sheets go into your printer. You cannot modify or create forms, but several are available in each database module.

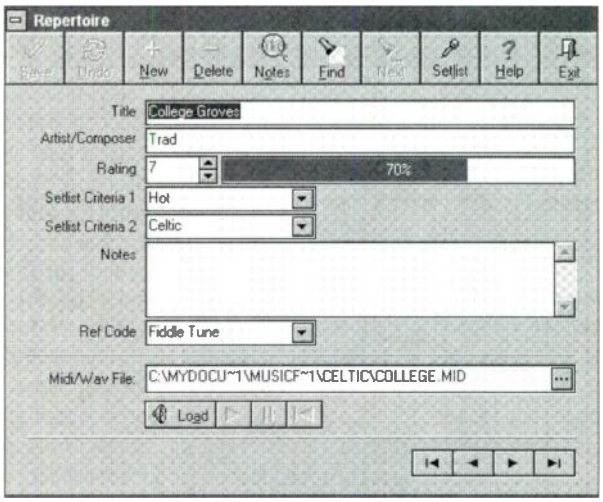


FIG. 2: The Detail/Editing window of the Repertoire module lets you enter descriptive information for your database. You can also play a MIDI or WAV file from this screen.

LITTLE BLACK BOOK

The Address module has the basic fields you would expect to find: name, two address and phone fields, date entered/updated, and a notes field. Unfortunately, there is no field for e-mail addresses. I am told this omission will be remedied in the 32-bit version of the program. You could put e-mail addresses in the second phone field, but the

field only allows 30 characters, so if you also need to record fax numbers, you are out of luck.

A reference field lets you enter short codes to create your own groupings. This allows you to keep lists for more than one band, in case your life is as complicated as mine. Just assign a reference code for each band name; then, to call up the list for that band, run a query for the reference code. Once you enter a code into the reference field of one record, it automatically appears in a drop-down list when you edit another record's reference field. If you delete a particular reference code from all of your data, it disappears from the list.

ONE FOR THE MONEY

The Money module is set up to keep track of income and expenses. You enter a description (for example, Amazon Queen at Joe's Bar), the date, amount, method of payment, notes, a reference code of your choice, and whether the item is income or an expense. If you click the radio button for income, you can enter the beginning and ending times for the gig. If you choose Expense, you can keep track of equipment you've purchased by entering the place you purchased it, the serial number, and the technical-support phone number. The program then allows you to create a report of your equipment by checking the box labeled "Include Expense Item in Inventory Report?"

Although there is no suggested method for keeping track of sales inventory by price, the program allows for a simple workaround using the Query and Reports functions. If you are selling tapes, CDs, and other merchandise, you can keep track of them by entering the band name and venue in the description field and then using the reference code for whatever item you have sold. Let's say you are tracking CD sales. First, enter the number of CDs sold in the Notes field. You can then do a query for records where the description field includes the name of the band and the reference field is CD Sales. Do an Income Report that includes the Note field, and you can see what you've sold and where.

This module is not intended to take the place of real money-management software such as Intuit's *Quicken*. It will not help you balance your checking account, and it will not export into tax software. It is merely intended to list your income and expenses. Make sure you've organized the descriptions and reference codes so that you can easily obtain the information you need.

REPERTOIRE

Now for the module that's the most fun: the Repertoire Module (see **Fig. 2**). After entering the title and composer for each song, you can enter a rating for the song from 1 to 10 based on your own evaluation scale. You could choose to rate songs by their relative importance; for example, perhaps you have to play "Twist and Shout" every night. You then have two set-list criteria boxes that take user-defined reference codes. You might set one of them up based on instrument changes and the other based on the type of song. And you still have another reference code box to use for band name, type of music for a specific gig, etc.

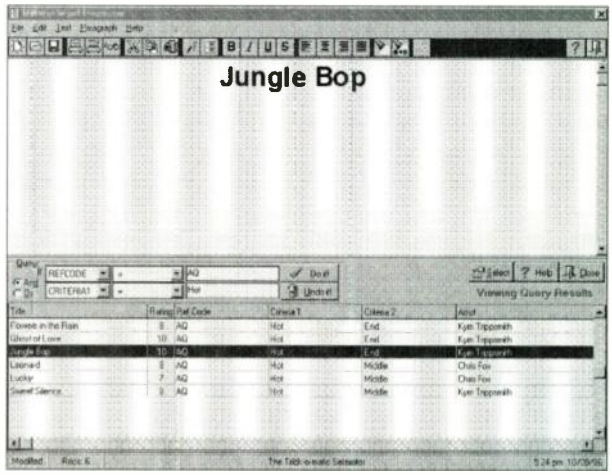


FIG. 3: The *Musician's Manage* Trick-o-matic Seterator builds a set list in the word-processing module, based upon a query in the Repetoire module.

At the bottom of this form is a box named Midi/Wav File. Now this is cool. If you have a sound card on your system and the right software, you can record a song and save it as a WAV file or create a song in a notation program, save it as a Standard MIDI File, and reference it here. You can then play it right from the

program's window. If you have Windows 95 and enough memory, you can even play it at a slower speed to learn it!

As a violinist and a workshop teacher for the Renaissance and Dickens faires, I need to keep track of a great deal of music by various criteria. This module has so many query possibilities that it made cataloging everything from rock songs to old Celtic ballads a snap.

TRICK-O-MATIC

But the raison d'être of the Repertoire Module is the Trick-o-matic Seterator (see **Fig. 3**). This feature allows you to choose tunes from a list at the bottom of the screen and add them to a set list created in the word-processing module at the top of the screen. You can use all the criteria chosen previously to help you build a set list for any given gig.

The word processor lets you edit and copy your set lists once they are built, but it is also a fully functional word-processing program that opens and saves files in Rich Text Format (RTF), which is compatible with all of the major word-processing programs. It gives you basic editing functions (except undo) and allows you to Print Preview and paste in graphics from the clipboard. It also includes templates for creating simple contracts and letters.

GETTING GOING

Although there is no printed manual for the Windows version of *Musician's Manage* (there is for the DOS version), the online help is very good, and there are bubble hints all over the place. Gracenote's representative told me that beta testers found the user interface so friendly they didn't need the manual, so the company decided to save a few trees and stop printing them.

Musician's Manage comes with quite a few fun and rather irreverent sample records to allow the user to experiment with the program and see how it organizes data. You can use all the Query and Report functions and get a feel for the program before you start using it. I recommend adding some of your own data to the sample records and then running some reports to see how you will need to set up your reference codes and use the querying capabilities for your needs. Then delete the sample data, enter your own, and run some reports at the end of the first month to see if it's working how you want it to.

One word of warning here: back up your data. (The program includes a utility for backing up to floppy disk.) If the program starts acting strangely, use the Pack and Reindex functions. Reinstalling the program should only be used as a last resort: it will write the sample data over whatever data you may have entered previously.

CONCLUSIONS

Musician's Manage is intended to be a simple tool for busy people who need organization but have neither the time to learn a lot of programs nor the money to pay for them. Although you cannot add fields to the databases or create your own reports, the program already has what you need to get organized. It's simple to use and easy to learn. The overall flexibility of the Query and Reports functions, in conjunction with the reference fields, allows you to adapt the program to your needs, which is a major strong point.

The three data tables use the *FoxPro* xBase file structure, so if you eventually need to move to a more powerful, full-service database program, you should have no trouble importing your data from *Musician's Manage*. In the meantime, you can get started by downloading a free 30-day trial copy of the program from Gracenote's Web site. So, what are you waiting for? Get it together!

Cat Taylor *is an eclectic electric and acoustic violinist currently performing with Amazon Queen and Haunted By Waters.*

Product Summary

PRODUCT:
Musician's Manage
business software
PRICE:
$89
SYSTEM REQUIREMENTS:
80386 or better PC with 2 MB RAM; Windows 3.1 or 95
MANUFACTURER:
Gracenote Software
tel. and fax (714) 533-4117
e-mail friends@gracenote.com
Web www.gracenote.com
CIRCLE #440 ON READER SERVICE CARD

EM METERS	RATING PRODUCTS FROM 1 TO 5
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EASE OF USE	●●●●◖
DOCUMENTATION	●●●●●
VALUE	●●●●◖

Musicware Nightingale 3.0 (Mac)

By Rob Shrock

An easy-to-use feature-laden notation program

Fast, effective notation software has been a dream of musicians ever since computers were first applied to music. However, the reality of computerized music notation often falls far short of a practiced hand putting pen to manuscript paper. After all, music notation is extremely dense, with more symbols than most languages, and it exhibits a 2-dimensional structure (pitch and time), which challenges even the most proficient programmers.

Once a program with reasonable transcription capabilities has been created, users normally must contend with a steep learning curve before they can get the most out of the software. Musicware is trying to address these issues with *Nightingale* 3.0, a fast, easy-to-use, full-featured desktop music-publishing program.

PRELUDE

Nightingale 3.0 is designed to run on a Macintosh Plus or better. It requires at least 2.5 MB of RAM with System 6.0.5 or higher or 4 MB of RAM with System 7.0 or higher to support one average-sized score and extracted parts. Of course, increasing the memory allotment enhances performance, maximum score size, and number of simultaneous open files. However, *Nightingale* can run in as little as 1.4 MB of RAM and still remain surprisingly fast and stable.

Musicware advises Power Mac users to turn off the Modern Memory Manager, which can cause the computer to crash when extracting parts. I ran *Nightingale* 3.0 on a 68030 machine with 4 MB of RAM (using Connectix's *RAM Doubler*), a 68040 machine with 12 MB of RAM, and a Power Macintosh 7100 with 24 MB of RAM. It was amazingly fast and stable on all three machines.

Nightingale 3.0 comes with a 172-page tutorial manual, which makes it easy to dive right in and start creating scores and extracting parts. This is much easier than having to pore over encyclopedic volumes before you can start using the program. Musicware has always been attuned to the educational aspects of their products; after a single afternoon with the tutorial, you will know the basics of the program and be able to create simple scores.

The manual is well written and easy to understand, with only an occasional mistake or typo. To make things even easier, a complete 1.4 MB online hypertext manual is available as a Microsoft Help file, complete with a word-search engine and related information only a mouse-click away. Not having to thumb through indexes in the back of thick manuals shaves weeks off the learning process.

INTERLUDE

Bundled with *Nightingale* is *NoteView* 2.2, a freely distributed, stripped-down version of *Nightingale* for posting, viewing, and listening to scores on the Internet. *NoteView* is a freeware version of *Nightingale* with all Save and Print functions disabled. Included with the *Nightingale* package, *NoteView* is also available free from Musicware's Web site. The program provides a score-viewing and MIDI-playback vehicle for composers and publishers to distribute examples of their work created in *Nightingale*.

This could be an exciting, innovative way to expose the global public to new composers, arrangers, and songwriters who might never be noticed otherwise. Publishers can now showcase examples of their catalog, possibly establishing contact with new talent in the process.

Also available for an extra $149.95 is *NoteScan* 1.04. This optical character recognition (OCR) program that lets you scan and import printed sheet music into *Nightingale* for editing and reprinting (see sidebar, "The Scan Artist").

EXPOSITION

You start by specifying page size and margins; number, names, and sizes of parts and staves; MIDI channels and patches; and instrument transpositions. After these basic attributes are determined, the clefs, key signatures, time signatures, tempo, systems, notes, and other symbols can be added.

A score can include up to 64 staves and 100 voices with a maximum of 31 voices in a single part. (A voice is an independent, polyphonic musical line played by one instrument.) The window displays a WYSIWYG view of your score, and the Reduce and Enlarge commands zoom in or out. In the current version of the program, you are limited to ten preset viewing percentages: 25, 38, 50, 75, 100, 150, 200, 300, 400, and 600 percent. According to Musicware, these percentages provide the most consistently accurate shapes with the Sonata screen font.

In addition, the program is limited to viewing the score as a series of pages;

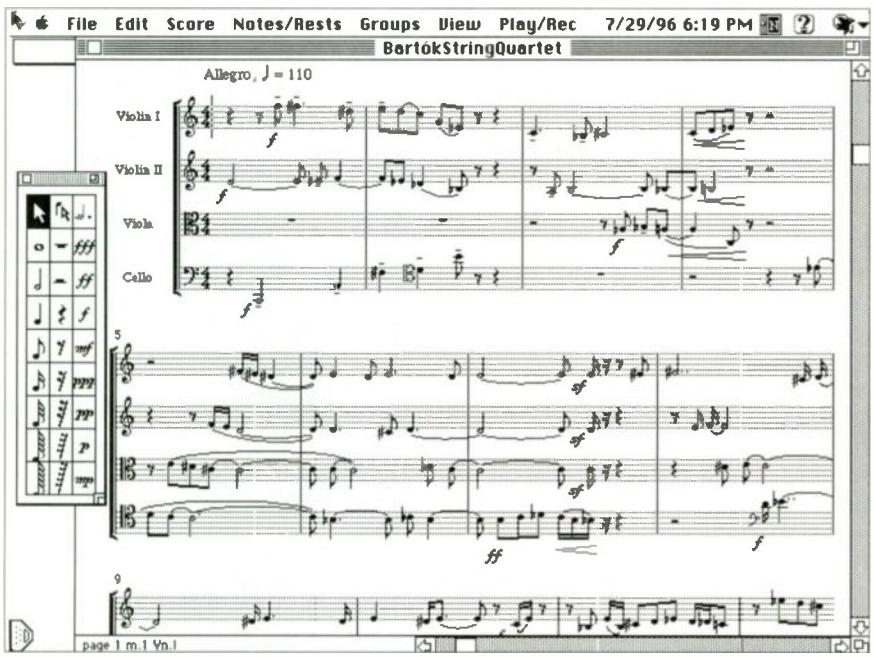

FIG. 1: When a score is opened or created in *Nightingale*, each page is visible by adjusting the scroll bars. Ten preset zoom levels affect the size of the visible score.

there is no way to view it as a continuous scroll. In all notation programs, layout and formatting must eventually be done in Page View, but Scroll View sometimes helps initial note entry go more quickly. Musicware plans to offer both user-definable zoom and Scroll View in the near future.

Unfortunately, *Nightingale* 3.0 allows only 5-line and single-line staves, and guitar tablature is sorely missing. There are elaborate workarounds to produce unremarkable tab notation, but this is a slow, tedious process. Musicware is well aware of these limitations.

Notes can be entered into the Master Score page (see **Fig. 1**) in several ways. Like other notation programs, you can click on the appropriate symbol in the Tool Palette (see **Fig. 2**) and place it on the staff. This palette includes the most commonly used symbols and can be resized as needed; as a result, the *Nightingale* desktop stays less cluttered than other programs that fill the screen with numerous palettes, leaving practically no view of the score. Many of the tools (e.g., the Key Signature tool) automatically open a dialog box that provides additional parameters.

If some of the less-used characters in the customized Sonata screen font are needed, they can be copied and pasted into the score, much like in a word processor, using the Text Styles command. All symbols have an equivalent computer keyboard command.

When you click on a symbol in the palette, the mouse cursor becomes that symbol, making it easy to place in the score. However, the cursor doesn't change size at different zoom levels, which is a bit disconcerting. In addition, when placing a note above or below the staff, the highlighted box surrounding the note obscures a clear view of the ledger lines. This could be corrected by temporarily extending the lengths of the ledger lines to the right and left until the note is clicked in place.

Durations can be as small as 128th notes. Tuplets can have as many as 56 notes in the space of a quarter note, but there are not as many ways to display tuplets as in some other programs. You have the choice of viewing the bracket or not, viewing the number or not, and viewing the number as a single digit or ratio (3 or 3:2 for a triplet). The location of the tuplet number can be dragged to any location. However, the bracket is only available as a squared type; you do not have the option of rounded brackets. In addition, the bracket cannot be displayed with a break in the middle for the tuplet number, which always appears above the bracket.

FIG. 2: The Tool Palette can be resized to take up less space on the screen. The individual icons can also be rearranged to display only the most frequently used tools.

INSTRUMENTALIST

In addition to using the mouse and key commands, notes can be entered into a score from a MIDI controller using step-time or real-time recording. *Nightingale* does not profess to be an extensive MIDI recorder, but dynamics are preserved.

There are two options for MIDI recording. With Insert recording, new measures are created from the insert point; with Merge recording, new notes are blended into existing measures. Each technique is suitable for creating

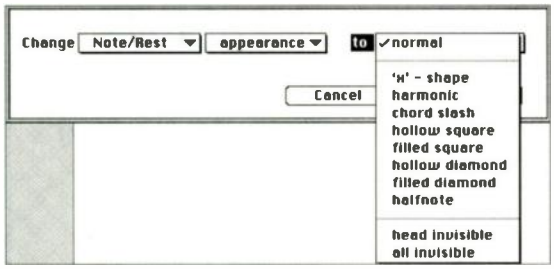

FIG. 3: Context-sensitive modifications can be easily made to selected events with the QuickChange command. Drop-down menus provide access to a variety of common variables.

certain kinds of parts. Step Record Merge is a good choice for hi-hat or bass parts, while Real Time Record Merge is a good choice for entering a piano part (as long as your playing is decent).

Using step recording, it doesn't take long to get a feel for choosing the note duration with a key command (letter Q for quarter note, letter E for eighth note, etc.) and playing the note on a keyboard with the other hand. This can be the quickest and most accurate way to input notes once you get the hang of it.

In real-time recording, notes can be quantized to a selectable grid and automatically transposed (say, for bass parts). In addition, a user-specified note number can be used to split a part into two staves (e.g., for piano parts), and the program can take a shot at guessing the beams by beat. Unfortunately, the internal metronome click from the computer is unstable, even on a Power Mac 7600. If you plan on playing parts in real time, use the MIDI metronome.

Another problem with recording in real time is that it doesn't begin until you start playing, which makes it difficult to input a phrase that begins with a rest. There is no countoff to begin recording, so you must play a dummy note to start recording and go back and delete it later.

Nightingale 3.0 supports OMS and Apple MIDI Manager and has a built-in MIDI driver so you don't need to use a system extension. (That's one less potential extension conflict!) I originally experienced some trouble getting *Nightingale* to communicate properly with my OMS 2.1 setup, and the online documentation is incomplete regarding OMS. MOTU's FreeMIDI extension is not supported.

MOVEMENTS

Selecting and highlighting are done with the Selection Arrow tool (solid arrow). In addition, Musicware has designed a useful MouseShaking feature that lets you "shake" the mouse from side to side in order to toggle between the Selection Arrow and the tool you were last using. This works well, and it's especially handy for inserting a symbol and immediately selecting it for editing after a shake of the mouse.

A separate Dragging tool (outline arrow) is used for dragging symbols around on the screen, and a unique Threader tool (pointing-hand symbol) lets you quickly select notes or rests for editing by holding down the mouse button and touching note heads or rests. Touching a note or rest with the Threader again deselects it.

Practically anything in the score can be moved once it has been placed. Noteheads can be dragged up, down, left, and right using the Dragging tool. Measures can be respaced by dragging the bar line, and beats can be resynchronized to line up with other parts in the score. Measures can be respaced manually or automatically, and systems can be easily justified and reformatted.

I found one dragging bug. I created a piano part with a melody line in octaves that I wanted to lower by one octave. I dragged the upper notes down two octaves, leaving the original lower notes where they were, thinking the notes would rebeam themselves. However, the beams went haywire; the only way I could correct this was to delete the notes and start over.

There are several other ways to accomplish my goal. For one thing, the whole line could have been selected and transposed, or I could have deleted the top octave and reinput the new lower octave, or I could have unbeamed the notes before dragging and then rebeamed. But it was only a few notes, so my intuition told me to just grab and drag.

I discussed this bug with a Musicware rep, who said that if it felt intuitively correct to perform a function a certain way, the company would like to provide the means to do it in that way. Based on that positive response, I feel confident that the next release of *Nightingale* will allow you to drag a note above or below another note to which it is beamed without consequence. I appreciate that Musicware works to improve their products by listening to customer feedback.

DEVELOPMENT

As mentioned earlier, up to 100 voices can exist in a score. A single voice can include an unlimited number of notes. However, if you want two independently beamed lines in the same staff (e.g., first and third French horn), a second voice must be created in the part. In addition, a part with multiple voices cannot be split into separate MIDI channels for playback.

In addition to the traditional G (treble), F (bass), tenor, and alto clefs, you can place a movable C clef anywhere you want. Complex time signatures are supported (e.g., 3+2/4 and 6+7/8), and a variety of nonstandard noteheads are available. In fact, any text symbol can be imported and used as a notehead, which could be useful for some types of avant-garde music.

Although key signatures can be inserted anywhere in the score, there is currently no provision for canceling the previous key. As a result, you must manually place any courtesy natural signs to help clarify the new key in each individual part of the score.

Transpositions can be independent in the master score and individual parts, allowing the score to be viewed in concert pitch while the parts are notated properly for each instrument. Discontiguous staves can be viewed by selecting unwanted staves and using the Hide View command.

Nightingale has a feature called QuickChange that is handy for making fast edits to selected events (see **Fig. 3**). After

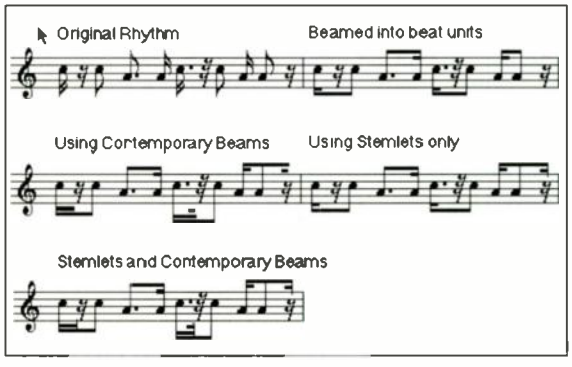

FIG. 4: Several beaming styles are available in *Nightingale*.

selecting a symbol or group of symbols, context-sensitive drop-down menus let you make changes to those symbols. For example, this feature allows you to change regular noteheads to percussion notes, such as the "x" noteheads used in hi-hat parts.

After these changes are made, the symbols remain highlighted, making it simple to continue editing or return to the original settings. This is a great feature; some programs automatically deselect the symbols after the edit takes place, only allowing you the option of undoing but not continuing on with further editing.

Notes can be beamed by groups or beats, or they can be automatically beamed. Beams can easily be flipped, resized, angled, or made invisible. Several styles of beaming can be applied when the part involves a lot of rests and syncopation (see **Fig. 4**). In addition, chord symbols can be added and moved or transposed if attached to another symbol such as a barline or chord. Chord slashes, cross-staff beaming, and fully programmable rehearsal marks are also available.

FEATURED GUESTS

Standard MIDI Files can be imported, but they can't contain more than 64 tracks. Even on slower Macs, SMF importation is impressively fast compared to some other programs, typically taking only a few minutes for an average-sized file (e.g., a 4-minute song with ten or twelve parts). Expect to spend a lot of time cleaning up the score, though; importing files is just a quick way to get notes into the program, and none of the current notation programs work magic in this area. Standard MIDI Files can also be exported.

As an added bonus, Coda *Finale* ETF (ENIGMA Transportable File) documents can be imported relatively intact into *Nightingale* 3.0. However, the process is not perfect, and some editing is still required (see **Fig. 5**). ETF files cannot be saved for exporting back to *Finale*. However, you can take advantage of *NoteView*'s capabilities by importing

THE SCAN ARTIST

NoteScan is an optional ($149.95) optical character recognition (OCR) program designed to import TIFF files of scanned sheet music into *Nightingale* for further editing and reprinting. This technology is imperfect, but if it continues to improve, it will let musicians create condensed scores, change keys and layout, and rearrange existing pieces of sheet music. It will also provide an alternate means of generating a MIDI file from the print medium.

The *NoteScan* User's Guide (which is part of *Nightingale*'s extensive online help file) includes detailed instructions on maximizing the scanning process for optimum results, and it provides specific setup templates for many of the currently popular scanners on the market. In its present incarnation, *NoteScan* only recognizes notes, clefs, time and key signatures, barlines and accidentals. No text is recognized, and all dynamics, fingerings, articulations, and pedaling must be added to the score later.

I tried scanning a complex piano piece using *NoteScan* and was not thrilled with the results. I chose this piece because it contains grace notes, clef changes, complex rhythms (6 against 9), and loads of accidentals; in other words, it's typical piano music. The resulting score was far from perfect. Many stems must be flipped, grace notes and some clefs are incorrectly recognized, and some notes are missing altogether. In this particular example, it would be easier to start the whole process from scratch.

I tried scanning a few other examples and achieved even less success. One TIFF image was a professionally hand-copied piano part, which did not translate at all. In all fairness, Musicware does not claim that NoteScan works with anything other than professionally published sheet music. In certain cases, *NoteScan* might help you input some of the bare basics of a score into *Nightingale*. However, OCR technology in general, and *NoteScan* in particular, have a long way to go before the process is effortless.

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See other side for details.

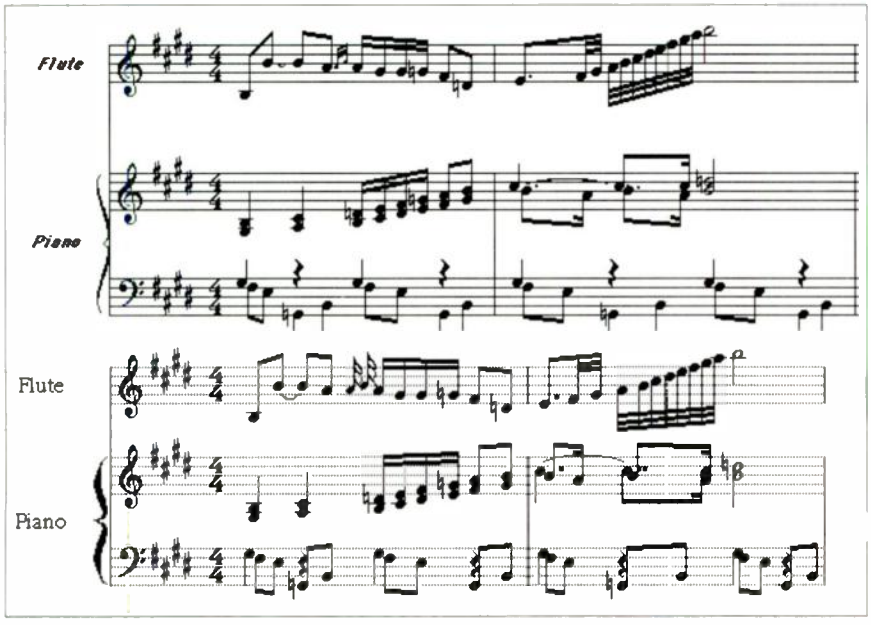

FIG. 5: The upper score is an excerpt from a *Finale* ETF file that was imported into *Nightingale*, the result of which is displayed in the lower score. Although many score attributes are preserved, a certain amount of editing is required for a finished look.

an ETF file created in *Finale,* touching it up in *Nightingale,* and posting it to the Internet. *Nightingale* does not currently support the Notation Interchange File Format (NIFF), but Musicware intends to include this in a future version.

LIBRETTO

The text editor is used to input lyrics, headers/footers, annotative text, and guitar fretboard symbols. Practically anything can be treated as a text item, which can then be attached to events in the score; as the score is adjusted, the item moves accordingly. There are numerous parameters for creating and adjusting text events, and the position and orientation of practically every symbol in the score can be numerically tweaked with the Get Info command.

Lyrics are handled in much the same way as other programs. Words are entered in the text editor, and you must make sure there are hyphens between syllables that are attached to different notes and underscore characters for melismas (syllables that include several notes). The Flow command attaches the syllables to the notes in the score, which automatically respaces to accommodate the text. Depending on your score and the amount of lyrics, you might have to do some manual respacing of measures and occasional reformatting.

Changing the lyrics poses no problem. The score automatically readjusts for the new text. There is extensive online documentation for the text editor, and it works well overall.

PAGE LAYOUT

Notation programs live or die based on how they handle page layout and part extraction. In *Nightingale,* extracted parts become new, separate files that are not dynamically linked to the score. Any changes to the part (such as a corrected note or misspelled word) are not reflected in the master score. This is relatively common in notation programs, but it is often possible to link the parts to the score. I raised this issue with a Musicware representative who said the company is considering this feature for the future, but there is some concern about its impact on performance and speed.

When extracting parts, you have the initial choices of extracting all parts or just the currently selected part; immediately saving the parts as new documents or leaving them as unsaved, open documents; and reformatting the parts or leaving the systems as defined in the master score. In addition, two parts in the score can be extracted to one separate part with relatively few steps.

Each part inherits titles, text, styles, measure numbers, and most attachments from the master score. You can control the amount of measure respacing that occurs after the part is extracted, and you can also preserve the score's system breaks if desired. *Nightingale* automatically creates multibar rests from consecutive whole-measure rests in the score. This extremely convenient feature can be configured to account for double bars, repeat bars, time and key signature changes, and rehearsal marks.

It is a simple matter to create individual parts from a score staff that contains two voices (e.g., first and second oboe). The Look At Voice command highlights the desired voice (which is similar to creating additional voices in the score), after which it can be extracted. It is also possible to extract several score parts to one individual part.

If you want to create small marching-band parts, save a copy of the master score and reformat the staves, pages, and so on to the desired size of the parts. When you extract parts from the score, they are automatically formatted with the proper staff and paper size. This beats reformatting each part individually.

Reformatting systems in the score and respacing measures in extracted parts can be extremely time consuming. However, *Nightingale* offers an extensive

Product Summary

PRODUCT:
Nightingale 3.0
notation software
PRICE:
$495
$295 for educators
SYSTEM REQUIREMENTS:
Macintosh Plus or better; 2.5 MB RAM (System 6.0.5 or higher) or 4 MB RAM (System 7.0 or higher); 6 MB of disk space
MANUFACTURER:
Musicware
tel. (800) 99-PIANO
or (206) 881-9797
fax (206) 881-9664
e-mail 75162.433@compuserve.com
Web www.halcyon.com/musicware
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FEATURES	● ● ● ◖
EASE OF USE	● ● ● ●
DOCUMENTATION	● ● ● ● ◖
VALUE	● ● ● ◖

parameter set that should even make most professional engravers happy. In addition, the package includes an application called *NightCustomizer*, which lets you redefine *Nightingale*'s key commands. Separate *NightCustomizer* Preference files can be stored to keep different setups readily available. If you are used to operating another notation program, you can configure *Nightingale* to operate in a similar fashion by modifying the key commands.

If you have a PostScript printer, you should purchase the Sonata PostScript font separately, although excellent results can be obtained on non-PostScript printers using Adobe Type Manager (ATM). There is presently no support for TrueType fonts, but support is planned for the future. In addition, Musicware will soon offer a new custom PostScript and screen font that emulates the look of professional, hand-copied notation.

CLOSING THEME

Nightingale 3.0 is a first-rate, professional notation package. Considering how much information there is to manipulate when producing high-quality notation, Musicware has done a remarkable job of striking the balance between extensive features and ease of use. Of all the notation software programs I've used, *Nightingale* is the least intimidating, especially in the beginning. The online manual is extraordinarily helpful in finding answers to questions, and there is another small but useful help file in the Apple menu.

▼

Nightingale 3.0 is a first-rate, professional notation package.

Of course, no software is perfect. My wish list for *Nightingale* includes extracted parts that are dynamically linked to the master score, user-definable screen-view scaling (rather than presets), Scroll View in addition to Page View, user-definable staves, previous-key cancellation, and guitar tablature. I would also like more immediate access to the complete Sonata font from the Tool Palette, an index in the tutorial manual, and a countoff for real-time MIDI recording.

All those considerations aside, *Nightingale* 3.0 is a winner. You can get up and running in relatively short order, and the complexity of the program is camouflaged by its well-organized design. The program runs quickly, even on slower computers, and the more you dig, the more capabilities you find. The added bonus of Internet publishing demonstrates that Musicware is looking toward the future.

Can you make professional, engraver-quality notation with *Nightingale* 3.0? For the most part, yes. There will always be engravers who want more, but, for the rest of us, it's almost all here.

Composer/producer **Rob Shrock** *is the keyboardist/arranger for Dionne Warwick and Burt Bacharach. He can be reached through Avatar Productions at avatarprod@aol.com.*

Aphex Model 109

By Rob Shrock

A warm, cuddly parametric EQ joins the Tubessence family.

In the ongoing, trendy quest for tube-colored audio, practically everyone is creating boxes that contain glowing filaments. Although many people wish to add the mild distortion of tubes to digital tracks, the noise generally induced by tubes is not always as welcome. Fortunately, new design approaches have led to hybrid tube/solid-state devices that provide digital control of components, quiet signal paths, and controlled tube distortion.

Aphex has been at the forefront of recent hybrid tube/solid-state designs with its Tubessence line of processors, including the Model 107 microphone preamp and the Tube Expressor compressor. (We checked out the Model 107 in "Hot Stuff" in the February 1996 **EM**; the Tube Expressor was evaluated in "Squeeze Boxes" in the July 1996 issue.) Both of these units have been well received in a variety of studios and live venues. Given its quality and the success of its predecessors, the 1U rackmount Model 109 4-band parametric equalizer appears certain to find a home in commercial and project studios, musicians' rigs, and sound-reinforcement systems.

An argument can be made that tube hybrids just don't sound the same as completely tube-based processors because the tube isn't being driven as hard. Assuming that's true, though, and that the hybrids sound good, their subtle coloration could make them a better choice for certain applications than pure tube designs. And the 109 sounds very good indeed.

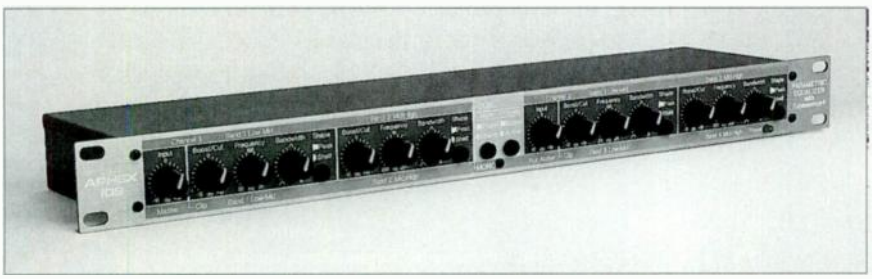
FLOOR PLAN

Using the Mode button, you can configure the 109's four bands as two bands on each L/R channel (Dual mode) or as four bands on one mono channel. The Dual-mode channels cannot be linked in stereo. In Dual mode, each channel has overlapping, sweepable low and high bands with variable bandwidth between 0.2 and 2 octaves and ±15 dB of boost/cut. The low band ranges from 20 Hz to 2 kHz, and the high-band range is 200 Hz to 20 kHz. In Mono mode, the bands are the same; you are simply given two each of the high and low bands.

Each band has a Shape switch that sets the band as a peak or shelving filter. Peak mode is the familiar arrangement where boost/cut is applied at a specified center frequency. In Shelf mode, the bandwidth control becomes inactive, and the Boost/Cut control affects all frequencies below the low band's cutoff frequency (active highpass) or above the high band's cutoff (active lowpass).

The master input-gain control for each channel is center-detented at unity, providing ±10 dB of gain change at the input to the processor. This is helpful for bringing up low-level signals or reducing the input when clipping is introduced by extreme equalization. A dedicated Process/Bypass switch is also provided on the front panel.

The unit's back panel consists of four ¼-inch TRS jacks (which work with balanced or unbalanced lines) for input and output to both channels. Each channel can be switched between -10 dBV and +4 dBu operation for proper



The Model 109 incorporates Aphex's sweet-sounding Tubessence hybrid tube technology into a flexible, 4-band parametric EQ. You can use all four bands on one channel or switch the unit to dual mono with two bands per channel.

matching of levels between devices. This is a good design decision because it allows for two separate signals with different operating levels to be processed simultaneously. When in Mono mode, the channel 2 input-gain control and the Boost/Cut control are deactivated. Power is supplied via a wall wart.

FOUNDATION

When the Process button is engaged, the Model 109 runs all audio through the Tubessence tube stage, which cannot be bypassed separately from the equalizer section. Setting all input and boost/cut knobs to the center-detent position (0 dB boost/cut) operates the Model 109 strictly as a subtle tube-coloration processor, which sometimes is all a track needs in the way of sonic spice.

As an illustration, while working on a grunge-type tune, I had previously cut some solid-state distortion guitar tracks through a cabinet simulator straight to ADAT. Simply running the tracks through the Model 109 with no EQ engaged gave them presence that wasn't there before. The guitars had a little more body and perceived loudness, although the level meters did not change between the processed and unprocessed signals. The effect was subtle but noticeable.

"If less is more," I mused, "perhaps more is even more." (Hey, I'm a creative musician, not a logician.) I decided to deliberately run the input gain into clipping by cranking the gain knobs just to hear what would happen. I didn't know whether the solid-state side or tube side of the Model 109 was clipping, but the resulting sound it produced was great for these nasty guitars! Although I had to run the tracks through another gain stage to lower the level before going back to ADAT, I reprinted the new guitar sounds to a slave tape and ultimately used these tracks in the final mix because they sounded so good.

▼

The 109 provided the right balance of tube coloration and precise control.

CONSTRUCTION

If the Model 109 is good as a mild tube-coloration processor, it is great as an equalizer. The equalization can be anywhere from tasteful to downright ugly, as you please. Unlike some other equalizers I've worked with, a small amount of boost or gain with the Model 109 did the trick for most applications unless I wanted to get extreme. The Aphex EQ is nice and quiet, too. I only had a problem with additional noise when I was pushing the top end extremely hard.

The EQ really shines when applied to a vocal track in Mono mode. Having four bands of sweepable frequency and bandwidth allows you to engage one of the low bands as a shelving high-pass filter to get rid of all the junk below 60 Hz, use a high band as a broadband filter to add some "air," and

still have two more bands available for general equalization. At the push of a button you are in Dual mode, where you can EQ stereo drum overheads or stereo keyboards with two bands on each channel.

I also used the Model 109 for mastering song demos and got excellent results. After mildly compressing the finished stereo mixes, I ran them through the Model 109. This gave me a final opportunity to adjust the frequency content and better match the sonic nature of each song to the other, adding a few more bottom-dwelling lows and some top end where appropriate.

Mastering your own product has its disadvantages, of course. For one thing, you can't draw on the know-how of a great mastering engineer, who also brings another set of experienced ears to the project. And you rarely have access to the state-of-the-art equipment used on a major-label release. Nevertheless, I was able to heighten the impact of the song demos by tweaking them out a little more with the 109. The Aphex EQ provided just the right balance of additional coloration from the tubes and precise equalization control without destroying the integrity of the much labored-over mixes. In this application, I appreciated the subtle nature of the tube stage.

Given that you can't precisely control the amount of tube distortion, I think that Aphex was wise to keep it mild. However, if you are mostly looking for a tube front-end for a digital recorder, you might want to check out units with more control of the tube-coloration effect.

DECORATIONS

The Model 109's user manual provides plenty of helpful tips throughout on general equalization theory and practice, which should prove useful to experienced owners as well as novices. The manual also displays a diagram of the knobs' frequency, bandwidth, and boost/cut settings that is more detailed than the silkscreening on the front panel. That's nice, but I wish the same parameters were actually printed on the unit itself. For instance, in addition to the 200 Hz, 1 kHz, and 20 kHz settings shown around the Mid-High Frequency knob, the manual also displays the positions for 400 Hz, 2 kHz, and 8 kHz.

In fact, my biggest gripe with the Model 109 has nothing to do with its sonic characteristics. The processor sounds great. But the layout of controls on the front panel caused problems for me because the four bands are divided up from left to right: low, high, low, high.

In Dual mode, each channel is controlled from one side of the unit (providing a high and low band on each side), which is fine when you are processing two separate signals. But this arrangement is inconvenient when equalizing stereo program material. Because the Frequency and Bandwidth settings are constantly variable, it is critical that the knobs for left and right

Model 109 Specifications

Inputs/Outputs	2/2 (¼-inch TRS)
Operating Levels	+4 dBu, -10 dBV (switchable)
EQ boost/cut	±15 dB
Frequency Range (low-mid band)	20 Hz–2 kHz
Frequency Range (mid-high band)	200 Hz–20 kHz
Bandwidth control	0.2–2.0 octaves (at full boost, Peaking mode)
Frequency Response	10 Hz–30 kHz (±0.1 dB)
Dynamic Range	108 dB @ +4 dBu/105 dB @ -10 dBV
THD	<0.15%
Input Gain	±10 dB
Hum and Noise (10 Hz–22 kHz, unweighted)	-86 dBu @ +4 dBu/-92.8 dBu @ -10 dBV
Crosstalk (10 Hz–22 kHz @ max. output)	-59 dB @ +4 dBu, -56 dB @ -10 dBV
L/R Stereo Link	no

channels be precisely matched. But the Model 109's design makes such important comparisons more difficult than they need to be.

If you are equalizing the low end of a stereo piano, for instance, it makes sense to have the low bands for each channel next to each other for easier matching of left- and right-channel settings. I lost count of how many times I screwed up my settings because I grabbed and twisted the wrong knob while equalizing stereo material, even after weeks of use. Furthermore, in Mono mode the two low and two high bands are intermixed. I think a low-low-high-high layout would be more logical.

All the knobs on the front panel look exactly alike. As I mentioned earlier, the silk-screening is only adequate. The Band designations, for example, are written in white on a lavender background—not an outstanding color combination. If I had owned the review unit, I would have taken colored felt pens to the knobs the first day. Perhaps Aphex can color-code the knobs by function in future production runs. If not, it's time to buy some colored Sharpies.

Product Summary

PRODUCT:
Model 109 parametric equalizer
PRICE:
$449
MANUFACTURER:
Aphex Systems
tel. (818) 767-2929
fax (818) 767-2641
e-mail sales@aphexsys.com
Web www.aphexsys.com
CIRCLE #442 ON READER SERVICE CARD

EM METERS	RATING PRODUCTS FROM 1 TO 5
FEATURES	● ● ● ◖
EASE OF USE	● ● ●
AUDIO QUALITY	● ● ● ●
VALUE	● ● ● ◖

OPEN HOUSE

The Aphex Model 109 is a winner, housing two useful processors in one unit. In addition to flexible equalization, you get the added bonus of a tube stage to fatten up sterile digital tracks or anything else that needs a little dose of character. Although I don't agree with the front-panel layout of the unit's bands, it won't prevent you from creating some great sounds that would be at home on any recording or in a live setup.

There's no doubt one could wish for more control over the tube section. The tube distortion is very mild, the tube cannot be bypassed, and the distortion is not user controllable. However, the Model 109 imparts a more vintage-style sound to your tracks than a standard solid-state equalizer would, without going over the top and producing massive coloration.

As an equalizer, the Aphex Model 109 is excellent, coupling extremely quiet operation with detailed control of frequency content throughout the audio spectrum. The ability to use the unit as a 2-band, dual-channel or 4-band, single-channel equalizer further strengthens the Aphex EQ's position as a versatile, useful tool. Roll out the welcome mat, invite one in for a spell, and get to know it for yourself. ●

bizarre-sounding acoustic guitar track by running the signal through a Leslie speaker cabinet. They documented the swirling sound with three mics: a pair of AKG C 414s positioned towards the top of the cabinet to capture a stereo perspective of the Leslie's rotating horns and an AKG D 112 placed near the speaker bottom to grab some bass frequencies. In the final mix, Luekens panned the two top mics hard left and right and the bottom mic in the center. "It sounded really strange and wonderful," he attests.

LIFE SAVERS

Okay, so how do you record a great acoustic guitar sound if you *aren't* lucky enough to book an excellent player with a superb guitar? Well, it's true that a polished you-know-what will never shine, but it may still be possible to salvage a third-rate performance on a bum axe.

For example, the sound of dead strings can be brought back to life a bit with some compression and EQ. Start with a 2:1 compression ratio to raise low-level signals and smooth out the sound. Then, depending on the tempo of the song, experiment with a medium to slow release to add sustain. Watch out for bass signals, though, which are likely to increase disproportionately as you add compression. (It may be necessary to dial out troublesome low frequencies with console EQ.)

You can also increase sparkle with a spectral enhancer (see "Enhancing the Spectrum" in the November 1996 **EM**) or by boosting the mid and high frequencies a few decibels somewhere between 5 kHz and 10 kHz.

A de-esser is another useful device for salvaging acoustic guitar tracks. This device is designed to attenuate sibilance on vocal tracks but can also be set to diminish annoying fret buzzes. Finger squeaks are difficult to tame, although an outboard graphic equalizer may be able to notch out some of these distractions. Remember, reverb can *emphasize* squeaks and buzzes, so go extremely light on the effect sends during the mixdown. To a limited extent, it is possible to use chorus and pitch-shifting effects to correct bad intonation or out-of-tune notes, but these tricks should be used as a last resort because the results on acoustic instruments are rarely very natural sounding.

LET IT RING

The only foolproof recipe for great acoustic guitar tracks is this: a great guitar, a great guitarist, an acoustically great room, one or two great mics, a great set of ears (yours), and a clean signal path. Throw in a great compressor and you're in business (or else you're in the wrong business). But even with less-than-perfect ingredients—such as a poor guitarist flailing away on a balsa-wood acoustic—you can produce an "edible" track by employing the same care and sonic sense that you'd bring to an ideal session. Just listen carefully, be open to experimentation, and do whatever it takes to make the track jangle and sparkle. If all else fails, try to get a hold of Chet Atkins' home telephone number! ●

to know what type of project it is, meaning CD-ROM, Web page, and so on; how long they plan to have it in the market; whether it will be distributed in the United States only or worldwide; and, with regard to music, whether they want to use an original composition or rerecord a composition. These are all crucial pieces of information to have figured out before we enter into any clearance negotiations."

When dealing in areas such as multimedia and the Internet, negotiations can get more complicated because no standards exist yet. "There is no set rate for this new-media usage," says Alofs. "If you're going to use music in traditional film, video, or television, where the unions have their rates, it's pretty much set, cookie cutter–type fees. There's not much precedent for Web pages or other new media, so in one sense, it's very difficult for developers because they don't know what they have to clear or how much it's going to cost. The positive side of the lack of precedent is that you can be very creative in your deals."

Alofs points out that most developers don't think these types of details through before looking to clear any audio or film clips for use in their projects. Developers also err by not starting the clearance process early enough or budgeting enough money for use of the clips. By working on the clearance issues at the outset, you'll know what you can and can't use and how much it will cost you.

"In regard to new media, just because there aren't set standards doesn't mean you're not responsible for knowing what the rights are," cautions Alofs. "You are responsible for not infringing on anyone else's rights—that is, using material in a way that it was not originally intended to be used. And if you're reusing the material in your project, that's probably not the way it was originally intended. Unfortunately, these issues are usually defined through litigation, and I certainly don't want my clients to be the ones defining all these rules. So I think it's better to be conservative and go to every rights holder that you need to and cover all of your bases. If you plan to stay in this industry, you're better off creating good long-term relationships with publishers and labels. It's a small industry, and you're probably going to come back to the same people whether you like it or not. You're just better off acting in good faith."

Mary Cosola *is managing editor of* **EM.**

MAKING CONTACT

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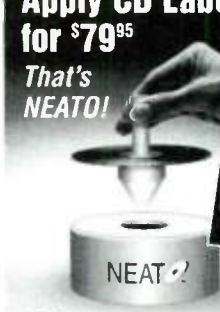

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TECH PAGE

When the SHARC Bites

This processing technology keeps its teeth pearly white.

By Scott Wilkinson

One of my first interviews as a music-technology journalist was conducted with film composer Gary Chang. When I asked him to speculate on the future of electronic-music tools, he described a hypothetical box that could perform various forms of synthesis and signal processing. You would download the appropriate software into the box, which would turn it into any type of synthesizer or signal processor you wanted.

Ten years later, Chang's remarks have proven to be prophetic. Analog Devices (tel. 617/461-3881; fax 617/461-3010; Web www.analog.com) introduced a new technology at the recent AES convention that reflects his vision with almost uncanny accuracy.

The hardware is the ADSP-2106x SHARC (Super Harvard ARchitecture Computer) DSP chip, which offers 32- and 40-bit, floating-point processing as well as 32-bit, fixed-point operation. It can perform 40 million instructions per second (MIPS) and sustain 80 million floating-point operations per second (MFLOPS); peak floating-point performance can reach 120 MFLOPS. With an instruction cycle of 25 ns, the SHARC can execute a 1,024-point, complex FFT in less than half a second.

The chip also is available with two or four megabits of onboard static RAM. Ten direct memory access (DMA) channels facilitate background data transfers without impacting the performance of the processor core. Six link ports allow designers to combine several SHARC chips for parallel processing, and two synchronous serial ports each support a throughput of 40 Mbps.

The SHARC chip has been around for a couple of years. The real news is the software, which is based on a real-time version of Csound, an academic music-programming language developed by Barry Vercoe at the MIT Media Lab. Vercoe spent two years reshaping Csound for real-time operation. The new version includes over 300 "modules" called *opcodes*. These opcodes provide various forms of synthesis, including subtractive, additive, sample playback, physical modeling, FM, and granular. All types of synthesis can be implemented simultaneously with a single SHARC chip, which provides a maximum of 64-voice polyphony and 64-part multitimbral operation, depending on the complexity of the selected synthesis methods.

Other opcodes provide many types of real-time signal processing, such as reverb, delay, chorusing, flanging, filtering, and mixing. More sophisticated effects are also available, including phase vocoding and waveshaping. One particularly interesting effect is formant-based pitch shifting/harmonization. Among other applications, this can be used to conform the pitch and rhythm of virtually any input signal to a melodic line within a MIDI file. Finally, there are many opcodes that provide extensive MIDI control.

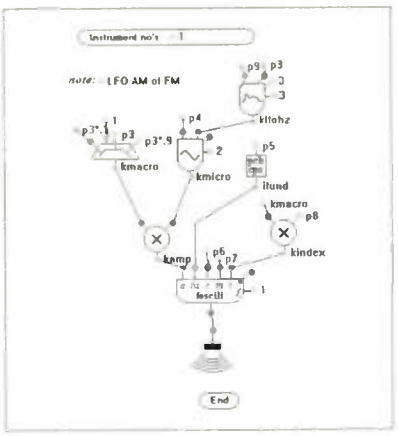

FIG. 1: *Patchwork* lets you assemble a Csound Orchestra with graphic modules.

To use the SHARC as a musical device, you must supply two pieces of information called the Orchestra and the Score. The Orchestra is a collection of opcodes that produce and modify sounds; this includes various forms of synthesis and signal processing as well as the desired response to MIDI messages. The Score, which can take the form of a MIDI file, tells the system what to play.

You can assemble an Orchestra using a PC program called *Patchwork*, which was developed by Keith Lent and Russell Pinkston at the University of Texas, Austin. (A shareware version of *Patchwork* can be downloaded via ftp from indigo.pac.utexas.edu /pub/music; for more info on the program, contact Pinkston at r.pinkston @mail.utexas.edu.) This software resembles Digidesign's *Turbosynth:* you connect various graphic modules (which represent the opcodes) on the screen to create an Orchestra (see Fig. 1). The software then compiles the data into Csound object code, which is downloaded into the chip. Clearly, this process brings Gary Chang's vision a giant step closer to reality.

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The DMTi was also designed to interface with Kurzweil's proprietary 8 channel digital protocol KDS-Kurzweil Digital Stream (offered as an upgradeable option for the K2500 series). The DMTi can perform sample rate conversion (in real time) on up to 4 stereo pairs of incoming digital data while acting as the master or slave clock; the DMTi can transmit 44.1K or 48K clock and can transmit or receive BNC word clock.

This device is well suited for use with popular digital mixers such as Yamaha's O2R, or Korg's Soundlink, or as a translation device from MDM to Digidesign's ProTools systems. The DMTi allows many different digital input formats to be user routed to a variety of digital output formats and sample rate converted. The Alesis and Tascam option cards are needed for conversion to and from these popular MDM formats. The DMTi can be seen and demonstrated at your local Kurzweil dealer.

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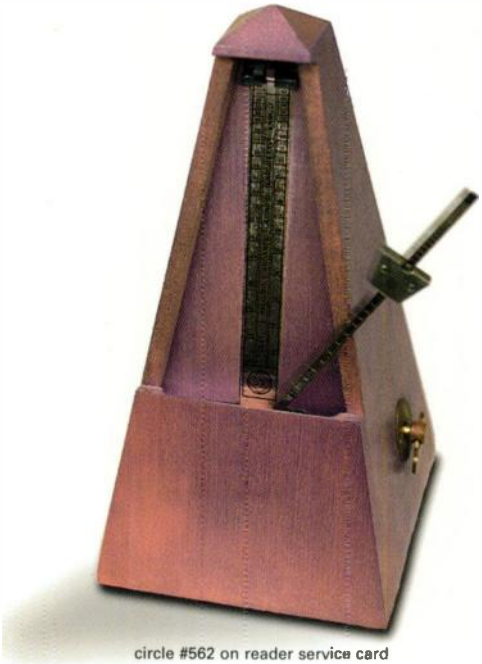

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