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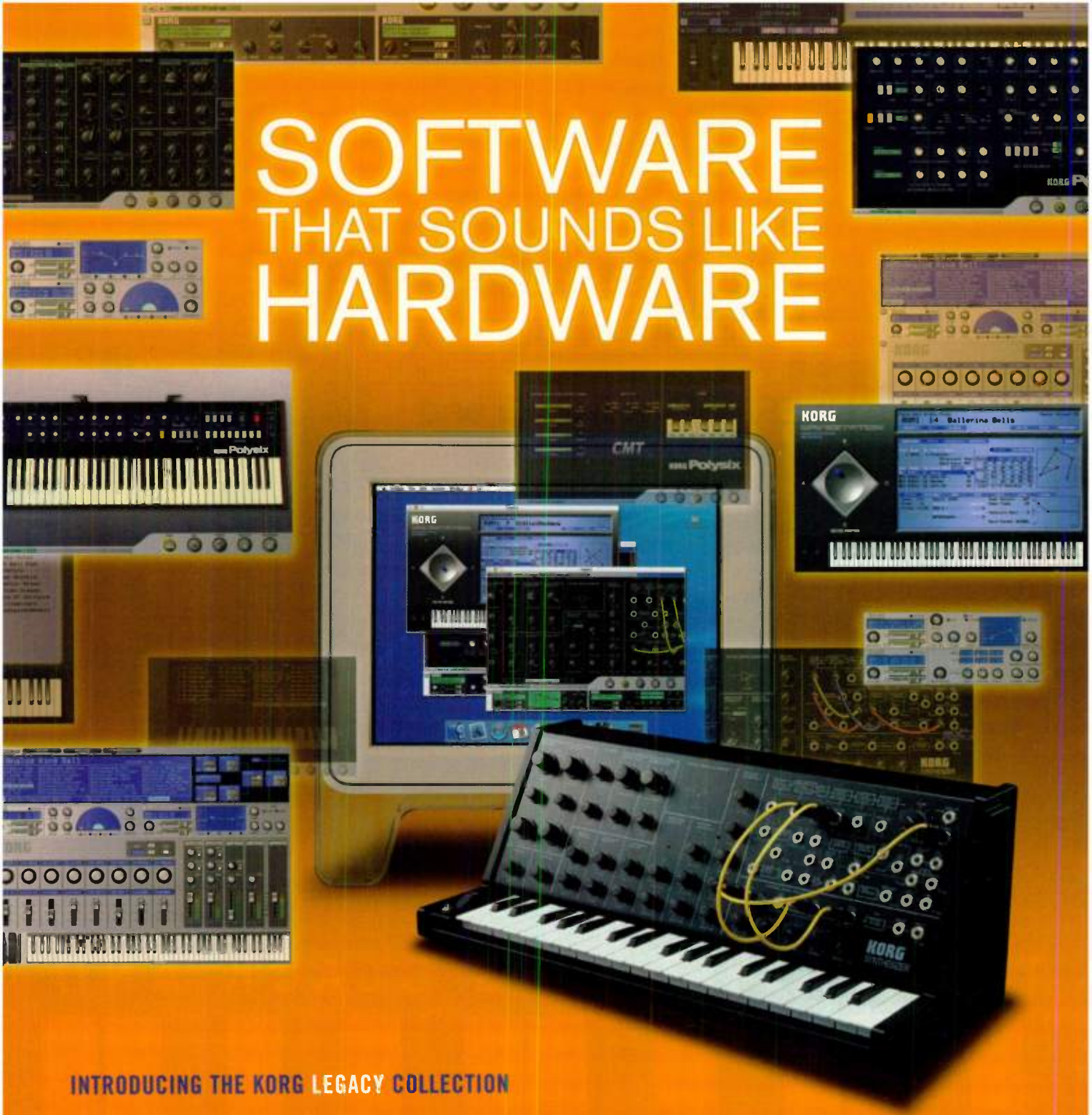

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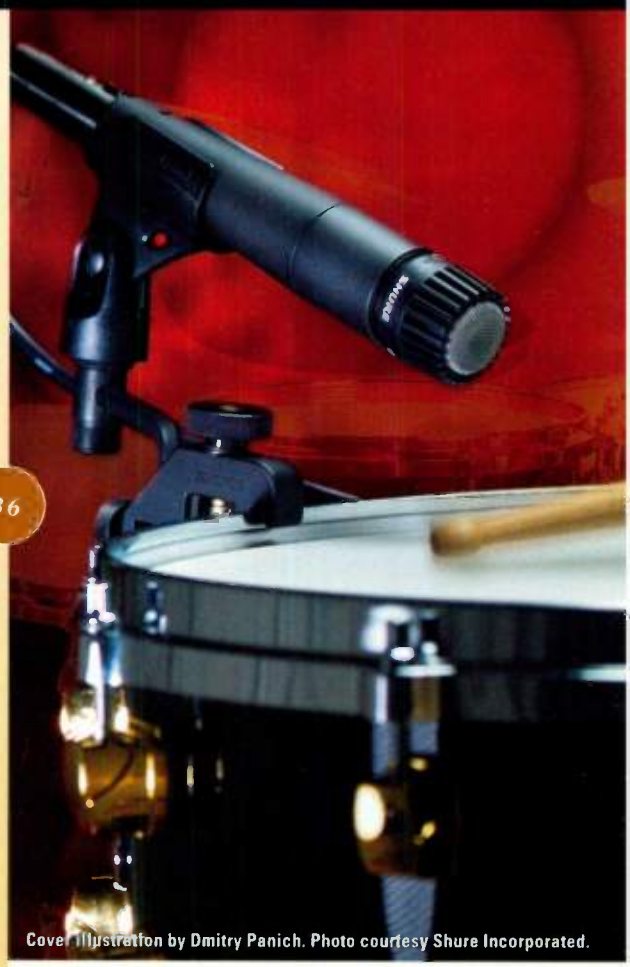
I N S I

FEATURES



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DEPARTMENTS

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Electronic Musician

JULY 2004 VOL. 20, NO. 8

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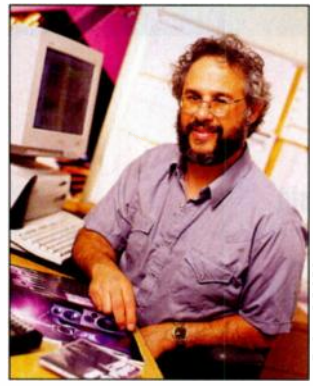
COLUMNS

REVIEWS

Get It While You Can


ANTHONY PIDGEON

In a world where many of us must perform multiple tasks at once, computers help organize those tasks and store vast amounts of information, enabling us to communicate quickly and easily. And that's before we hit the Record button and use our computers to work those sonic miracles that we could never create with recording tape and a razor blade.

Because we can do all these things, people increasingly try to do more of them faster, often attempting to work on many tasks at once. But human multitasking, as Larry the O points out in this month's "Final Mix" column, can become so emotionally and physically taxing and inefficient that it can lead to disaster.

In this month's feature story, "Keep It Simple," author Nick Peck offers practical and philosophical responses to some of the problems Larry points out, as applied to working in a personal studio. In essence, Peck explains that while you might have to keep many balls in the air, sometimes you can reduce the number of balls, not toss them quite so high, and generally make juggling them simpler and easier.

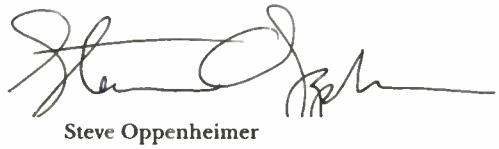
Learning to eliminate nonessential tasks and streamline workflow is very important. Facing up to the dangers of multitasking is crucial, as well. But we also have to make more fundamental choices about how many things we should do and, perhaps more important, what sorts of things we should do in our limited time. People's answers will all differ, of course, and they will probably change at various times in each individual's life.

For example, when I was learning my trade as a magazine editor I worked absurdly long hours each day, and then went home to do more of the same late into the night and right through the weekends and holidays. I went way beyond what I needed to do to earn my paycheck, and it was rewarding in many ways. I've worked hard all my life, but at this time I became a workaholic. Friends and family (including Larry the O) warned me about it, but I loved my work and ignored the danger. Over the years, the nature of my job changed, and I learned to work faster and more efficiently. However, with a few exceptions, my life was still almost entirely devoted to my job.

Then, a few years ago, a close friend (and EM author) died unexpectedly at age 38. I recalled my paternal grandfather's dying words to me many years ago: "Get it while you can; it won't last long." You never know how much time you have left.

So I finally admitted to being a workaholic and changed my lifestyle. It was the best decision I've made in a long time. I remain devoted to EM and to our new magazine, MET. I still work hard, but I have reclaimed my evenings and weekends and don't work on holidays. I pursue a wide variety of personal projects.

I now seek the balance that Larry discusses in his column. Grandpa O, always a man of few words, provided the answer in one sentence.



Steve Oppenheimer
Editor in Chief

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K2661 Evolution of a Species

Kurzweil's newest member of the K2600 family is more powerful than ever. The K2661 truly has it all: Kurzweil's award-winning V.A.S.T. synthesis, mind-blowing effects, KB3 Organ Mode, Live Mode, and Triple Modular processing are joined by exciting new features including built-in 8-channel ADAT I/O, a SmartMedia™ card slot for storing patches and samples, 24-bit D/A, and General MIDI mode. Program memory expansion and 128MB of sample RAM* are also included as standard. All in a very portable, 61-note, synth-action keyboard. Now, you can take it with you!

And of course, the K2661 sounds amazing. With the Orchestral and Contemporary ROM options built-in, the K2661 features the new "Best of V.A.S.T." sound set, which is also available for download for K2600 users. The Stereo Dynamic Piano and Vintage Electric Pianos ROM options are available for both K2661 and K2600 users. The optional sampling card is also available.

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* Note: Sample RAM is standard with models sold in the USA only.

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Immediate

The ISA 428 Pre Pack features four classic transformer-based microphone pre-amps with switchable input impedance, and Focusrites latest eight channel, 192 kHz A/D converter option - the perfect future-proof interface for your DAW or digital console.

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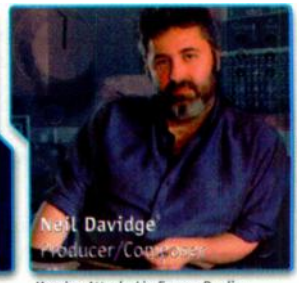

Massive Attack, Liz Frazer, Prodigy

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LIZABETH HEAVERN

THE EARS HAVE IT

Thank you for the article "Ears to You!" in the May 2004 issue. I ask that all musicians please heed Dr. Charles Limb's advice.

I have permanent hearing loss due to chemotherapy for cancer. I lost hearing from 4 kHz on up in January 2002. Within 24 hours, my musical and professional life was literally changed. As I am now cancer-free, I have no regrets, but hearing loss is what musicians can expect if they do not take precautions.

Protect your most valuable instrument and turn your music down.

Bill Bogus
via e-mail

LIKE A SURGEON

I've just finished reading your interview with BT ("Production Values: Sonic Surgeon," May 2004). I write for the UK magazine *Computer Music* and have interviewed industry heavyweights. My interviews were always looked over by an editor so that any potentially incendiary and ignorant comments wouldn't make it to final publication.

I'm referring specifically to BT's comment regarding PCs: "I detest PCs, but they're kind of a necessary evil." Is this an ad for Apple or a supposedly open-minded interview in a major music magazine?

BT's opinions regarding the superiority of Macs are exactly the opposite of my experiences running Emagic's Logic under Windows XP. My PC is insanely fast, powerful, and reliable. With a Mac, I worry about whether the system will crash and whether I can run as many effects and soft synths as I can on my PC. But do I write articles in which I proclaim that I detest Macs? No, because even if I were stupid enough to blurt out such an ignorant statement, my editor would delete it—leaving my public persona of charm, class, and intelligence intact.

Shame on you, BT, for using your highly influential public platform to propagate such a negative opinion that thousands of young readers will think is the truth because *you* said it.

Ted Perlman
Los Angeles, California

Ted—Although I disagree with BT's statements about PCs and with your statement about Macs, I have no problem with publishing both. We're not going to engage in an extended Mac versus PC debate, because I think that's pointless. But neither of you made racist remarks or something similarly offensive that we would refuse to publish. We asked BT for his views, and he gave them. You offered your views, and we published them. Enough said.—Steve O

I haven't heard BT's music, so I'm in no position to critically assess his productions. But after reading the May 2004 interview, it is hard to doubt his technical skills. He obviously has an exceptional aptitude for using the most sophisticated technology available for music production.

But what an egomaniacal blowhard! BT has "literally thousands of soft synthesizers. That's not an exaggeration." And he says he uses all of them. Roll over Beethoven and move over Stockhausen: the future of music is BT!

Then he raves that all the technological advances in home-studio gear will lead to "better music . . . than we've ever heard before." I suppose then that one might conclude that Aphex Twin and Moby have consigned the Beatles, Charles Mingus, and Debussy to insufferably quaint obsolescence! I suggest going to your local record store and looking through the "electronica" bins to see what vast quantities of "better music" are now being produced by any half-wit joker who can get his hands on a DAW or a sampler.

Give me a break—or at least, some semblance of humility.

Ferrara Brain Pan
San Francisco, California

It is unfortunate and surprising that *Electronic Musician* did not question BT's assertion that hiring "an A-list string section for three hours . . . costs $200,000 to get the best players." It's ironic that such a wildly inaccurate figure was published in an article devoted to "surgical precision."

Under the various provisions of the Low Budget Sound Recording, Limited Pressing Sound Recording, TV/Videotape, and Low Budget Motion Picture contracts (to mention just a few of the AFM agreements available), a large to very large "A-list" string section can be hired for a fraction of the figure mentioned by BT.

As a member of RMALA and Professional Musicians Local 47 who has worked as a composer, an orchestrator, and a conductor with Los Angeles

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musicians for more than 20 years, it is vexing to read such wildly inaccurate reporting at a time when anyone with an Internet connection can discover the facts for themselves.

Bruce Babcock
Glendale, California

Bruce—Thanks for pointing that out. I understand your frustration with BT's statement. However, to term it "wildly inaccurate reporting" on the part of EM *is missing the point. It wasn't something we reported; it was a direct quote in an interview story. I think BT's point was simply that he uses real strings when he can—when the studio pays for them—but that it isn't cost-effective to do so when he's producing his own music for his CDs.—Mike Levine*

SOLID PITCHING

Sean D. Carberry's article ("Recording Musician: The Winning Pitch," May 2004) couldn't be finer if I'd written it myself. I too have shifted from total damnation to reserved endorsement of intelligent pitch correction.

I am horrified by the "always on" approach. It strips the character out of a solo voice for the sake of correctness, and makes individual group voices less distinguishable from person to person. It also eliminates the subtle phase intermodulation that occurs naturally between tracks and adds interest to a mix. On the plus side, I have noticed that artists who have a limited range of expression will take more chances with a melody knowing that the machine will rescue their sour notes.

Having said that, I am happily using the Antares Auto-Tune plug-in. A cheap way to get involved with pitch correction is to download Oberheim's OB-Tune DirectX plug-in, which licenses Antares's basic technology for $39.95 from MusicYo.com.

One other observation: pitch correction works best with minimal vibrato. If you are hoping to pitch-correct someone's constantly wavering trill, your time will be better spent recording takes and comping them down.

Gary Sula-Goff
Wonderwindow Production Studio

BETTER TECH-SUPPORT TIPS

Here at Korg USA's Product Support Center, we read Steve Oppenheimer's "Technical Nonsupport" editorial ("First Take," April 2004) with great interest. It prompted me to describe some things a customer can do to get the most out of technical support, regardless of who they call.

Be in front of the product when you call, and make a note of its operating system while you're there. I can't tell you how many calls we've had in which a customer is calling from a cell phone—often while driving—asking questions about things that need to be qualified at the source. In addition, if the problem exists when the product is used in conjunction with another, set up the other product, as well, and try to bring both to the "problem state" before you call.

Make time. Take your schedule into consideration when planning the call. You'll get more out of the call if you aren't rushed and you have the time to talk and listen at a comfortable pace.

Formulate your questions. Take a couple of minutes to form your observations before you call. Even better, write them down along with your questions. This will help the agent get to the bottom of your problem more quickly. It also helps to document when the problem arose and what you were doing at the time.

Have something to write with. Even if you anticipate what the response to your question will be, be prepared in case there's an "and" or a "but." Remember that you're speaking with a specialist who may have some valuable additional information that you didn't expect. Technical explanations are also sometimes quite involved, and relying on one's memory can be daunting.

Know the model of the product you're calling about. We frequently hear from friends or spouses of users who have a question about a "Korg keyboard" or a "D-series recorder." On more than one occasion we've had callers incorrectly identify their products.

James Sajeva
Product Support Manager
Korg USA, Inc.

ERROR LOG:

April 2004, "Review: IK Multimedia SampleTank 2 (Mac/Win) sample player plug-in," p. 106. The "Minimum System Requirements" sidebar was omitted. The requirements are:

MAC: G4/400 MHz; 512 MB RAM; 3 GB of hard-disk space (5.5 GB for XL); Mac OS 8.6 or OS X 10.2; compatible plug-in host

PC: Pentium 4/1 GHz; 512 MB RAM; 3 GB of hard-disk space (5.5 GB for XL); Windows 2000 or XP; compatible plug-in host

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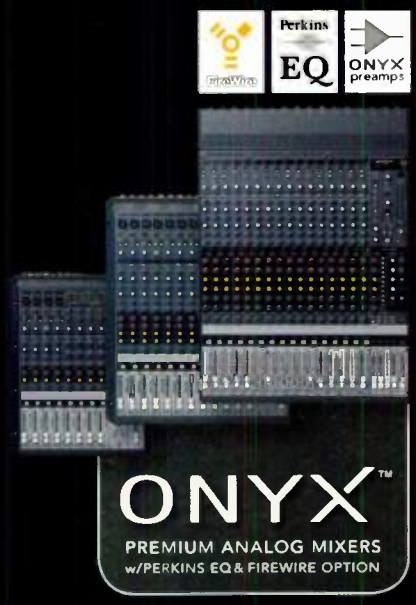
While the other guys were busy adding switches and knobs, we reinvented the whole thing.

GREG MACKIE,
our founding father, shows off a killer shirt... And the Onyx 1220.



Perkins EQ

ONYX preamps

It's time to rethink your mixer. We did.
Introducing Onyx, the next generation of premium analog mixers from Mackie. Over two years in the making, Onyx mixers boast all-new circuitry, incredible sounding Onyx mic preamps, a built-in Talkback section, balanced direct outs on every channel, and our own Perkins EQ, which gives you the warm, musical sound of "British" EQ with a true hardware bypass. And if all that wasn't enough, Onyx mixers also sport a FireWire I/O option – making them superb multi-channel audio interfaces for your Mac or PC.

Put simply, Onyx is the compact mixer, reinvented. Visit your Mackie dealer to learn more.

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DOWNLOAD OF THE MONTH

SYNTHTEST, RAX, AND CHAINER

Ever get the urge to play one of your synth plug-ins without having to go through the rigmarole of booting up your full DAW? Chainer (Win, $60.00), Rax (Mac, $29.95), and SynthTest (Mac, free) are each designed for just that. Chainer is a Windows application that hosts VST instrument and effects plug-ins. Rax and SynthTest host Audio Units (AU) instrument and effects plug-ins in Mac OS X; you can use VST plug-ins with them, too, if you use a wrapper such as FXpansion's VST-AU Adapter (www.fxpansion.com).

Manyetas SynthTest 1.2.1 (www.manyetas.com/creed/synthtest.html) is the simplest of the three to use. It allows you to run one AU instrument and one AU effect. Just open SynthTest, load your plug-ins, select a MIDI source, and off you go. You can load and save presets directly from SynthTest's control panel, which offers no other controls such as volume or pan. However, you can usually open the plug-in's control panel for those functions as well as preset editing. SynthTest is open source and is intended primarily as a developers' tool, but it's an excellent AU host in its own right.

Granted Software's Rax 1.0 (www.grantedsw.com/rax) presents you with a 16-channel rack into which you can install AU instruments, insert effects, and send effects. Menus for MIDI input and audio output let you select among all installed devices. MIDI can be sent to instruments and effects (for MIDI automation), and each item in the rack can receive on any or all MIDI channels. The rack contains a built-in 16-channel mixer; when an instrument is installed, it is automatically wired to the next free mixer channel. Each channel has eight send buses, and multiple effects can be inserted in each bus. Insert and master effects buses are also available. Depending on your audio interface's capabilities, you can even plug multiple channels of live audio into the rack. You can download a free version of Rax that will host a single AU instrument and one mono or stereo audio input. Registration enables full multi-instrument capability.

Xlutop Chainer 1.03 (www.xlutop.com/html/chainer.html) is similar to Rax in that it allows you to play several VST instruments at the same time and lets you

follow them with multiple VST effects plug-ins. It enables you to create complex signal paths, and you can use it to process ASIO audio input. Chainer runs as a standalone application, and VST plug-in versions for instruments and effects are provided as well. Using Chainer, you can create complex effects chains within other VST host applications. Download one of these handy utilities, and when that postmidnight urge to play one of your synths strikes, you will be prepared.

—Len Sasso

NOMAD FACTORY ROCK AMP LEGENDS

This year has been a banner one for computer-toting electric guitarists. Nomad Factory, collaborating with former Aerosmith guitarist Jimmy Crespo, has released a guitar-amp simulator plug-in called Rock Amp Legends (Mac/Win, $399). It provides impulse responses of several British and American guitar amplifiers and cabinets. Rock Amp Legends comes with more than 80 presets that range from clean rhythms to screaming solos, and you can save your own presets. A pop-up menu lets you select from 18 combinations of amp and speaker types.

On the front panel are all the controls typical of a full-featured guitar amp, including knobs for reverb and overdrive. An A/B switch lets you quickly toggle between two amp simulations. Flip the FX switch to reveal a virtual processor rack with controls for stereo imaging, 3-band parametric EQ, compression, noise gate, tremolo, autopan, delay, chorus, and phaser effects.

When you download Rock Amp Legends, you will also receive the Nomad Factory Free Bundle, which is composed of Free-Tremolo, Free-Sweeper, and Free-Phaser plug-ins. Rock Amp Legends requires at least 512 MB of RAM and either a Mac G4/867 MHz with OS X 10.2 or a Pentium III/800 MHz with Windows 98, 2000, or XP. It supports RTAS, HTDM, and VST on both platforms. Nomad Factory; e-mail info@nomadfactory.com; Web www.nomadfactory.com.

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ANTARES AUTO-TUNE 4

Antares Audio Technologies is shipping Auto-Tune 4 (Mac/Win, $399 to $599; upgrades $99 to $149), an update to its best-selling pitch-correction plug-in. The most noticeable enhancement is a revised user interface sporting additional controls and a virtual brushed-metal background. Auto-Tune 4 supports audio rates as high as 192 kHz and provides several new configuration options.

Like previous versions, Auto-Tune 4 offers Automatic and Graphical modes for solving pitch problems. Automatic mode now displays an onscreen keyboard for controlling target notes and their behaviors in individual octaves. A Hold function helps you analyze pitch errors, and new parameters give you more control over vibrato. In addition to expanded editing tools and navigation controls, Graphical mode has a larger Pitch Edit display with a choice of three color schemes. Within the display, Cursor Pitch gives you immediate visual feedback of the cursor position's exact pitch value. Vibrato Scaling lets you change vibrato depth without altering its shape or character. Graphical mode also lets you use Undo and Redo up to 20 levels deep.

Auto-Tune 4 is available for VST, MAS, RTAS, or TDM on the Mac and DirectX or RTAS in Windows. An Audio Units version and Windows VST support should be available soon. Antares Audio Technologies; tel. (888) 332-2636 or (831) 461-7800; e-mail info@antarestech.com; Web www.antarestech.com.

DIGIDESIGN PRO TOOLS 6.4

Digidesign has announced its Pro Tools 6.4 software (Mac/Win). An upgrade costs $150 for registered owners of Pro Tools TDM 5.x or 6.x, and the version is free for Pro Tools LE 6.x owners. For balancing recordings that are made at lower levels, the newest version's +12 dB fader gain offers more latitude than the previous +6 dB maximum. One of the other new features is support for Digidesign's Icon console and the Command|8 control surface. In addition, Pro Tools 6.4 is the first version to organize plug-ins by category in hierarchical menus.

Several enhancements are unique to Pro Tools|HD. Automatic Delay Compensation rectifies delays that are caused by plug-in latency and mixer routing, so that using Pro Tool's mixer feels more like using an analog console. TrackPunch allows you to arm and punch individual tracks on-the-fly. TrackInput enables track-by-track switching between the source and disk playback. RecordLock prevents you from having to re-arm Pro Tools when it encounters discontiguous time code.

Pro Tools 6.4 runs in Mac OS X and Windows XP and must be used with Digidesign hardware. Digidesign; tel. (800) 333-2137 or (650) 731-6300; e-mail prodinfo@digidesign.com; Web www.digidesign.com.

CAKEWALK GUITAR TRACKS PRO 3

Cakewalk has released Guitar Tracks Pro 3 (Win, $209; upgrades $79), a revision that adds new tools and functionality to its multitrack software environment for guitarists. Featuring a redesigned user interface, GT Pro 3 can record 32 audio tracks, supports up to 32 hardware inputs and outputs, and allows up to 32 simultaneous real-time effects. It now supports multiple tempos and meters, and a sync generator lets you work with external recorders and sequencers. Complete automation is available for controlling track parameters and effects.

GT Pro supports VST plug-ins using Cakewalk VST Adapter and DirectX effects, and you can monitor effects while recording. In addition to the GT:FX suite, the package includes IK Multimedia's AmpliTube LE. GT Pro can also import and export audio files in WAV, MP3, and WMA formats. Other features are a new setup wizard, Navigator view, a chromatic tuner, unlimited Undo and Redo, support for control surfaces and MIDI floor processors, and support for 24-bit, 96 kHz audio.

GT Pro 3 works with ASIO and WDM hardware. Minimum system requirements are a Pentium/800 MHz, Windows 2000 or XP, 128 MB of RAM, and 100 MB of free disk space. Cakewalk; tel. (888) CAKEWALK; e-mail sales@cakewalk.com; Web www.cakewalk.com.

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By Scott Wilkinson

Wood and Dried Squid

An old material leads to a new speaker-cone design.

Forty years ago, Toshikatsu Kuwahata had a dream. As an audio-equipment engineer for JVC (www.jvc.com), he believed that speakers should do more than just reproduce the sound of musical instruments—they should *be* musical instruments. He noticed that many of the most beautiful-sounding instruments, such as the violin, guitar, and piano, are made of wood, which inspired him to develop speaker cones made from the same material.

Kuwahata's was no specious dream. He was convinced that the materials traditionally used to fabricate dynamic speaker cones left something to be desired. Specifically, he was concerned about the relationship between the speed at which sound propagates through the material and the vibration-damping factor. For example, polypropylene exhibits low propagation speed and high damping factor, whereas aluminum is exactly the opposite. Paper exhibits relatively low propagation speed and a moderate damping factor.

Kuwahata believed that a superior speaker-cone material should have high propagation speed and a modest damping factor, and he found that wood fit the bill perfectly. In addition, sound waves propagate at different speeds through wood, depending on the angle between the direction of travel and the orientation of the wood grain, which tends to minimize resonance within the material.

He was able to pare the cores of birch logs into long, thin sheets, but he didn't know how to form a sheet of wood into a speaker cone. His first prototype was constructed of fan-shaped wedges glued together. Kuwahata heard improvements in sound quality, but he knew there was no way to mass-produce such a driver. He also tried stamping thin sheets of wood into cone shapes, but they inevitably cracked under the strain.

Kuwahata was about to give up on his dream when the answer suddenly came from an unexpected direction. One of the engineers on his team, Satoshi Imamura, was ordering dinner one evening when he noticed that dried squid was one of the dishes offered on the menu. He knew that dried squid is very chewy, and he wondered how it was prepared in order to be palatable. He learned that it was soaked in sake, which transforms it into a delicacy.

Imamura realized that this could be the answer to the wood-cone problem: if the wood were soaked in alcohol, it might soften enough to withstand the pressure of being stamped into a cone shape. Imamura told Kuwahata about his idea, and they experimented with all sorts of liquor. They found that cheap sake worked best, perhaps due to its impurities. (This provides an uncanny parallel to the world of musical instruments—it is said that Stradivarius soaked his wood in a secret liquid, which gave his instruments their uniquely wonderful sound.)

Another problem that Kuwahata had to overcome was that the stamped cone did not retain its shape over time, especially under conditions of high temperature or humidity. After four years of experimentation, his solution was to apply a thermosetting resin after stamping, which succeeded in maintaining the form even in 90 percent humidity at temperatures up to 60°C/140°F.

The first product to incorporate wood-cone speakers is the JVC EX-A1 microshelf system ($550). The cones are 8 cm in diameter, and the wood is only 0.28 mm thick. Each speaker has a long-throw voice coil, which moves a lot of air and enhances bass reproduction, even from small speakers.

Future models include a two-way design with a 10 cm cone woofer and a 19 mm dome tweeter, as well as a floor-standing design with four 8 cm cones and a 19 mm dome tweeter. Kuwahata is also working on 13- and 16 cm cones. It will be interesting to see if this technology lives up to its promise. Pass the sake! ❀

COURTESY JVC

After soaking in cheap sake, a thin sheet of wood can be stamped into a cone shape without cracking.

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Keeping It Simple

LAURA WILLIAMS

How to streamline your work flow and get your creative act together.

By Nick Peck

How often have you come up with a great musical idea only to have it stymied by technology? Your computer crashes, your synthesizer isn't receiving on the correct MIDI channel, there's ground loop running rampant through your mixer, you can't find the right adapter or cable, your hard disk needs defragmenting, or your digital audio simply isn't syncing from one device to another. You start chasing down one of these technical glitches, and before you know it your creative energy is drained, and the idea has disappeared.

The problem is that your creative focus gets blurred when you shift your attention to the technical aspects of making music. It is commonly believed that technical and creative tasks are processed by different parts of the brain, so switching focus between creative and technical tasks can be difficult. See the sidebar "Ten Tips to Streamline Your Work Flow" for ways to avoid distractions in the studio.

Scheduling separate tech time in your studio allows you to minimize the impact that technical problem solving has on your creativity. Use your tech sessions to focus on getting the kinks out—test cabling, configure software preferences, label all devices and files, create signal-flow diagrams, and even create how-to documents that list all the steps to complete complex or unintuitive tasks. Put up a white board with a "problems to solve" area. If something breaks, buzzes, or crashes during a music session, write the problem on the board and move on with making music. Come back to the problem during your next tech session.

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Keeping It Simple

Software updates come with a special set of technical problems. Updates usually improve the stability and expand the feature set of your software. However, it's important to remember that updating can often lead to compatibility problems whose solutions require more updating. That can be a costly and time-consuming process.

Operating system updates can cause a particularly pernicious chain reaction. Once you update to the latest OS, your digital audio workstation (DAW) no longer works. So you update your DAW only to discover that your plug-ins also need to be updated. Additionally, all your nonmusic software may need updating. And finally, the shareware applications that save you hours every day have not been ported to the new OS, and they never will be because their authors have often moved on to other things. So do your homework before updating your software, and schedule the time necessary to do the job.

FIG. 2: Bouncing related tracks to a single stereo mix can save clutter and CPU cycles. Shown above is a 12-track drum mix being bounced to stereo in Logic Pro 6.

FREE FLIGHT

Needless to say, distractions and responsibilities are a part of daily life. You need to take care of life's details as well as relax and have fun. But try to eliminate distractions from your music work space. Turn off e-mail and Internet browsing—or better yet, relegate those functions to a different computer. Turn off the phone. Set boundaries with people around you, or make your work space off limits. You need clarity of mind and freedom from distraction to work at your creative best, so communicate to others that your music time is precious, and should not be frivolously disturbed.

FIG. 1: Keeping all files related to a project in a single folder will help save time. Shown here, audio, MIDI, artwork, and text files are all kept in the same project folder along with the project documents for Reason, Live, and Logic.

Everyone has their own way of tapping their creative energy. Sitting alone playing your instrument without having any specific musical goal can produce a flood of ideas. Similarly, auditioning files from your loop- and sample library without any specific use in mind can lead to unexpected associations. I like to begin my compositional process away from any instruments or forms of technology. That helps me free my imagination from the playing habits I've developed as well as the synth patches, audio clips, and software features that influence my daily work. Whatever your method is, set aside time for it.

Simply getting a clear sense of a musical problem is often all that your subconscious needs to surprise you with an answer. The subconscious is your most powerful, flexible, and unfettered creative tool. When allowed to wander freely, it will come up with new ideas and novel associations. But it can't be forced. Once you have identified a problem, put it aside, and your subconscious will often come up with a solution for you.

TOOLS AND RULES

It's a good idea to carefully plan a session or composition before you start. Sift through your sample libraries to put together the drum kit for the song, go over the session goals with the other players, and make sure that your instruments are set up and tuned. In other words, make sure to get the technical details out of the way ahead of time. Set up all your equipment, test everything, get sounds and levels set, organize your DAW or recorder track layout, and then take a break. Relax and unwind so that you can walk back into the studio refreshed and get down to creating. Establishing a plan before you begin eliminates wasted time.

As part of your plan, set up a separate folder on your hard drive for all data related to a project (see **Fig. 1**). As you add sound files, create custom plug-in presets, and archive early versions of the project, make sure that

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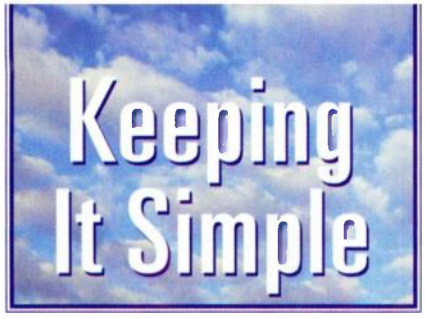

they all wind up in the project directory. Include any text, database, spreadsheet, and graphics files that are related to the project. Many software DAWs give you the option of saving a project as "self-contained" after the fact. This option, however, often overlooks material used in earlier versions of the project that you may want to go back to. Creating your own self-contained file takes less time and can save you the effort of having to search for missing files.

There are thousands of terrific products out there, but you can't possibly learn to be proficient at all of them. If you try to use every new instrument, effect, and sample library that hits the market, you will get only superficial results. Flexibility and options are great, but too many choices can blur your focus and sap your creative energy. It's much better to limit yourself to essential tools that you fully understand than to have a studio full of stuff that you only vaguely know how to use.

It's also a good idea to limit your track count at the outset. It's terrific to have virtually unlimited track counts, but limiting yourself to 24, 32, or even 48 tracks forces you to think in an organized way about your arrangement. It also forces you to abandon takes that don't work as you go along. Remember the adage When in Doubt, Throw It Out. If you just can't bear to part with it, back up the project under a different file name, then continue on with the pared-down version. The 200-track pop mixdowns that you hear about most likely could have been created with 48 or even 24 tracks and a little more organization.

One way to keep the track count and CPU usage down is to *bounce* submixes whenever possible (see **Fig. 2**). For example, if you've tracked each drum in the kit separately to allow for separate processing, mixing, and automation, and you're now done with the drum tracks, bounce them to a stereo or surround mix, releasing whatever effects were used and freeing up tracks. The same applies with background vocals or any other musical element that you've recorded on multiple tracks. In those cases, it pays to archive the individually tracked version so that you can do a remix if necessary.

Open-ended effects processing is another area in which it's easy to lose sight of the big picture (see **Fig. 3**). It's tempting to keep all your plug-ins active until CPU limitations force you to print them. But CPU considerations aside, any decision you make now is one you don't have to deal with later. Furthermore, there's always some risk that you'll lose the custom settings you've labored over before you print the effect. That's especially likely after you've updated the plug-ins, the host software, or the OS. In the case of hardware effects, you typically have only one instance to work with. Printing the effect allows you to reuse it on another track.

BACK TO SCHOOL

The flip side of limiting your resources is learning to fully use the resources that you have. You will continue to collect new hardware, software, and sound libraries as long as you make music. With each acquisition, however, comes the job of learning how to use it. You can't become a virtuoso or power user of everything—but without at least some significant effort, any acquisition is just a waste of money.

Auditioning the presets and reading enough of the manual to learn how to access the critical controls are the most basic requirements for learning how to use a synth or an effects box (hardware or software). You may not want to take the time to understand the finer points of programming a complex new synth, but you should know enough to tweak the presets. For synths and effects that you expect will become a part of your long-term arsenal, spending the time to really learn how to program them is well worth the effort. There's nothing like creating your own sound or effect from scratch, or knowing exactly which control to reach for to instantly modify the sound to match your imagination.

For a new sound library, learning

TEN TIPS TO STREAMLINE YOUR WORK FLOW

1. Schedule regular tech sessions in your studio, and whenever it's possible, avoid having to interrupt your music sessions to fix technical problems.

2. Schedule ample time for the job before updating any piece of software. Be sure to do your homework by researching what the update will entail.

3. Eliminate distractions such as telephones, e-mail, and other personal interruptions from the studio.

4. Plan your session in advance so that the session can be devoted solely to music.

5. Keep your projects self-contained. Maintain a separate directory or folder for all files related to a given project.

6. Limit yourself to essential tools that you fully understand.

7. Bounce whenever possible to keep your track count and active effects to a minimum.

8. Create short pieces devoted to exploring a new sound library. For new gear, audition the presets and learn the basic controls.

9. Don't use unnecessary equipment and software features simply because they are available.

10. Consider selling gear that you haven't used within the past year. In the case of software or sound libraries, consider archiving them on CD or DVD.

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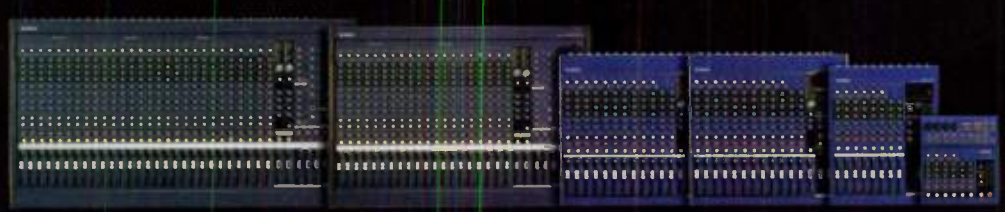

the basics means auditioning all loops and clips and knowing how they are organized and meant to be used. The next step is using them in various contexts to see how they work in your style and with your other resources. One effective approach is to create some short pieces exploring the library. Thirty to 60 seconds in length and with four or fewer tracks is a good format to use because it forces you to stay focused. If you want to get deep into the guts of a sound library, try re-creating one of the demo songs. You may not ever get it exactly right (and it's not uncommon for some key elements to be missing from the library), but you'll definitely know the library inside and out when you're finished.

BECAUSE IT'S THERE

You don't have to use a feature just because it's there. For example, theory tells us that the fewer conversions between analog and digital signals, the better the signal quality. On paper that makes sense, but my experience with digital I/O provides a good example of how those advantages can come at a price.

A number of years ago, I designed my new studio around a digital mixer. I connected two Pro Tools systems, a DAT machine, a DA-88 8-track digital recorder, and several digital reverb units to the mixer's digital inputs. The primary clock source was a Digidesign USD (Universal Slave Driver), with a MOTU Digital Timepiece thrown in for good measure. Designing and implementing

SELLING YOUR GEAR ON EBAY

Online auctions are a great place to pull some extra cash out of the gear you don't use anymore. While there are a number of online auction sites to choose from, eBay is the leader in terms of potential bidders and the number of items listed. Here are a few tips to help you get the most from your auction. Although the focus is on eBay, much of the information can be used with any online auction.

Do your research. Before selling anything, do a search for the item at the auction site. What are people listing the item for initially? What does the bidding trend look like? Also, check out what the item has been selling for in the past few weeks by clicking on the box marked Completed Items Only, which is located on the Advanced Search page. This will give you a realistic price range to expect, depending on the item's rarity, desirability, condition, and so forth.

Include alternate spellings. An auction's search engine is your friend, so make sure your item shows up there. First, spell the name of your item correctly in the Title field. If the name of your item can be misspelled easily, include potential misspellings in the Title field as well as in the description. For example, if you're selling a theremin, you may want to include *theramin*, *theramen*, and *teremin* somewhere in your text. If the product name has a hyphen, include a version without the hyphen, as well as versions with and without a space (for example, GR-20, GR20, GR 20).

Graphics sell. A good-quality photo of your item is essential if you want to get the most out of your auction (see **Fig. A**). Potential buyers want to see what they are bidding on, and because eBay lets you place one graphic for free, do it. If your item has many parts or is rare and desirable, additional photos will help give bidders a better chance to see what they will be getting. And the extra eye candy makes good bait for bidders who are on the fence. Of course, you will be charged for the additional photos, so you will have to decide whether it's worth the extra fees.

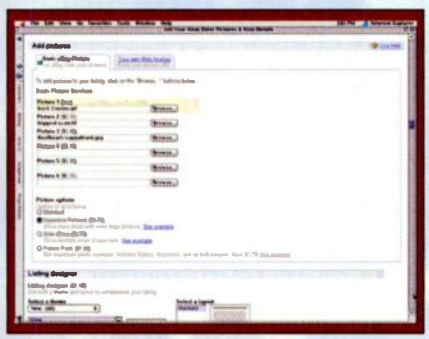

FIG. A: A good photo will help you get the most for your auction item. The first photo is free, and it is often worth the investment to upload additional shots.

To ship or not to ship. Get an estimate of the shipping costs of the item (including packaging and insurance for what you think it is worth) before you list it. Avoid the urge to guesstimate. Even if you don't intend to list the shipping cost, it is good to know the amount ahead of time, in case a potential buyer inquires before placing a bid. If your item is big, heavy, or fragile, and you don't want to deal with the chore of packing it for a trip across the country, specify that it's for local pick up only. If you live in a large metropolitan area, chances are good that your pool of potential buyers is large enough that you'll still get a decent price for your item. If you're selling something rare and valuable, buyers may consider driving from other parts of the country to collect their prize.

Know thy bidder. Or rather, know as much as you can about their behavior in an auction situation. Check out your bidder's feedback ratings by clicking on the number next to their name. Many sellers are wary of bidders who lack any feedback, but since we've all been in that position, it's not a good idea to dismiss them completely. However, if you're selling a big ticket item and your main bidder is a newbie, consider sending the person a note to see if he or she has any questions for you. The reply may tell you what it will be like to do business with the bidder.

—Gino Robair

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Keeping It Simple

the digital signal flow took a week. Finally, I was able to digitally pass a signal from any unit to any other, and each device could function as the clock source when it was transmitting data. Although I was able to get that to work, certain signal paths were used quite infrequently. When I did need to use them, it would invariably take a half hour of head scratching and manual consulting to properly configure all my devices. Even simply switching sampling rates from 44.1 to 48 kHz would take multiple keystrokes on several devices.

Managing the signal flow was time-consuming, and when 96 and 192 kHz sampling rates came along, the digital I/O of the mixer, signal processors, and recording devices were all rendered obsolete. The solution I chose was to sell the digital mixer, both Pro Tools systems, and the USD, and use the proceeds to purchase a newer, more powerful Pro Tools rig. Nearly all connections to the new system are analog. My digital reverbs that max out at 48 kHz function perfectly within a 96 or 192 kHz session because the analog signal path functions as a translator between different devices. In practice, the A/D and D/A converters have become so good now that I don't notice the minuscule amount of additional distortion resulting from several conversions. For the most part, my digital-I/O and synchronization headaches are a thing of the past.

THE ONE-YEAR RULE

A terrific way to keep your clothes closet in order is to follow the one-year rule: if you haven't worn a particular item of clothing in a year, it is time to send it to Goodwill. Someone else will make good use of it, and you will have space to get something that is more in line with your current tastes.

▼

You'll get more done and have more fun if you have just the tools you need.

The same principle applies to gear. If you haven't used something in a year, there is probably a good reason—either it has been supplanted by something else that performs the same function, or your musical interests have gone in a direction that no longer favors that type of equipment. Either way, you can think of that piece of gear as a potential source of money. Sell it and use the money on something that meets your current needs, such as other gear, production costs for your next CD, or relief from the need to earn that amount of money by doing something that takes away from your music time. See the sidebar "Selling Your Gear on eBay" for helpful tips on how to get rid of surplus gear.

But money isn't the only reason to weed out your studio. Carrying the mental and physical baggage of hardware, software, and sound libraries that you will never use again takes your attention from the tools that you do use. Regarding hardware, the lesson hits home if you have to relocate or redesign your studio. That's a great time to get rid of stuff, but if you're not relocating, just imagine what you'd get rid of if you were.

With hard-drive space no longer at a premium and CD and DVD burners on every desktop, you can probably afford to keep every piece of software and every sound library you've ever had. In the case of sound libraries, the problem is finding what you need; in the case of software, you must make sure you avoid software conflicts. Getting the irrelevant stuff off your hard drives (remember it's only a CD or DVD away) will streamline your work flow and reduce your headaches.

All the terrific gear available now at reasonable prices gives everyone the opportunity to produce their own music. That's a great thing. I strongly believe in having the right tools for the job, but the easiest way to simplify your creative life is to remember that consuming is not creating. Time spent earning money to pay for gear, talking about equipment, browsing the classifieds, buying the gear, and installing and learning how to use it is time that you are unable to spend making music. You'll get more done and have more fun if you have just the tools you need.

FIG. 3: Effects can put a strain on the CPU. Shown above, 12 insert effects plug-ins have pegged Cubase's System Performance meter. Printing the effects by bouncing the tracks alleviates the problem.

Nick Peck *(www.underthebigtree.com) is a composer-keyboardist-sound designer who lives and works in the San Francisco Bay Area. You can reach him at nick@perceptivesound.com.*

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Capturing the Kit

A SIMPLE AND PRACTICAL APPROACH TO DRUM RECORDING.

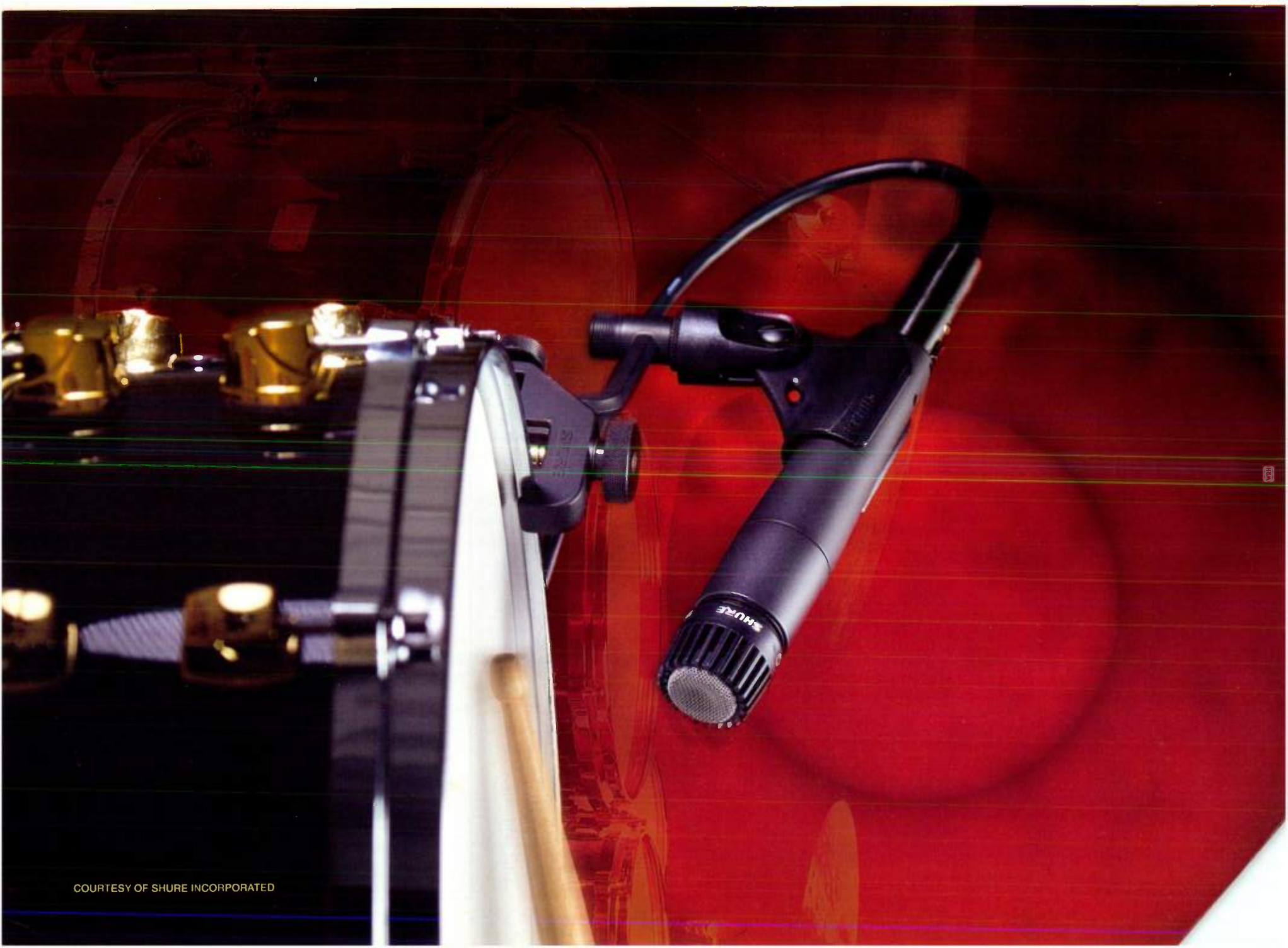
BY BRIAN KNAVE

The drum set is generally considered the most difficult instrument to record well. One reason for this is that a drum kit is a hodge-podge of many instruments: typically four or more drums, a half-dozen or so cymbals, and any number of bells, blocks, and other percussion instruments. The resulting "instrument" produces a huge range of sounds. In terms of frequency alone, a drum kit can cover the entire audible spectrum, from the rib-rattling lows of a big kick drum to the shimmering, harmonic-rich highs of the cymbals. And don't forget the snare and toms, which nicely fill out the mid-range.

The drum set is also capable of producing extremes in dynamic range: on the one end, the whisper of brushes; on the other, the potentially deafening pound of a bass drum. Add to that the challenge of integrating the sounds of so many disparate pieces. Although the drum set is considered a single instrument (based on how it is played), in terms of recording, it is considered both as one instrument and as many.

Is it any wonder, then, that the drum kit's complex blend of sounds has given rise to so many different recording techniques? For those people trying to educate themselves about drum recording, the problem, ironically, is a glut of information: countless books, articles, and interviews, each with a different take, a different favorite microphone (typically one that is too expensive), and, of course, contradictory advice.

If that information overload has you in a pickle, you've come to the right place. The following "holistic" approach to recording drums simplifies the process, helping you get the best sound with the least amount of hassle (and gear). I assume you are a personal-studio operator working without an assistant rather than a professional recording engineer. You probably have only a handful of mics at your disposal, and you might be forced by space limitations to track in the same room where the recording gear is set up. No matter. This approach will help you get the most from the tools you have and capture a drum sound you can be proud of.



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Capturing the Kit

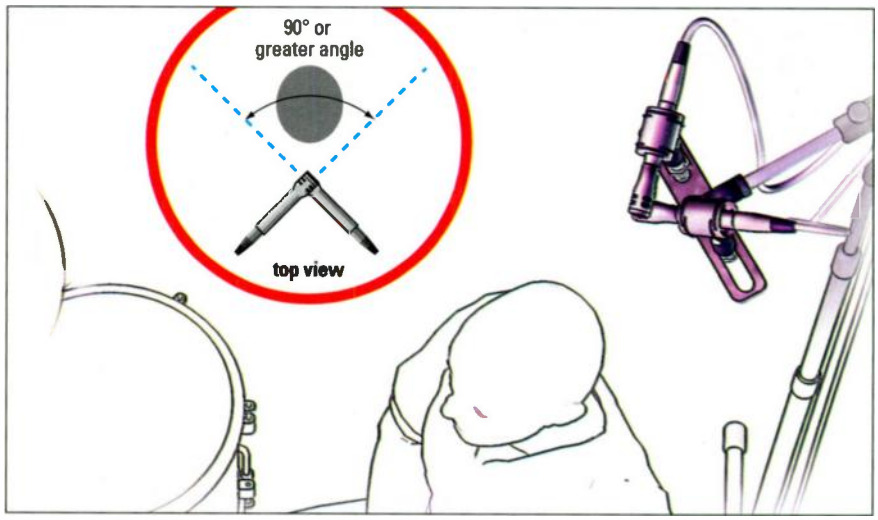

FIG. 2: Usually the quickest and easiest stereo-mic setup, an XY pair positioned over a drum kit will capture a stereo image complete with movement and a realistic sense of the space.

FREE YOURSELF

One way to make anything easier is to reduce the fear of failure. To get over any trepidation you might have about recording a drum kit, remember that there is no right or wrong when it comes to recording drum-set sounds. The only "right" drum sound is the one that works best for a given song. Sometimes that results from a stellar kit surrounded by a dozen or more microphones and processed with a ton of gear. Other times it results simply from a brush smacking a phone book miked with an inexpensive dynamic mic. Either way, it's the song that dictates the drum sounds, and not the other way around—unless, of course, the song began with, or was inspired by, a drum beat.

Just as there are no right or wrong drum sounds, there are no rules for recording drums—at least ones that can't be, and haven't been, successfully broken. Let that fact free your mind so that your creative juices flow. The world is always open to a new drum sound.

That said, there are some general recording rules you should follow. Being general, they hold true for recording any instrument. The two most important ones are to avoid phase problems and maximize the signal-to-noise ratio for each channel. (I'll expand on these two points later in the article.)

SIMPLIFY, SIMPLIFY

Although drum recording can be as complex and exacting as the inner workings of a Rolex, some of the finest drum sounds ever recorded were captured using a minimum of gear. Consider, for example, any number of tracks from Sun Records, Motown, the Beatles, and Led Zeppelin. The point is that unless you are a seasoned drum recordist, it is often best to opt for taking the simplest path.

BREAK IT DOWN

A way to simplify drum recording is to break the process down into parts. That helps organize things, and keeps your eye on the big picture.

You can break down any kit recording into four key components, each of which is fundamental to the quality of the finished recording. These components are the drummer, the drums, the recording room, and the recording gear. Having a serious problem with any one of those four elements can doom your efforts, no matter how good the other parts are. In other words, you can't shrug on any of them.

If, however, you're recording a great drummer playing a well-tuned set of quality drums in a room with good sound, most of your worries are over. All you have to do is capture the resulting sound.

For most of us, though, the confluence of a great drummer, great drums, a great room, and great mics is an uncommon occurrence at best. Be prepared, therefore, to bolster the quality of each component.

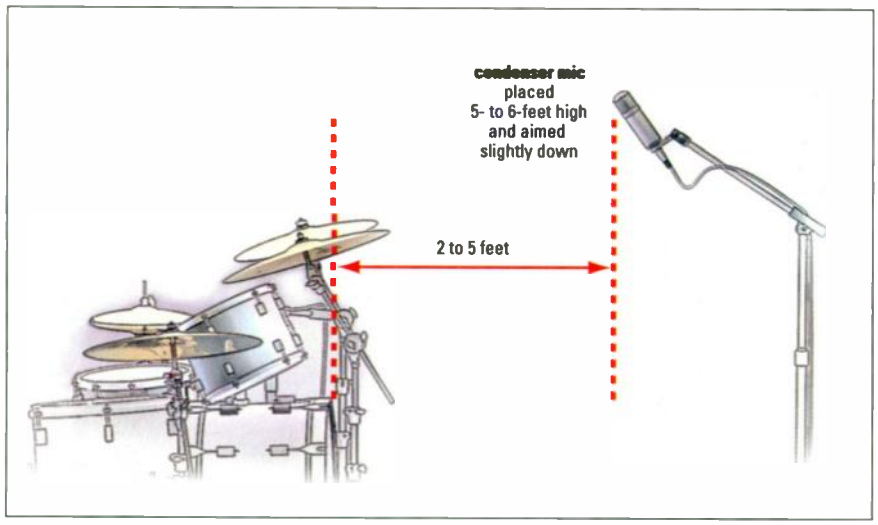

FIG. 1: You can capture a surprisingly good-sounding drum track using only a single, high-quality condenser mic positioned in front of a drum kit.

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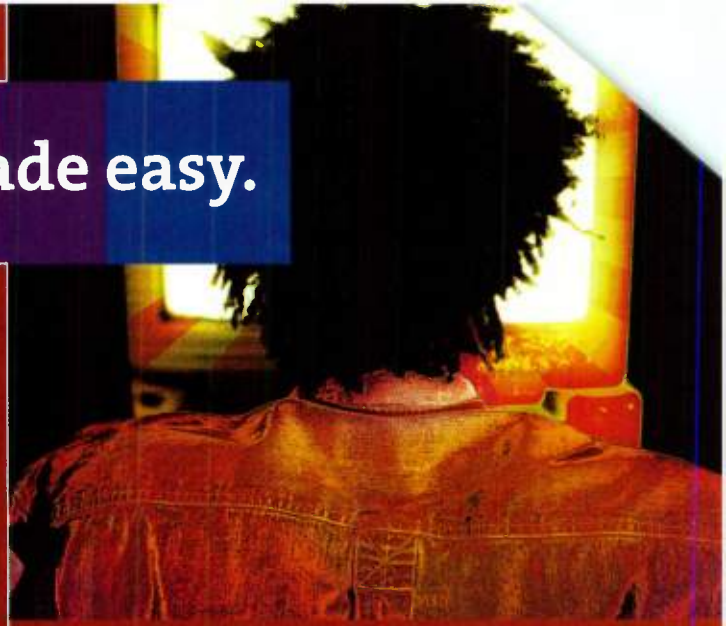

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US-224	2-in / 2-out	2 XLR mic[1], 2 ¼" TRS (bal./unbal.)	YEP	2 RCA (unbal.) + headphone	N/A	two S/PDIF	24 bit	16 ch.	GigaStudio & Cubasis VST	4 faders[2] + stereo master, Phones, Line, Jog and Transport	ASIO, WDM, GSIF, Core Audio, MME, Sound Manager
US-122	2-in / 2-out	2 XLR mic[1], ¼" TRS (bal./unbal.)	YEAH	2 RCA (unbal.) + headphone	two inserts	N/A	24 bit	16 ch.	GigaStudio & Cubasis VST	2 rotary input level controls, Line, Phones, Direct Monitor	

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THE DRUMMER

Generally speaking, the better the drummer, the better tuned the drums will be, and the more musical the cymbals will sound. Nevertheless, it's astonishing how great a crappy old drum set can sound in the hands of a seasoned pro. Indeed, a gifted and inspired drummer can make all the difference in how your recording turns out—no matter what you record it with. Conversely, all the killer gear in the world can't salvage a bad performance.

You should therefore do whatever it takes to get the best from a player. In addition to being personable and helpful, you should see to the drummer's basic creature comforts such as providing fresh water to drink, pleasant lighting, and comfortable room temperature. A relaxed musician is closer to his or her muse than a distracted one.

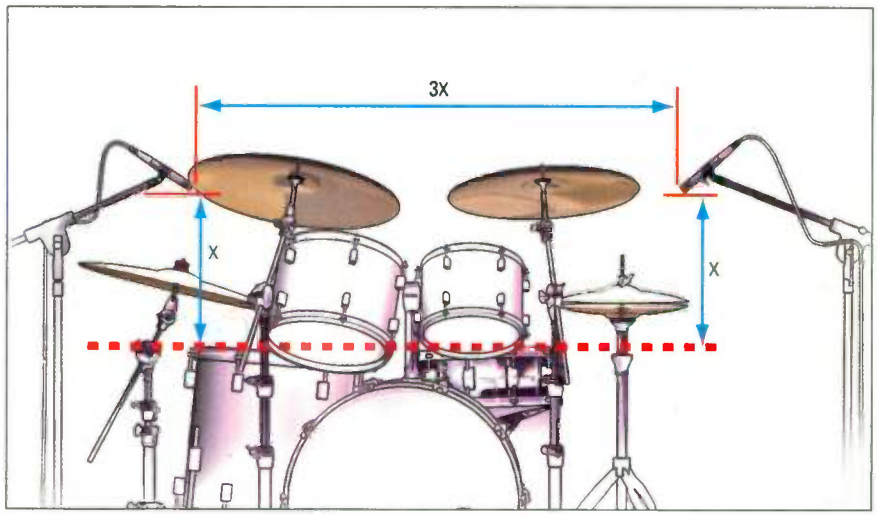

FIG. 3: The 3-to-1 rule, meant to minimize phase problems, is helpful any time you need to put two or more microphones in close proximity.

THE DRUMS

If the drums don't sound good from the start, then there's work to be done—sometimes a lot. Can you hear squeaks and rattles as the drummer

DRUM TUNING 101

A drum is essentially a cylinder with a vibrating head stretched tightly over each end. Hoops, typically made of metal but sometimes made of wood or plastic, hold the heads against the bearing edges of the drum shell. The hoops are held in place by a number of tunable lugs (typically 5 to 12, depending on the size and type of drum) positioned equidistantly around the drum shell on either end.

Your first objective when tuning a double-headed drum is to get each head in tune with itself. Start with either head. While the drum is suspended (whether on a mount or by hand), tap the head with a finger, stick, or mallet at each lug point and tune so that all points are the same pitch. Now do the same for the other head.

Next, get the drum's two heads in tune with each other. Generally, the best interval to start with, at least for toms, is with both heads tuned to the same note, or in "unison." The note does not have to be a particular pitch; it should, however, be in a comfortable range for the drum, neither too high (tight and choked sounding) nor too low (loose and flappy). When you find the just-right combination, the drum will sing its note freely.

From unison, there are two directions you can take the tuning: make one head tighter than the other, or make it looser. (Tip: tune from the bottom head so as not to affect the desired playing response of the top head.) Experiment with these tuning variations to learn how they affect the sound differently—there's a big range of sounds at your disposal.

On a snare drum, the bottom head is typically thinner and, for many styles (such as jazz), tuned higher than the top (batter) head. Such a tuning maximizes a "crisp" response from the snares. Conversely, tuning the bottom head looser than the top makes for a lower, heavier sound—more rock and roll.

Note, too, how the tension of the snares affects the drum sound. As with the heads, if the snares are too tight, the sound gets choked. If the snares are too loose, they're likely to buzz excessively.

Like toms, double-headed kick drums usually sound fine—if not best—with the heads tuned in unison. Often, though, drummers tune the batter head lower than the front one so as to get a lower note from the drum. Though a low kick-drum note is generally desirable, make sure the drummer hasn't made the batter head too loose—the head should not be flappy to the point of showing wrinkles, or else the drum's resonance can "dry up" beyond recognition (at which point you might as well be recording a cardboard box).

An overly-resonant double-headed drum can be dampened by leaning a pillow or blanket against the front head. A thick blanket thrown over the whole drum—a helpful technique for isolating the kick-drum mic—can also dry up the tone a bit. If the kick has only one head, or a sizable hole in the front head, you can adjust the balance of attack (dry thud) and decay (resonance) by altering the position of the dampening material inside the drum (blanket, pillow, or whatever). Push the material more up against the batter head to increase attack, and pull it more away to increase resonance or "tone."

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plays? Locate and squelch them. If the drumheads are full of dents and sound dead, have them replaced. If the cymbals sound unmusical or just wrong for the song, get better ones. (Yes, you should be listening for such things.) Have a drum key ready and offer to help if the drummer doesn't know how to tune the drums well.

In some cases it might be best to postpone the recording session to allow enough time to get the drums sounding their best. The sound of the kit, after all, is a key component of the final recorded sound. As the recordist, it's your responsibility to listen carefully and ensure that the sound coming from the kit is as good as it can reasonably be, and certainly not marred by readily "fixable" things.

Given the fundamental role of drums in contemporary music styles, it makes sense that engineers, especially self-starters, learn how drums work and how to tune them. The basics are not hard to grasp. (See the sidebar "Drum Tuning 101" for a quick lesson.) If you want to learn to record the drum set well, it helps immeasurably to become familiar with its many parts and how they function.

Once all the drums are well tuned and any squeaks and rattles are tamed, you might also need to dampen one or more of them. A well-tuned drum, especially if it's of a good quality, will resonate freely, and will produce a much longer decay than a poorly tuned drum. Those free-ringing toms probably will not present a problem if you are miking only from a distance (with stereo overheads, for example). Close-miking, however, can present certain problems: for example, even when the toms aren't being played, they resonate sympathetically with the other drums, which gets picked up by the close mics, resulting in a murky rumble that can spoil the kit sound. (In certain jazz settings, however, that rumble might be regarded as part of the sound—and the drummer would have your head for tampering with it.)

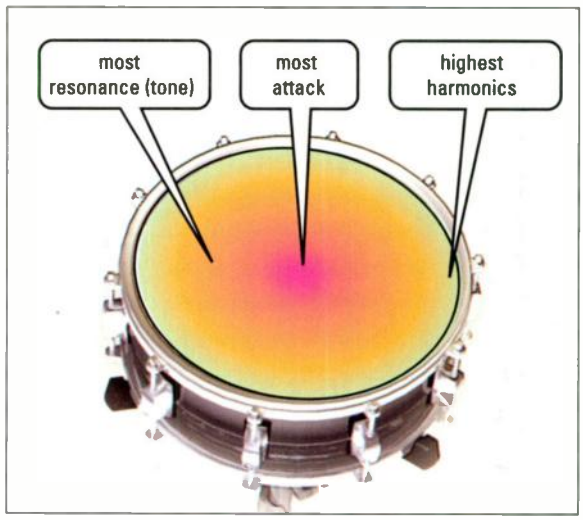

FIG. 5: The area on any drum head nearest the rim will ring freely and have a preponderance of high harmonics. The area between the rim and center will produce the most resonance, and the area at the center of the drum head will sound driest and produce the most attack.

Most drummers dampen their kick and snare drums, at least a little, as a matter of course. But be prepared to dampen further, if necessary. Common items for dampening kick drums include blankets and pillows (down pillows usually sound best), strips of felt (running beneath one or both heads), and various patented dampening systems. "O-rings"—donut-shaped pieces of Mylar, cut the same diameter as the drum head—are good for dampening snare drums. O-rings can quickly be laid on top of the drum or removed, as needed. They also allow the drummer access to the full playing surface of the snare drum.

To dampen toms, tape a folded cotton handkerchief, a small rectangle of foam rubber, or some similar material (tissue, cotton gauze) onto the top of the drum head, close to the rim and away from the drummer. In most cases, a small amount of material taped an inch or so from the rim is sufficient; to increase dampening, move the material toward the center of the drum, use more material or do both. Tip: use sturdy masking tape or some other type that's easy to remove—duct tape will muck up heads and hardware. (See the sidebar "Engineer's [Secret] Drum Toolkit" for some useful items to keep on hand.)

FIG. 4: The sound that the snare mic picks up will be determined not only by the particular drum and mic, but also by where the mic is aimed, its angle, and how close it is to the drum. Start with the mic cap two inches above the head, angled 30 or 40 degrees, and aimed just shy of center.

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Capturing the Kit

THE ROOM

Along with making the kit sound its best, you also have to consider the recording space and how it affects the drum sound. Sound can't exist in a vacuum; it is always part and parcel with the surfaces reflecting it (an effect made eerily clear in anechoic chambers, in which you can hardly hear yourself speak). The drum set is usually played loudly, making room reflections more apparent. The room can't help but be a component of the drum sound; it's impossible for its "sonic imprint" *not* to appear on the tracks. Without fail, you will be working with—or against—that imprint at every stage of the mix.

In short, the kit has to sound good in the room. Fortunately, this part of the puzzle can be enhanced with practically no knowledge of acoustics. You just need ears and a willingness to experiment.

The goal is simple: find the place where the drums sound "best" in the room. Of course, if they happen to sound great just where they're sitting, then you can advance directly to Go. The idea, after all, is to *improve* the sound. Just as the slightest movement of a microphone can result in a big change in what the mic hears, relatively small changes in the orientation of a drum kit inside a room can make significant differences in the overall sound.

Sure, moving the drums from one spot to the next can be a hassle. But if you care about getting the best sound, it's worth it. Finding the "sweet spot" for the drum kit is something that can hardly be done without experimentation. True, the more you do experiment, the easier it gets, and in time you will develop a sixth sense about it. (Experienced drummers often instinctively set up in the best-sounding spot.) But ultimately, rooms are quirky and unpredictable, and you'll find that experimentation will yield surprising results.

The main thing to listen for is a favorable balance between the kick, toms, and snare drum. Turn the snares off so you can better hear the drums ringing. Is anything noticeably out of whack in terms of volume, resonance, or decay? If so, try a different location. Does one of the drums cause the snares to buzz excessively? Repositioning the drum set—or just that one drum—might solve the problem. Is the kick drum lacking in oomph? Try setting up the kit so that the drummer sits in a corner looking out into the room (assuming a rectangular room, that is). Are the toms sounding thin? Try pulling the kit more toward the center of the space. Is the room just not working for you? Then find another. I have rented church halls (surprisingly affordable), warehouses, and art galleries to get the drum sound I was after.

The other side of the coin is acoustical treatment for the room. You can tame, for example, nasty flutter echoes with judicious placement of acoustical foam rubber. Don't have the budget for that? Try hanging blankets, thick

ENGINEER'S (SECRET) DRUM TOOLKIT

Here are some items that can prove indispensable to getting you through the obstacle course of a drum-recording session. The trick is to pull them out only when necessary; you don't want to give the impression you're infringing on the drummer's territory. Fortunately, most pro players will have all contingencies covered; that's one of the reasons they carry a trap case. But you can never be too prepared.

Essentials:

Drum key(s)
Cymbal felts and sleeves
Can of lightweight oil (to squelch squeaks)
Roll of fine, strong twine (for securing snares and other duties)
Assorted pieces of cloth, felt, and foam rubber (for drum dampening)
Sturdy masking tape (for attaching drum-dampening materials)
Duct tape (for just about anything else)
Scissors and single-edged razor blades
Adjustable wrenches, screwdrivers, and heavy pliers
Assorted blankets, pillows, and towels
Drum rug (preferably with a rubber bottom)

Handy Additions:

Snare-drum batter head (14-inch coated is standard)
Snare-drum bottom head (14-inch clear thin is standard)
A pair or two of drum sticks (5A and 5B are common sizes)

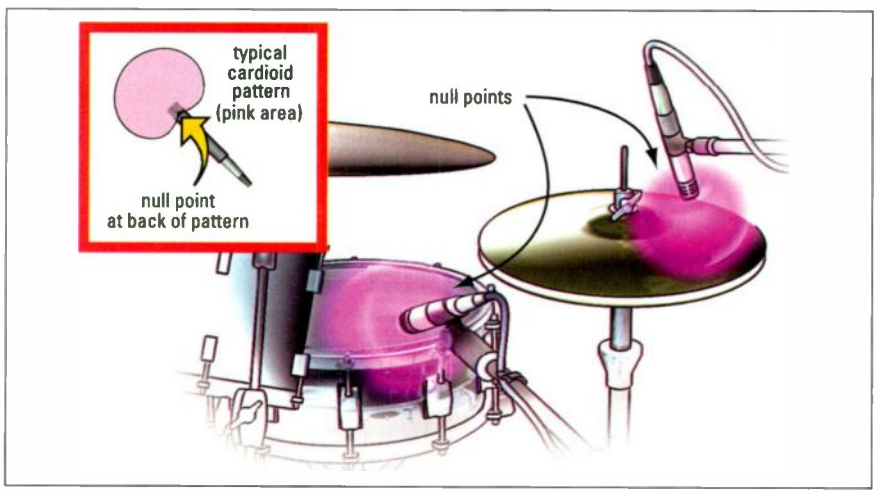

FIG. 6: All directional mics naturally form a null point, or an area at which no sound is picked up, behind the front of the diaphragm. This can be used to your advantage when you need to isolate one mic's pickup from another's—a most useful technique on close-miked drums.

curtains, or rugs from the walls or the ceiling (or both), or positioning large pieces of furniture, full bookshelves, or what have you around the room so as to deaden it. If the room is too dead sounding—a small, thickly carpeted and curtained study, for example—try such things as arranging large wood panels along one or more of the walls, setting up the drums on a wooden riser, or both—anything to provide some helpful reflective surfaces.

SHELLY DOTY

A fully miked drum kit provides the greatest amount of control over the sound at mixdown. Here the author plays a session at Guerrilla Recording in Oakland, California. Not visible is the room mic: a large-diaphragm tube condenser positioned close to the floor, "looking" at the kit from a distance of 12 feet or so.

THE GEAR

Many assume that gear is the most important component in capturing a good drum sound; more specifically, the mics and mic preamps. Good mics and preamps are a tremendous help, but as long as the source sound is less than desirable, that's all your good mics and preamps will capture—something less than desirable.

We're trying to make drum recording easier, so let's start by paring the signal chain down to the basics. All you really need for each channel is a microphone, or mic preamp, and a track to record on. Leave your compressors, EQs, and other doo-dads for the mix. That saves time, and it forces you to find the best mic (and mic position) for the job, as well as to set the gain properly on the preamp.

Setting the gain right is no big mystery, but you do need to pay attention. While the drummer is playing at record volume, simply dial in the hottest levels you can get for each track—for the whole song—before peak distortion. That's called maximizing the signal-to-noise ratio.

Note the differences between digital and analog level setting. Analog tape has a higher noise floor (hiss) than digital

12 TIPS FOR BETTER DRUM RECORDING

1. **Keep it simple.** Learn to get a great sound using a minimum of gear, and then build on your successes from there.
2. **See the big picture.** The final recorded sound is determined not only by the gear and how you use it, but also by the drummer, the drums, and the recording space. Do what you can to bring out the best in each.
3. **Learn the basics of drum tuning,** and acquaint yourself with the drum kit's many parts and how they work. To be fully prepared, keep a drum toolkit on hand (see the sidebar "Engineer's [Secret] Drum Toolkit").
4. **Seek out great-sounding rooms to record in**—if you don't have a decent drum room, that can make a huge difference in the final sound. Think large rooms, high ceilings, wood floors (churches, art galleries, warehouses).
5. **Find the drums' sweet spot** in the room. A drum kit will project different tonal balances depending on where it's positioned in a given space. If you're after a great sound, it's worth the effort to suss out the best-sounding location for the drums.
6. **Make the drummer comfortable.** Much depends on his or her performance.
7. **Select microphones by type.** Typically, dynamics are used for close-miking kick, snare, and toms (with the largest diaphragm reserved for the kick), and condensers are used for overheads, hi-hats, and assorted percussion. But don't be afraid to buck convention—use what sounds good and works best for the song.
8. **Use proper stereo recording techniques.** They can not only add a delicious spatial realism to your drum recordings, but a stereo pair can also cover the whole kit sound when you don't have enough close mics to go around.
9. **Minimize phase distortion between mics.** Use the 3-to-1 rule, but also do test recordings and listen in mono to ensure phase coherence, especially between drum overheads and other mics.
10. **Maximize signal-to-noise ratio for each track.** With digital, the loudest hits should use up most of the bits; with analog tape, hit it till it hurts, then back off a touch.
11. **Angle the mics** rather than positioning them so their diaphragms are parallel with drum heads. That can lead to problematic phase interactions caused by reflections between the parallel surfaces.
12. **Use your ear,** not your eye, to do final mic-position tweaks.

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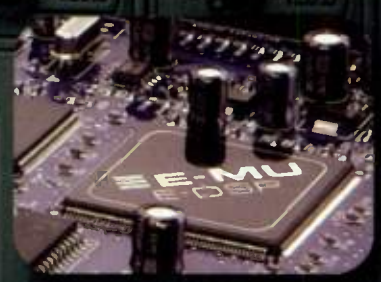

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recorders, so it's especially important to maximize the signal-to-noise ratio when recording to tape. After all, the hotter the signal, the less tape noise you hear.

In fact, drums are typically recorded "hot" (above 0 VU, that is) to tape to create a particular sound. The hot signals saturate the tape, which flattens out transients, resulting in a compressed sound that many engineers and musicians favor, especially on kick, toms, and snare. (Overheads and hats are usually recorded "colder," to preserve the accuracy of transients.) You should familiarize yourself with the tape machine, because decks differ in how they handle distortion (based on tape size, tape speed, types of heads, and so on). But generally speaking, any decent multitrack tape recorder will let you run drum levels +3 or more into the red with no problem; indeed, that's where the sound starts getting good.

With digital you never want to go "into the red." In other words, make

MILKING THE KICK MIC

Where you position the kick mic can make a big difference in the sound that you capture. However, kick drums and kick-drum mics vary so much that it's hard to generalize. You need to get familiar with a mic to accurately predict what it will do in a given situation. Moreover, a different model won't always behave in a similar fashion—it might behave quite differently.

Here are a few generalizations. As with any drum, the attack is greatest at the center of the batter head. To increase attack, aim the mic diaphragm more toward where the beater strikes the head. You can also increase attack by moving the mic closer to the batter head, at least up to a point. To increase resonance, turn the mic away from where the beater strikes and more toward the resonant part of the head, or position the mic farther back from the drum, or both.

Kick drums come in three head setups: single headed, double headed (no holes), and double headed with a hole in the front head (perhaps the most common setup). Single-headed bass drums excel at producing a very dry, "thuddy" kick sound. The amount of thud can be fine-tuned by moving the packing material inside the drum. Miking single-headed kick drums is fairly straightforward: start with the mic somewhere between "slightly inside" and "all the way inside" the drum, and tweak from there.

FIG. A: To minimize ill effects of air venting forcefully from the "sound hole," angle the microphone on both the vertical and horizontal planes. The drum pictured here is very resonant, even with all the dampening. Thus, the mic is angled also to aim at the beater-strike area, which increases attack.

Double-headed kick drums with a hole in the front head present more miking options—a good thing, because they're typically harder to get a great sound from. A good starting point is with the mic diaphragm flush with, or slightly inside of, the hole, tilted so that it looks at the beater-strike area from an angle (see **Fig. A**). If the mic picks up too much resonance from this position—not uncommon—try taking the mic off the stand and laying it inside the drum (on top of a blanket or whatever) more or less in the middle, with the diaphragm facing the batter head at a slight angle.

If the sound is still boomy, try putting the mic on the other side of the kick drum, next to the pedal. Position the diaphragm so that it looks at the beater-strike area yet minimizes sound coming from the pedal (see **Fig. B**). This position will greatly reduce resonance and provide a strong, solid attack (although isolation will suffer, naturally). Of course, another alternative is to remove the front head from the drum. However, that might not sit so well with the drummer.

Double-headed kick drums with no hole in the front head leave few options for miking. All you can do is put the mic a few inches in front of the drum, facing the head, and move it around until you find the best position. As with the double-headed-with-hole kick, if the drum sounds too boomy (or lacking in attack) no matter where you position the microphone in front, try miking from the batter side.

FIG. B: For a quick alternative to removing the front head on a kick that's just too boomy, mic the drum from behind with the diaphragm aimed at the beater-strike area.

Although it's generally best, when miking any drum, to avoid aligning the mic diaphragm parallel with the drum head, on kick drums, you might want to experiment with breaking that rule. With the microphone aimed directly into the batter head at close range (one to three inches), the reflections bouncing between the parallel surfaces, not to mention the sheer force of air, can make for a radically slamming sound. But again, you never know until you try—different mics respond differently in that kind of situation.

sure that the drummer's hardest hits register close to 0 on the meters (so you use up as many bits as possible), but not above it. Unlike analog distortion, the sound of digital distortion is never pleasant. Fortunately, even if you do get a few digital peaks, you might be all right. Most digital recording devices allow a few decibels of headroom above the red before distortion kicks in. Moreover, distortion from a few peaking drum hits might not be audible anyway, due to the fast transient nature of loud hits. So if peaking is there but you can't hear it—both before and after processing the track—then don't worry about it.

CHOOSE YOUR WEAPONS

Next comes determining how many mics you will need to work with and where you should put them on the kit. First, sort the mics by type; use dynamic microphones (moving-coil mics, not ribbons) for close-miking duties and condenser mics on everything else.

There are several reasons that dynamic mics are generally preferable for close-miking. They can better handle the extreme SPLs generated by hard-hit drums, and they're less likely to be damaged by stick hits. In addition, dynamics typically provide better rear rejection, and they often feature a presence boost around 5 kHz, which helps accentuate the attack of the stick striking the head. Because they are directional mics positioned close to the drums, they can enhance the low end as well (thanks to the proximity effect).

BRIAN KNAVE

FIG. 7: When miking both the snare drum and hi-hat, you can minimize leakage by using directional mics and angling them slightly away from one another.

Condenser mics are typically more sensitive and accurate than dynamics. Thanks to their extended frequency response (especially on the high end) and more accurate handling of transients, they capture the true sound of instruments better, whether at close

Capturing the Kit

range or from a distance. That makes them ideal for hi-hats, overheads, and room mics, as well as for miking percussion arrays.

Alas, condensers are also more costly and fragile than moving-coil mics. You should think twice about putting even modern condensers, some of which can handle surprisingly high SPLs, within reach of flailing drum sticks. With drummers whose playing you trust, however, you can use condenser mics on kick, snare, and toms and get great sounds. The only potential problem is leakage—no matter how tight the polar pattern, condensers usually capture more sound than you want coming from the rest of the kit (and the room).

As for microphone sizes, the only rule you need to follow is to reserve your largest-diaphragm mic (usually a dynamic) for the kick drum. Ideally that will be a mic designed for bass drum or other low-frequency duties. But if all you have is an assortment of handheld dynamics, you should audition them all to see which gives you the lowest, most authoritative kick sound. The same goes for determining which mics are best for the snare and toms; you just never know until you've tried each mic in each position.

It's advisable to reserve a few hours for mic testing before the session begins, especially if you're new to drum recording. Only by systematically testing various positions with each mic (and keeping careful notes, of course) can you determine what's going to give you the best results.

ONE, TWO, THREE?

Do not despair if you have only one microphone, especially if it's a condenser. You can get a great drum sound from a single, high-quality condenser mic placed strategically near the drum kit. In this case you are mixing drum levels on the spot, so listen carefully to the tonal balance as you position the mic. You are also mixing ambience—the further back the mic is from the drum kit, the more room sound it captures.

Start with the mic a few feet in front of the drums, five or six feet above the floor, and tilted down a bit so as to include some kick drum (see **Fig. 1**). That's usually a good single-mic position because it lets you get the most kick in the mix. Other viable positions are behind and above the drummer. Try recording a minute or so of drumming from all three positions—in front, behind, and above the kit—and compare the results. Then make your decision based on what works best for the music. If the song features an important tom fill, for instance, you might prefer the overhead position because it highlights attack from the toms.

If you have two identical or very similar

mics, you also have two good options: either put one mic on the kick drum and the other positioned as just described, or use the pair as stereo overheads. Allow the musical style to determine which approach is better. For example, you might take the former setup for a dance tune, and the latter for a jazz ballad.

THE STEREO EFFECT

Stereo recording is a huge topic unto itself, and one deserving of study by any recordist. But it's especially worth the effort when you are limited to only a few mics, or any time a natural sound is desired. (For more information on stereo recording, see "Double Your Pleasure" in the June 2000 EM, available at www.emusician.com.)

Two simple but effective stereo-miking techniques are *XY coincident* and *AB stereo*. Both techniques require two identically modeled microphones, preferably a matched pair of condensers (although I've captured some great-sounding drum-overhead tracks using mismatched mics; generally speaking, though, use like mics).

Directional polar patterns—cardioid, hypercardioid, and supercardioid—work best for XY coincident techniques, whereas any polar pattern can work for an AB pair (though traditionally, each mic is set to the same pattern).

The XY coincident technique is usually easiest because the two mic capsules are positioned as closely together as possible (coincident). That eliminates time-of-arrival (phase) differences between the left and right signals, thus maintaining the frequency response even when the signals are summed to mono.

The "XY" refers to the orientation of the mics, which often are adjusted with a 90-degree angle between the two caps (see **Fig. 2**). But don't hesitate to experiment with the angle—I often stretch it out to 110 degrees or greater to get a more dramatic stereo spread. Typically, the XY pair is positioned a few feet above the drummer's head, pointed down at the kit. Raise the pair higher to capture more room sound, or lower to get a closer, more focused kit sound.

Also, experiment with the mics positioned a bit behind or in front of the drum set.

The AB stereo setup, also called a spaced pair, is more challenging because of the risk of phase discrepancies between the left and right signals. But this bugaboo can foil *any* drum recording employing two or more mics—not just stereo tracks.

The trick in avoiding phase discrepancies is to heed the 3-to-1 rule, which states that the distance between any two mics should be at least three times the distance between the mics and the sound source (see **Fig. 3**). In rooms that are small, acoustically dead, or both, I prefer omnidirectional mics for spaced overheads; in larger, more reverberant spaces, I typically opt for directional mics.

COUNT THE WAYS

If you have three mics and two of them are identical, use the like pair as overheads and put the third mic on the kick drum. There are several workable positions for the kick mic: in front of the drum, slightly inside it, all the way inside it, and behind the drum. Try them all out, and get familiar with how each affects the sound differently (see the sidebar "Milking the Kick Mic"). Then you're more prepared to handle whatever comes your way—an unusual-sounding drum, a funky-sounding space, or whatever.

If you have four mics including a like pair, use the pair as drum overheads and put the other two mics on the kick and snare drums, which typically are the two most important drum elements in the mix. (The snare-drum mic can be positioned to pick up some of the hi-hats as well—actually, it's hard to do otherwise. Just pull the mic back a tad and give it a slight tilt toward the hats.)

By the way, I use this four-mic array on a regular basis, both for demos and albums. Not only is it fast and economical, but as long as the first three components (drummer, drums, room) are sounding good, this mic setup is often all you need. It gives you individual control of the main drum elements (kick and snare) and provides a true stereo image to fill in the rest of the kit. The resulting tracks, when well mixed, can sound huge—and in some cases indistinguishable from a fully miked kit.

HEADS UP

When positioning mics on drums, especially close mics, a good guideline is to angle the mics and avoid forming parallels between mic diaphragms and

drum heads. This technique helps to prevent adverse phase interactions caused by reflections between the head and the diaphragm.

On the snare drum, start with the mic above the head, angled downward 30 to 40 degrees toward the head, with the diaphragm roughly even with the rim of the drum (see **Fig. 4**). The trick is to find the balance between what sounds best to the mic, what provides the best rejection of unwanted sounds, and what's comfortable for the drummer. This can take some doing. You can't just position mics in a haphazard or formulaic way and expect to automatically capture the best sounds.

Note first how any (unmuffled) drum's response varies across the head (see **Fig. 5**). The sound is "ringy" and high pitched (rich in high harmonics) near the rim, and it is most resonant exactly between the rim and center. At dead center the sound is driest, and thus strongest on attack (this is because the resonance gets dampened by cancellations along the head).

The drummer has primary control of these tonal variations, by striking the head in different spots. But the mic, in addition to hearing what the drummer is doing tonally, hears the drum differently depending on which of the three areas it is focused on.

When close-miking with a dynamic, you can also affect how much "air" you put around the drum. Many engineers automatically position snare and tom mics as close as possible to the drum head, often with the mic angled steeply so that the cap points into the head close to the rim. That may give better isolation, but capturing the best sound from the drum is ultimately more important than reducing leakage.

Typically, putting the mic too close tends to choke the sound. The drum will sound fuller and more natural if you give it a bit of breathing room. Simply pull the mic back an inch or so, and reduce the angle of attack. That will open up the sound and increase attack simultaneously (assuming the mic is aimed more at the center of the head). And you might even find that

Capturing the Kit

the extra leakage works to integrate the sound of the kit.

To sum it up, you have three things to play with when listening for the sweet spot: the angle of the mic, how close it is to the drum, and where it's pointing. Whenever possible, make final mic-position tweaks based on what you're hearing, rather than on what you're seeing. Just close your eyes and let the sound guide where you position the mic. (A tip for those recording in the same room in which the gear is set up: wear closed-ear headphones and have the drummer play softer than usual, at least to get started. That gives you a fighting chance of determining the best place to put the mic.)

Another way—more tedious, but also more telling—is to record identical passages of drumming with the mic in a slightly different position each time. Three or four variations is usually sufficient. Afterward, compare the tracks on the monitors.

HIGH FIVE

If you have a fifth mic at your disposal, common practice would dictate putting it on the hi-hats. And depending on the song you're recording and the emphasis you're going for, the hi-hats might well be the best place for it. But unless the song cries out for separately miked hats, consider using the fifth mic to record room sound, particularly if you're loving the sound of the drums in the space.

A condenser mic with an omni or a figure-8 pattern is usually best for room miking, but try whatever you have. Sometimes a funky old dynamic can capture a hip ambience. No matter what mic you use, though, the mix engineer will probably be thankful; a good room track can add that extra something special to the drum mix.

Assuming that you already have one or more overhead mics on the kit, try positioning the room mic on the other side of the room, far from the drums. Walk slowly around the space while the drummer plays and listen—with ears only at first—for an appealing balance of drum elements (lows, mids, and highs) and ambience (room sound). After finding the general good area, put the mic there and fine-tune the position while listening through closed headphones (or the control-room monitors if you have a control room and an assistant). Move the mic around a lot and listen—it might even sound best aimed into a wall or a corner.

Experiment with putting the room mic up near the ceiling and down close to the floor. Not surprisingly, many pro

engineers prefer a stereo pair of mics for capturing room sound. Again, it all depends on the sound you're going for and the gear that's available.

Of course, depending on the song, that precious fifth mic might be better used elsewhere—for example, on a tom that is played repeatedly as part of the groove. If you use it on the hi-hats, be sure to position the snare and hat mics (assuming they're directional) so as to maximize isolation between them. You do that by taking advantage of each mic's null point, which is the sound-rejecting area directly behind the capsule (see **Fig. 6**). The reason you want maximum isolation is so the two tracks can be processed differently in the mix without affecting each other too much. It's frustrating to bring up the hi-hat track and have it ruin the snare sound.

When miking hi-hats, position the mic so it's looking down toward the bell of the top cymbal, or just to the edge of the bell, at a slight angle. This emphasizes the clear highs coming from the bell area, and keeps the diaphragm clear of air blasts coming from between the cymbals. It also avoids the gonglike quality that cymbals typically produce when miked near the edge.

In addition, keep the hat mic a safe distance from the top cymbal. Placing the mic any closer than three inches to the cymbal will cause you to risk picking up a weird-sounding phase change as the cymbal moves toward and away from the mic. You also risk the cymbal crashing into the mic. Some drummers keep their hats quite loose and with a fair space between the cymbals, so pay attention to the throw of the hat cymbals as the drummer plays them hard. To keep it safe, position the mic capsule at least five inches above the hats, angled slightly away from the snare drum to reduce leakage (see **Fig. 7**).

Finally, when recording hi-hats, be sure to engage the mic preamp's high-pass filter (assuming it has one). Many hat cymbals are quite thick (especially the bottom one) and produce an unpleasant low-frequency roar that is better filtered out from the start.

MIKE CRUZ

Mastering Waves

Beyond the basics with the Waves Platinum Native Bundle.

By Bob Reardon

A few years ago, creative effects processing required racks of outboard equipment. Few people imagined then that a powerful computer with the right software could ever replace all that gear, but today almost every conceivable effect is available as a digital signal processing (DSP) plug-in. In this Master Class article, I'll start by describing some different uses for regular processing, such as dynamic EQ and upward compression. Then I'll cover some of the Waves Platinum Native Bundle's more unusual plug-ins. That should give you a hint of the excitement that's in store for you if you take the time to fully explore the creative possibilities of all 26 plug-ins in the bundle. Although I highly recommend that you reconstruct these examples for yourself, you can download most of the settings described in this article from the EM Web site at www.emusician.com (see **Web Clip 1**).

Suppose you have an acoustic piano track that sounds pretty good but could use a bit of brightening when the pianist is softly comping the verse. You set up your EQ only to find that it sounds harsh when the pianist plays an octave higher and plays harder in the bridge. You could, of course, create two tracks with different EQ settings or automate the EQ on a single track, but those solutions are unnecessarily tedious. Here's how to set up Waves C4 as a dynamic equalizer to more easily address the problem.

C4 allows you to set two EQ curves: one for low level and another for high level. The transition between the two EQ curves is controlled by separate threshold settings for each of the four EQ bands. The frequency ranges of the four EQ bands are determined by low, mid, and high crossover settings. (You can also use the Linear Phase Multiband EQ for this, which provides phase-linear crossover and five frequency bands.)

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Mastering Waves

Fig. 1 shows the bell-shaped curve that I have created for the band between the Mid and High crossover points, which are set to 2.15 kHz and 6.48 kHz, respectively. The Q setting and the width of the band control the shape of the curve, with higher Q values and narrower bands resulting in a sharper curve. The Gain control sets the boost or cut of the band, and I've set it to 7.3 dB. That results in a boost centered around 4 kHz when the level is below the threshold of –41.9 dB. What happens above the threshold is controlled by the Range setting.

The Range setting determines how the EQ is reduced above the threshold, and I've set it to –15 dB in this example. The effect is indicated by the purple area under the curve. If you scroll the setting you'll notice that positive values result in a boost above the threshold, whereas negative values result in a cut. The yellow line displaying the EQ curve animates during playback, giving real-time visual feedback of the effect of the range and the threshold settings.

In this case, I've focused on the dynamic EQ of a mid-high band designed to brighten soft piano comping as well as to soften harder playing. The other bands can, of course, be used either dynamically or fixed (by setting the Range to 0 dB) to further EQ the track. For example, you might set the high band with both negative gain and range to create a high-shelving EQ that increases with dynamics.

UPWARD COMPRESSION

If you've used the L1 or L2 Ultramaximizers for peak limiting to make your mixes louder, you've undoubtedly discovered that using peak limiting to reach for *competitive level* often entails a considerable sacrifice in dynamics. Here's one way, called low-level or upward compression, to raise the volume of the softer sounds in the mix without getting the squashed sound of too much peak limiting. This technique has long been popular in classical music, in which wide dynamic range is common and peak limiting is rarely used.

For upward compression, use the Linear Phase Multiband EQ. Start with the factory default settings, then use the Master controls to set all the bands to these settings: Threshold, –50 dB; Range, –6 dB; Gain, 6 dB (see **Fig. 2**). Those settings will increase the level of soft passages while leaving the louder passages untouched.

The audio just below the Threshold value will be compressed upward a maximum of 6 dB. The high-level signals (the ones significantly above the threshold) will have no gain change, because at high levels the Range and Gain settings, which are set to complementary values, combine to equal unity gain. You can see what's happening by observing the yellow Dynamic Line display. You'll see a significant difference in the curve when the input signal is low compared with when it's high.

Experiment with the threshold and crossover points in the individual bands to fine-tune the compression to suit your program material. Always make sure the Range (negative) and Gain (positive) controls are set to exactly offsetting values, and that Makeup is set to Manual. Fine-tuning the Master Threshold control will allow you to lift the lower dynamic sections of the mix without sitting on the peaks.

As a final touch, use an L2 placed after the Linear Multiband. Set it for no more than 3 dB of gain reduction. That provides competitive level with punch that sounds more musical and interesting.

FIG. 1: The C4 Multiband Parametric Equalizer offers four bands of dynamic equalization. The third band, between the Mid and High crossover settings, applies the boost indicated by the yellow bell-shaped curve as long as the signal level in that band remains below the Threshold setting. As the signal level exceeds the threshold, the EQ curve changes shape as indicated by the shaded region.

COOL SPACE, FRESH BEATS

Multitap delays have many uses. Two of my favorites are creating space around an instrument or vocal and creating new rhythmic grooves from a drum loop. Here's how to do both using SuperTap.

Adding space around a vocal. To create space for a vocal, start with a pair of taps set to very short delay times followed


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by two pairs of taps set to progressively longer delay times (see **Fig. 3**). Optimum delay times depend primarily on the tempo but also on the style of the song. Set the delay times for each pair to close but not identical values. The gain settings are also critical. You can set each pair to the same level by shift-clicking both controls before setting the value.

Rolling off the high frequencies above about 2 kHz for the first pair of taps allows you to pan them wider and make them louder without smearing the high frequencies. On the longer delay taps, a wide bell with maximum boost reduces both the low and the high frequencies to minimize the low-frequency mud and make the highs more natural. (In most acoustic situations, long echoes have the highs rolled off.)

Other interesting adjustments include setting all the delays on one side slightly longer and softer than those on the opposite side. The sense of space can be further enhanced by deriving your reverb send from the delayed signal or from a mix of the delayed and original.

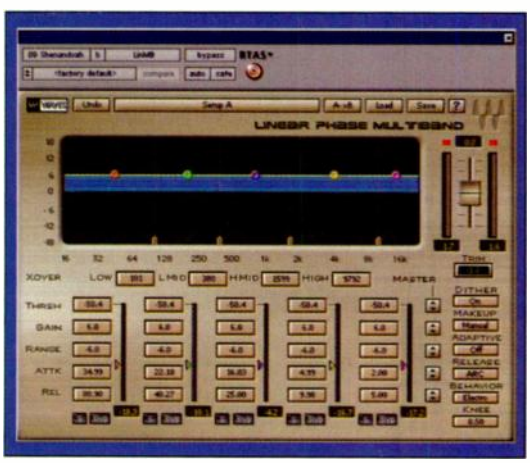

FIG. 2: Linear Phase Multiband has five bands of dynamic equalization with phase-linear crossover between bands. Its linear response makes it an ideal tool for applying dynamic EQ to some bands while leaving others untouched.

New beats. You can also use SuperTap to rearrange a one-bar drum loop using the plug-in's Beat Grid mode. Although designed for drum loops, the effect can be useful on any kind of rhythmic material. For example, U2 has used it with different filter settings to process guitar to great effect.

To sync to the tempo of the loop, choose a quarter-note grid, turn on only the first tap, set its Delay slider to the four-beat mark, and tap or adjust the bpm setting. Now switch the grid to 8th or 16th notes, and experiment with different Delay settings to find an interesting rhythmic beat.

In my example (see Fig. 3), the first tap emulates an 808-style kick drum using lowpass filtering, while the next three taps produce hi-hats by using very steep highpass filtering. The bottom two taps yield snare sounds due to the narrow bell curve EQ. I suggest setting the EQs first, then auditioning the taps one at a time to find a useful rhythm and sound (see **Web Clips 2** and **3**).

TUNED DRUMS

A favorite effect of mine is to start with a conventional drum kit and turn it into a percussion kit, to sound like log drums, for example. The aptly named Enigma is just the tool for that. Try this experiment starting with the SuperTap treated drum loop from the previous example. Use the Enigma settings shown in **Fig. 4.**

The Mix and Depth settings are the most critical. Very low depth settings are needed in order to achieve the tuned-drum sound, and you probably don't want any original signal in the mix. The Feedback Delay setting is also important. You'll get interesting variations on the resonance effect by changing the number of notches, reversing the phase, and varying the feedback delay settings; the settings that are shown in Fig. 4 are the most natural sounding (see **Web Clips 4, 5,** and **6**).

The pitch variation comes from the Modulation settings.

FIG. 3: SuperTap is a 6-tap feedback delay line. Its delays can be synchronized to tempo and set in note-value increments or can be free running and set in milliseconds. The top setup shown here adds space around a vocal or lead instrument. The bottom setup, synched to tempo, fills out a drum loop with new beats.

Of course, the Modulator needs to be turned on, and a square wave should be used to avoid pitch sliding (though that can be interesting, too). The tuning of the pitches is controlled by a combination of the settings of the filter Frequency controls (under the top graph) and the Minimum frequency and Maximum frequency controls in the Modulator section. Very low Rate settings produce the most realistic drumming, but also try higher settings. Interesting and musical edits can be achieved by altering the number of notches and the Feedback control for this Enigma preset. The factory presets labeled "perc" near the bottom of the Load menu are all variations on this theme.

M-S FOR THE REST OF US

M-S is an abbreviation for the mid-side microphone recording technique. A cardioid microphone is used for the

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M (middle) signal and pointed toward the center of the performers, while a figure-eight microphone is used for the S (side) signal. The figure-eight mic is oriented 90 degrees from the cardioid, facing the left/right axes of the performers. Even if you don't do M-S recording (and I encourage you to look into it), you can use M-S processing to separate an already mixed stereo signal into its mid and side components for independent processing. This technique would be useful, for example, with a mix in which the lead vocal is in the center and a bit buried, whereas the guitars and drums are more on the sides and already bright enough.

The Waves S1 plug-ins provide the conversion back and forth between an M-S signal and the normal stereo signals. The conversion to M-S format is done by the S1 MS Matrix plug-in, and the conversion back to stereo is done by the S1 Imager plug-in. Any stereo processing that you sandwich between those plug-ins will affect the mid and side signals separately, with the mid signal processed as the left channel and the side signal as the right. In the

Plug-ins by Category

Category	Effect	Features
Compression/Expansion	C1 Parametric Compander	Compression and expansion with gating, sidechain, and filtering options.
Compression/Expansion	Renaissance Compressor	Compression and expansion with the simple interface of the Renaissance series.
Compression/Expansion	Renaissance Vox	Compression and expansion tailored for vocal material with the simple interface of the Renaissance series.
EQ	Linear Phase Equalizer	Phase-linear EQ.
EQ	Q10 Paragraphic EQ	Multiband parametric EQ with 1-, 2-, 3-, 4-, 6-, 8-, and 10-band versions to save processing.
EQ	Renaissance Equalizer	Multiband parametric EQ with 2-, 4-, and 6-band versions provided to save processing.
EQ (Dynamic)	C4 Multiband Parametric Processor	4-band parametric EQ with individual compression for each band.
EQ (Dynamic)	Linear Phase Multiband	5-band parametric phase-linear EQ with individual compression for each band.
EQ/Compression Combo	AudioTrack	Channel-strip module with 4-band EQ followed by compressor/expander.
EQ/Compression Combo	Renaissance Channel	Channel-strip module featuring 4-band EQ with sidechain and compressor/expander. Either EQ or compression can come first.
Limiting	L1 Ultramaximizer	Peak limiting, level maximization, noise-shaped dithering.
Limiting	L2 Ultramaximizer	Peak limiting, level maximization, noise-shaped dithering.
Munging	Doppler	Simulates the effect of a moving sound source.
Munging	Enigma	Notch-filter, resonator, feedback-delay combo effect.
Munging	MetaFlanger	Flange, phase, chorus effect.
Munging	MondoMod	Amplitude modulation (tremolo), frequency modulation, and rotation effect.
Munging	Ultrapitch	Formant-corrected pitch shifting and harmonizing.
Reverb	Renaissance Reverberator	Stereo reverb with separate control of early reflections and reverb tail.
Reverb	TrueVerb	Stereo reverb with separate control of early reflections and reverb tail.
Utility	DeEsser	De-essing.
Utility	MaxxBass	Bass enhancement by creating upper harmonics from which the ear re-creates the fundamental.
Utility	PAZ Psychoacoustic Analyzer	Visual frequency and stereo field analyzer.
Utility	Renaissance Bass	Bass enhancement by creating upper harmonics from which the ear re-creates the fundamental.
Utility	Renaissance DeEsser	De-essing.
Utility	S1 Series	Conversion between mid-side and stereo formats.
Utility	SuperTap	6-tap stereo delay with filters

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aforementioned example, you might place a stereo EQ, such as the Q10, between the S1 plug-ins to EQ the vocal separately from the other parts.

After setting up the chain of plug-ins, make sure that the S1 Imager's input is set to M-input-S mode, that both polarities are set to +, and that channel swapping is not active. The top of **Fig. 5** shows left-channel settings for Q2 (the two-band version of Q10) that might be used to brighten up a centered vocal. The bottom of Fig. 5 shows a slight low-end boost that adds a little width to the low end of the side signal. Remember to turn Q2's Link mode off, so you can apply separate EQ settings to each channel.

It's also a good idea to turn down the input gain on one channel of the EQ at a time, so you can home in on the way the mid and side signals are being affected. But, you should always do the final check and adjustments in the context of full stereo.

BASS REPLACEMENT THERAPY

MaxxBass is one of the least understood, yet most useful of the Waves plug-ins. It appears to boost the fundamental harmonic in a bass recording, but it uses neither subharmonic synthesis nor EQ and compression to do so. It tricks your brain into thinking that the fundamental is present. You can also use Renaissance Bass for this job.

FIG. 4: Enigma is a feedback resonator effect based on notch filtering. The setup shown here turns a standard drum kit into a percussion kit with sound like a log-drum.

MaxxBass uses the fact that the brain can reconstruct a missing fundamental from the harmonics that the ear hears. For example, if you have a radio or television with small speakers, the

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bass guitar in a rock song or the double bass in a concerto can't be reproduced—the fundamental pitch is below the speakers' range. However, the higher-pitched harmonics are reproduced by the speaker. Your brain recognizes that these harmonics are related to each other and are correlated to the fundamental and therefore senses the presence of the missing fundamental. Interestingly, this phenomenon has been known for centuries—pipe organ builders used exactly this technique to emulate very long pipes.

In practice, you can use MaxxBass to make your mixes compatible across a wider range of playback devices, bumping up the low end on small speakers and still getting a great sound on larger ones. It doesn't matter whether you apply this technique on individual tracks or on the final mix. Here's how to process a music mix intended for small television speakers. I strongly recommend using a pair of small TV speakers to monitor the results.

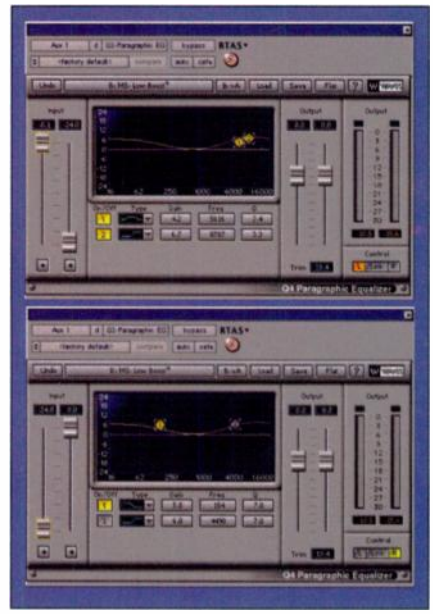


FIG. 5: The Q10 Paragraphic Equalizer comes in configurations ranging from one to ten bands. You can unlink its stereo channels to apply different EQ to each side of the stereo image.

Start by inserting MaxxBass on the final output channel to see that it can work its magic in the context of the whole mix. Choose the bottom high-pass filter curve from the HighPass menu in the Harmonics section. That provides a steep 24 dB-per-octave rolloff of the low end. (You may prefer a gentler low-frequency rolloff for compatibility with larger speakers.) The Decay setting controls the rolloff of the higher harmonics that are added to the signal to create the MaxxBass effect.

Begin with a setting of –15.0 dB per octave. Set the Freq control to 100 Hz and set the Original Bass fader to 0 dB. Those settings replace the original bass with a MaxxBass-created low end, and they roll off the low frequencies below 100 Hz. Finally, set the dynamics ratio to 2:1 and the response to 20. That adds some compression to the harmonics.

SEEING IS BELIEVING

Although it does no processing of its own, the PAZ Psychoacoustic Analyzer is a very handy mixing tool. In the frequency domain, PAZ offers a 52-band spectral analysis, with the bands chosen to closely approximate the constant-Q critical frequency bands of the ear. That results in a good correlation between what you hear and what you see in the PAZ frequency analyzer. The Peak/RMS and Response controls let you choose the type of analysis and response time. In Peak mode, peaks in each frequency band are displayed. The Response parameter controls the release time. In RMS mode, the energy is averaged over a length of time, and the Response value controls that length.

When making adjustments to the overall frequency balance of your mix, it can be quite helpful to switch between Peak mode (with a fast response time) and RMS mode (with a slow response time). In RMS mode, use a Response setting between 400 and 600 ms. That closely matches the perception of the overall frequency balance that you hear, providing a good visual reference.

In addition to allowing you to smooth out irregularities in the frequency spectrum of your mix, PAZ is great for tracking down and fixing masking. Masking occurs when a louder sound in a particular frequency range prevents softer sounds in that frequency range from being heard. Masking problems abound in dense productions, in which many parts compete for attention. Panning is a good place to begin addressing the problem, but EQ is also useful.

PAZ can be helpful for fine-tuning your pan and EQ settings to bring out masked parts by accentuating different frequencies or by moving them to a different part of the stereo field. Start by soloing an instrument and using the A-B button to store its frequency plot. Then audition competing instruments, and compare their frequency plots. Target the overlapping regions for EQ, or separate the instruments by panning.

In this Master Class feature, I've been able to cover just a few of the 26 plug-ins in the Waves Platinum Native Bundle. For a full list of the plug-ins and a brief description of each, see the table "Plug-ins by Category." The examples in this article have been chosen to encourage you to move beyond the boundaries of basic EQ and compression. You can then look forward to many pleasurable hours of tweaking and munging as you work your way through the rest of the bundle. (For more information on Waves software, check out the company's Web site at www.waves.com.)

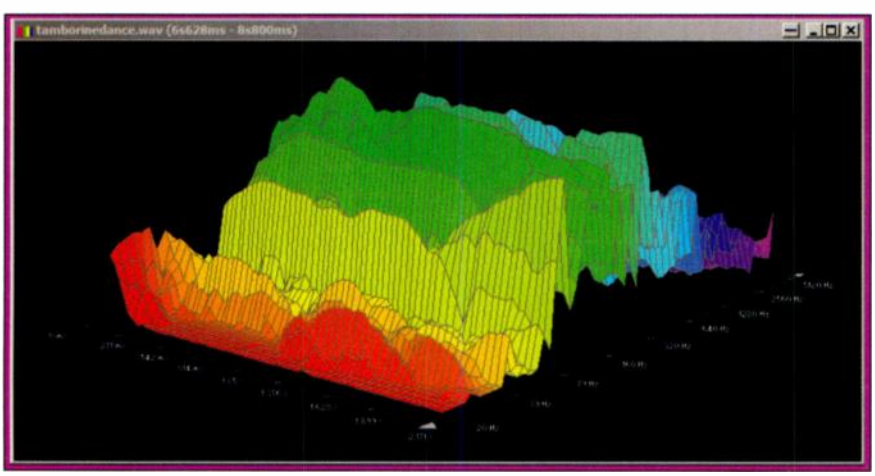
Bob Reardon *resides in the Boston area, works for Waves, and has a background in music recording, audio post-production, and synthesis.*

Look Through Any Window

Seeing sound from the inside out.

By Mark Ballora

Conventional notation has allowed performers, composers, and theorists to discuss music for centuries. More recently, people have used images of electronic and digital signals to help facilitate those discussions. Signal views are useful when attempting to examine a sound's spectrum or internal structure. Although the ability to view the inside of a sound may have been an absurd notion a century ago, technology has evolved to a point where viewing the makeup of a musical signal is a regular occurrence in recording studios and electronic-music labs. In this column, I will present an overview of some of the "microscopes" that are commonly used for looking at the inside of music today.



CATCHING THE WAVE

In his seminal work *On the Sensations of Tone* in 1855, Hermann Helmholtz, a scientific renaissance man, studied and represented complex waveforms. His only tools were resonators that he had invented, trigonometry, and a pencil and paper. Representational techniques improved by the end of the nineteenth century with the advent of the Electronics Age. Spark-gap oscillators produced electrical current that flowed in accordance with simple harmonic motion—quick movement that flows from one direction to the other and then back again. This alternating current produced a corresponding magnetic field, which induced a corresponding electrical signal in a nearby conductor. The transmitted signal could be amplified and sent to a loudspeaker, and thus the radio broadcast was born. Along with the birth of the radio broadcast, however, came the problem of creating and maintaining better broadcasting equipment.

In 1897 Karl Ferdinand Braun invented the cathode-ray oscilloscope, which enabled electrical signals to be viewed in real time (see **Fig. 1**). The

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FIG. 1: An oscilloscope displays a time-domain view of a waveform.

oscilloscope display was created with a horizontal-sweep oscillator, which produced a sawtooth signal. This acted as a guide for the scanner to sweep linearly from left to right, thousands of times per second, plotting a glowing dot as it went. The sawtooth oscillator later became the basis for television displays, and the oscilloscope and other testing devices became the key components of an electrician's toolkit. Square- and triangle-wave oscillators were created for testing purposes. Feed a square wave into an electrical system, then view the system's output with an oscilloscope. If it does not look square, then there is a problem with the wave. Decades later, those same oscillators became the raw sonic material in analog synthesizers.

Using a Web search engine, you can find a variety of oscilloscope demonstration applets. One example of an applet is The Virtual Oscilloscope, (www.virtualoscilloscope.com), created by Peter Debik; another example, created by Professor Fu-Kwun Hwang, can be found at National Taiwan Normal University (www.phy.ntnu.edu.tw/java/oscilloscope/oscilloscope.html).

IT'S JUST A MATTER OF TIME

Oscilloscopes create time-domain images, meaning that information is displayed as a function of time. Air-pressure changes of acoustic events are converted to changes in current. The current is input to the oscilloscope, which shows pressure/current changes along the vertical axis and time along the horizontal axis. Although time-domain views of waveshapes may be interesting, they don't tell us much when it comes to examining complex waveforms.

While investigating the propagation of heat in 1815, French mathematician Jean-Baptiste Fourier introduced a theorem that became a keystone in wave analysis. The Fourier theorem states that all complex periodic waves are the sum of a set of sinusoidal waves. The frequencies of these waves are all harmonically related, meaning that they are all integer multiples of the lowest frequency (called the fundamental). Each frequency has its own relative amplitude and phase with respect to others. One notable feature in Helmholtz's book is his premise that since musical (pitched) tones are periodic, the Fourier theorem was an effective starting point for studying them.

Perhaps you have seen the rainbow pattern of colors that appears when light shines on a prism. That rainbow pattern appears because white light is composed of several different frequencies (which we perceive as color). Passing the light through the prism causes each frequency to refract by a different amount (due to the differing wavelengths), and thus they separate and can be viewed individually. A Fourier analysis (or transform) is a mathematical procedure analogous to a prism for any complex periodic waveform. The spectrum of a waveform is often a much more useful view than the time-domain waveshape produced by an oscilloscope.

The combination of simple waves, all at their respective amplitudes and phases, forms the spectrum of a complex wave. A spectral domain (or frequency domain) view of a wave shows amplitude as a function of frequency, meaning that frequencies are represented along the horizontal axis and their respective amplitudes are shown on the vertical axis (see **Fig. 2**). Because phase is not a significant audible factor in unchanging waveforms, it is generally not represented on spectral plots.

THE FAST AND THE FOURIER

The Fourier transform, however, is a theoretical procedure and tends to have certain drawbacks when applied to real-life situations. For one thing, it is a mathematically intensive process and is not something that can be simply programmed into a cathode-ray system such as an oscilloscope. With digital technology, a sampled waveform can be analyzed with a discrete Fourier transform (DFT). The DFT is also computationally expensive and cannot produce spectral displays in real time. A more efficient algorithm, called the fast Fourier transform (FFT), was created in 1965. With some restrictions, (for example, the number of samples in the

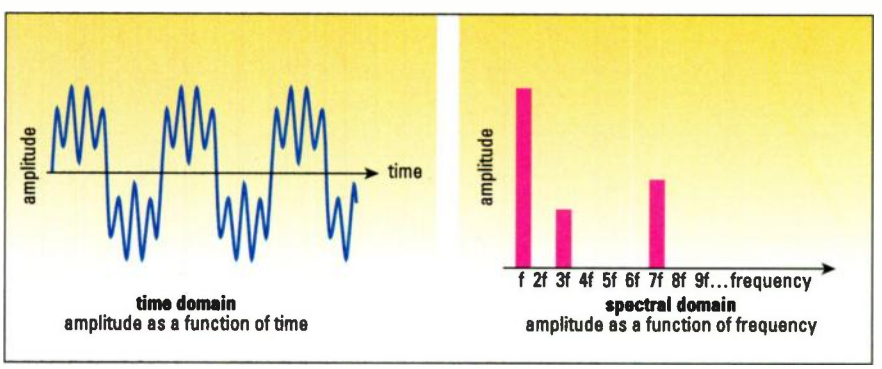
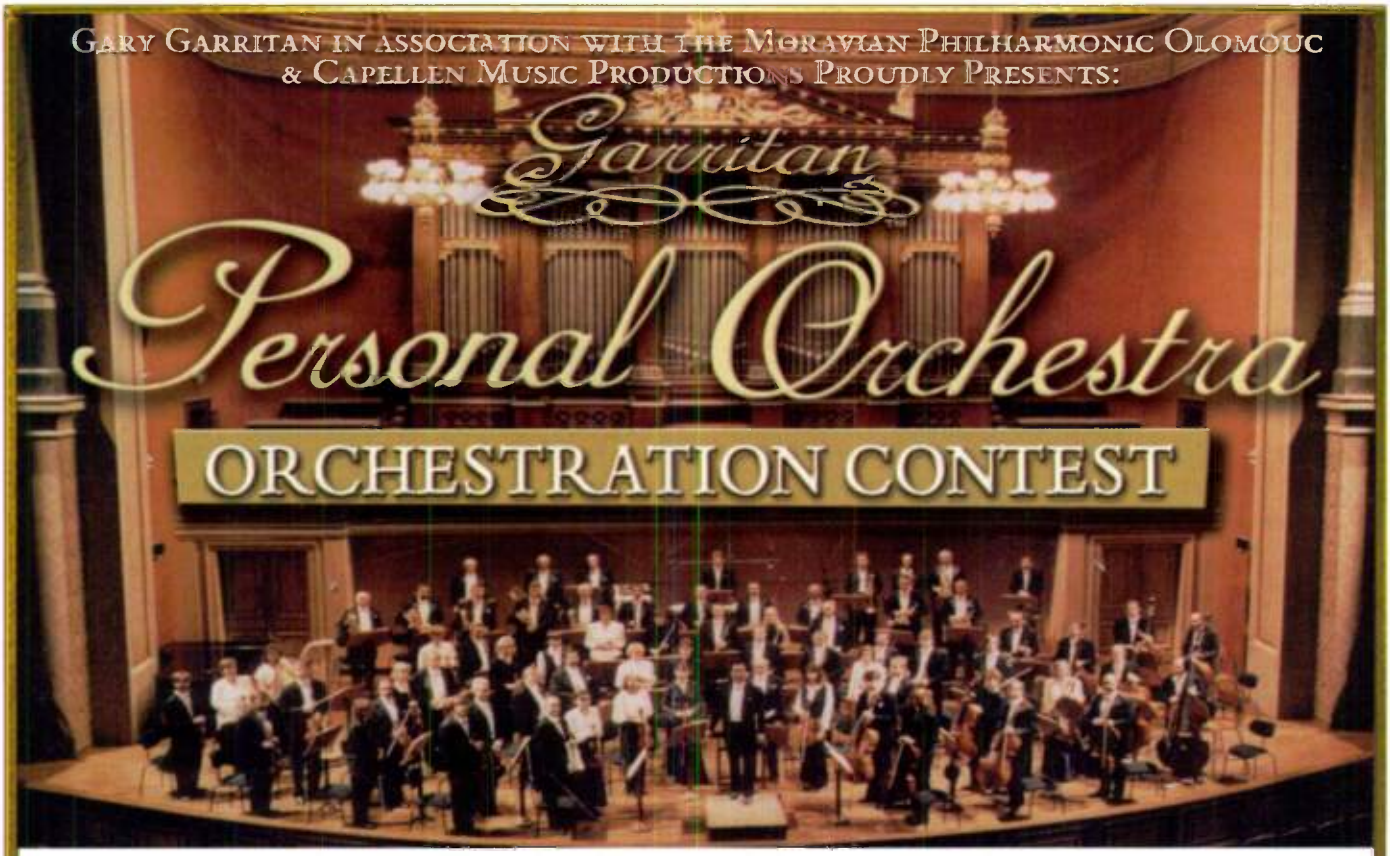


FIG. 2: The frequency content of a time-domain view is not always easy to determine. A spectral view shows the fundamental frequency to be at full amplitude, the third harmonic at one-third this amplitude, and the seventh harmonic at one-half this amplitude.



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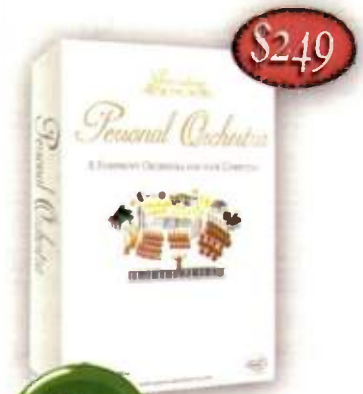

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signal must be a power of two), the FFT offers tremendous improvements in speed and allows spectra to be displayed in real time.

Fourier transforms are used in many scientific fields. If you have waves of any kind (electromagnetic, heat, fluid, and so on), sooner or later you'll need a spectrum. When used for music, however, the Fourier transform turns out to have a few drawbacks. For one thing, the transform works for waves that are perfectly periodic, but music is, at best, quasi-periodic. Most sounds produce noisy, irregular transients at their onset followed by periodicity (pitch).

In an attempt to capture the changes in frequency over time, the material is analyzed piecemeal, in short segments or windows. A generous estimate for transient times is 12 ms (368 samples of CD-quality audio). To capture attacks effectively, window sizes generally vary from 128 to 1,024 samples. The transform turns each sample into a filter: a window of *n* samples becomes *n* fixed-bandwidth filters, with center frequencies at multiples of the sampling rate divided by the window size. This ratio is also equal to the bandwidth of each filter. But there is some overlap among the bands, and frequencies that fall between the filter center frequencies confuse the transform. These "in-between" frequencies get stripped out and are moved over to the output of neighboring filters. This is called spectral "leakage" or "clutter." The shorter the window, the fewer filters there are, and the more leakage they produce. Longer windows have more filters and less leakage.

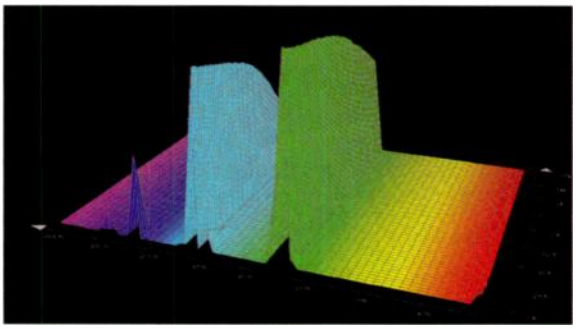


FIG. 4: This time-varying spectrum shows that the strongest frequency components are the fundamental and 4th harmonic (1.176 kHz). The transient consists of short appearances of what appear to be the 3rd harmonic (882 Hz), the 10th harmonic (2.940 kHz), and the 15th harmonic (4.368 kHz).

Another problem is that transients get "smeared" over the length of the window, which makes them lose their precision. This is called localization blur. Localization blur is reduced with shorter window lengths. Shorter windows, however, produce a less accurate view of the frequencies in the spectrum. Window size is a tradeoff: longer windows result in increased spectral resolution; they also, however, result in diminished time resolution. Shorter windows produce increased time resolution; they also, however, result in diminished spectral resolution.

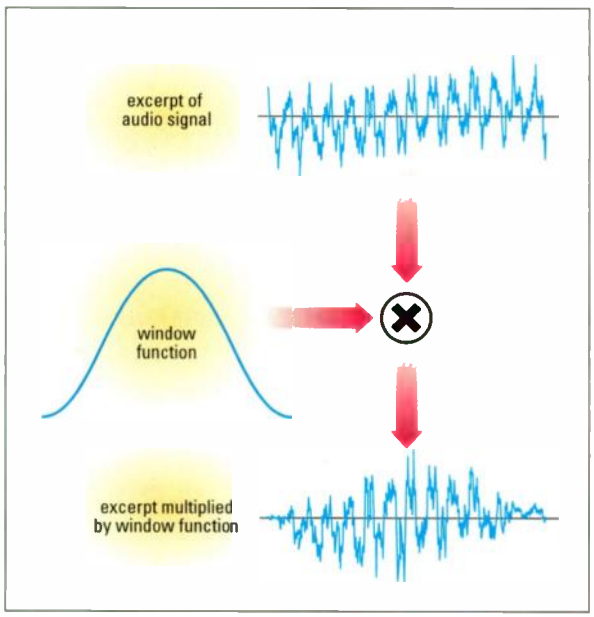


FIG. 3: Each sample in a windowed excerpt is multiplied by the corresponding entry in a window function. The result more closely resembles a periodic wave and produces a more accurate spectral analysis.

THROUGH THE LOOKING GLASS

The inaccuracies of spectral leakage and localization blur are minimized by "arithmetic preheating." A set of samples is not simply lifted from the signal and then analyzed. Normally they are all multiplied by a symmetrical window function before the analysis (see **Fig. 3**). When every sample in the window is multiplied by a corresponding value in a window function, the excerpt appears more periodic to the transform, resulting in an analysis that is a better approximation of the signal's actual frequency contents.

Most programs offer a choice of window functions, with names such as Hamming, Blackman, or rectangular. A rectangular window represents no window function being applied at all, though the differences among the others is often subtle. Choose anything other than rectangular, and you should get a good spectral rendering.

WHITER SHADES OF GREY

When you make changes to EQ settings, you are fine-tuning the spectrum of your signal, adding or subtracting from different spectral regions. Some meters give you the option of a dynamic EQ display. This is analogous to motion-picture frames, which are a series of short-time Fourier transforms (STFTs) that are viewed in rapid succession. This can be an invaluable visual reference when you're mixing. Does your material have too much high end? This is easy to see on a dynamic EQ. Some multiband EQ plug-ins include a graphic frequency response. Equalizing, however, is analogous to adding spice to food—it's best when done in moderation.

Other times, you may want something more static so that you can make a more detailed investigation. Perhaps you're trying out the latest convolution plug-in and want to know what a given impulse response is adding to (or removing from) your material. In this case, a three-dimensional, time-varying

spectrum may be what you want. This is essentially a series of STFTs stitched together as a 3-D graph to show spectral evolution over time. Time and frequency are represented on two horizontal axes, and intensity is represented by height. **Fig. 4**, which was created using Steinberg's WaveLab, shows a time-varying spectrum of a vibraphone playing the note D3 (about 294 Hz). Although 3-D displays can be effective for viewing the behavior of individual harmonics, the contours of many envelopes are obscured by the contours of others.

Another time-based representation, the spectrogram (or sonogram), manages to collapse the three acoustic dimensions into two spatial dimensions. Spectrograms show frequency as a function of time, meaning time is represented by the horizontal axis and frequency is represented by the vertical axis. The intensity of a given frequency region is represented by shading, with greater contrast representing greater intensities. **Fig. 5** shows a spectrogram of a vibraphone sample created in AudioXplorer. Spectrograms are often used in phonetic analysis. They also can be helpful in targeting potential problems. An intrusive band of noise might show up as a dark horizontal streak. If you viewed which frequencies it spans, you might be able to remove the noise with filtering.

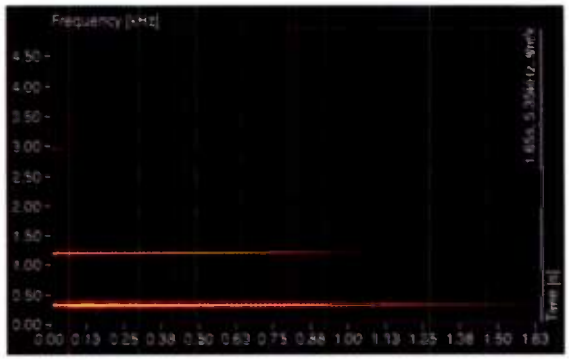


FIG. 5: A close look at a spectrogram shows what looks like low-level noise in addition to the harmonics, particularly within the first 5 ms or so.

LOOK WITH YOUR EARS

The various time-domain and spectral views of musical material are to audio producers what scales are to musicians. They are essential to audio ear training. The illustrations reinforce what you are hearing. Whether you are troubleshooting musical material in a hurry or making a careful study of your work, knowing how to look at what you are hearing can sharpen the decisions that you make.

Mark Ballora *teaches music technology at Penn State University.* *Special thanks to Kevin Larke and Kurt Hebel for their help in preparing this piece.*

JOHN HARRELL

Bassist Rudy Sarzo created the popular *Workingman's Bass* loop library. His list of credits includes playing with Ozzy Osbourne, Quiet Riot, and Whitesnake.

Award last year) and writing for sample and music libraries, including full-length CDs for Megatrax, Sonoton, FirstCom, Zomba, and ABC TV's Fifth Floor Music. He also managed to find time to release an album, *Electro Ave.* (Roir, 2002), which he and collaborator Chris Ingram recorded under the band name Asphalt Jungle.

Since the dawning of the age of MIDI, the soundware market has grown exponentially. You can buy patches for your synths, samples for your samplers, and loops-a-plenty for your DAW. A wide range of material is available, but hungry electronic musicians are always looking for fresh ideas. Perhaps you have what it takes to produce a sample-CD volume that people want to buy.

Musicians aren't the only people who are hungry for samples and loops. "Video producers, especially many nonlinear editors, are really getting into assembling their own music beds," explains Spotted Eagle. "Sony Vegas with Acid, Adobe Premiere Pro with Audition, and Apple Final Cut Pro with Soundtrack bring loop-based music to an entirely different audience."

MUSIC, MUSIC EVERYWHERE

Most of the music that you hear on radio commercials and talk shows and TV is not written specifically for a particular program. Producers rely on production music or library music to provide many of the tracks that they need for their projects. Dozens of companies supply music for both broadcast and nonbroadcast use. It is possible that you have what it takes to compose, record, and deliver a library music CD that people want to buy.

So, you have decided to investigate these opportunities further. Although there are a few subtle differences between the two markets, the basic approach remains the same: research your idea, create a demo, pitch your idea, produce your CD, and get paid.

BEEN THERE, DONE THAT

Typical sample and library CDs stick to a single theme or musical genre. The companies marketing these products want either something new or a unique spin on a tested theme. If you are offering the same old thing, you won't make the sale. Before you pitch your idea, first see if there is a need or a gap. A quick surf to a few Web sites will let you survey the landscape (see the sidebar "A Sampling of Music and Sample Libraries").

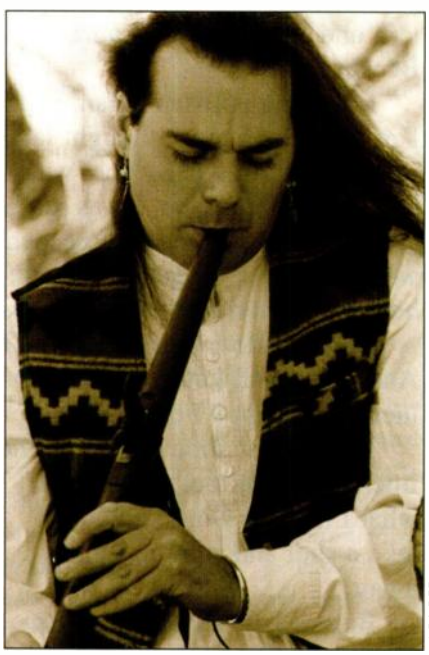
Pay careful attention to what's already on the market, and then gauge how your idea stacks up against the competition. Don't become discouraged if you find that there's a disc already out there that's similar to your idea. Check its release date; it may have been out for a while. Rethink your approach to come up with a fresh take on the idea.

Music libraries look for instrumentals in a wide variety of music genres. Although demand always exists for knockoffs of today's hottest trends, there continues to be a need for good solid tracks in common styles: upbeat sports themes; big, bold corporate themes; ethnic pieces; and new-age ditties and other low-key tracks for use under narration.

The typical track lasts between two and four minutes. "Producers prefer one CD that contains a number of variations on a particular theme—long versions, 29- and 59-second versions, short 5- or 10-second stingers, and alternative mixes without the lead instruments," says Tarquin.

All three musicians agree that a good sample library should have a wide variety of elements but should avoid overdoing it or getting too complex. "The loops or samples must offer tempos and grooves that are usable," says Sarzo. "Approach it like a studio musician—playing for the song rather than your own ego. If you're just soloing and showing off, it may sound great, but will it work in the context of somebody else's song?"

"Think simply and provide material that works with other things," Spotted Eagle echoes. "If you're going to do weird stuff, you'd better offer riffs that go with it and grooves for underneath. If you play an unusual chordal structure, provide other things that relate. Odd meter? Include drum loops or percussion to support it."



LINDA DEE

Douglas Spotted Eagle is an award-winning musician and producer. He has produced two sample libraries consisting of his unique Native American music and has worked on many *Loops for Acid* CDs.

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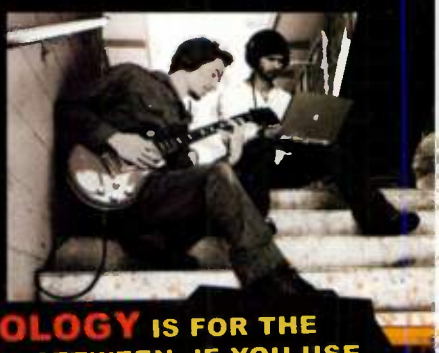

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A DEMO A DAY

While Tarquin, Spotted Eagle, and Sarzo possess extensive industry credentials, the rest of us need to put together a demo of the idea before taking it to possible buyers. "Music libraries are very approachable," admits Tarquin. "Research what they have already, then put together a demo of your best stuff that fills a gap in their catalog. When I worked at a jingle house in Manhattan, we had tons of demos and shelved projects that we earmarked for music libraries."

Do you have some unused tracks, songs, sketches, or miniscores sitting on tape or disk somewhere? Don't let all that music gather dust when it could put money in your pocket. For a recent pitch to a music library, I assembled 18 music tracks on CD—some full-length pieces and some shorter ones—all in a similar style and approach. The library accepted a few tracks as submitted, made suggestions for a couple of tracks, and trashed the rest. When complete, the final music-library disc will have about ten tracks.

"When writing for a sample library, have a clear-cut project and a solid idea of who the audience is. Show the company how you can fill its gap, and compile a demo CD that proves you can do it," suggests Tarquin. He says adding a biography and list of credentials can also help reduce any skepticism a company might have. "You don't need outstanding credentials as long as you have a good idea, can do the work, and appeal to the people you're selling to."

MAKE THE PITCH

Two events convinced Spotted Eagle to create a sample library. "I was getting a ton of calls from directors and producers to do flute cues. Then I heard a sample from one of my recordings used without permission in a major film. I thought that putting together a sample CD with Native American vocals, drums, and flutes would solve both issues."

Spotted Eagle approached Q Up Arts at a NAMM show and explained his vision of mapping drum hits and velocity

A SAMPLING OF MUSIC AND SAMPLE LIBRARIES

SAMPLE LIBRARIES

Big Fish Audio tel. (800) 717-3474 or (818) 768-6115; e-mail info@bigfishaudio.com; Web www.bigfishaudio.com

EastWest tel. (800) 969-9449 or (718) 932-6328; e-mail info@soundsonline.com; Web www.soundsonline.com

Ilio Entertainments tel. (800) 747-4546 or (818) 707-7222; e-mail info@ilio.com; Web www.ilio.com

M-Audio tel. (626) 633-9050; e-mail info@m-audio.com; Web www.m-audio.com

Q Up Arts tel. (800) 454-4563; Web www.quparts.com

Sony Pictures Digital Media Software (Loops for Acid) tel. (608) 256-3133; Web mediasoftware.sonypictures.com

MUSIC LIBRARIES

FirstCom Music, Inc. tel. (800) 858-8880 or (972) 446-8742; e-mail info@firstcom.com; Web www.firstcom.com

Fresh Music tel. (800) 545 0688; e-mail support@freshmusic.com; Web www.freshmusic.com

Killer Tracks tel. (800) 454-5537 or (323) 957-4455; e-mail production@killertracks.com; Web www.killertracks.com

The Music Bakery tel. (800) 229-0313 or (972) 414-0313; e-mail HelpNow@MusicBakery.com; Web www.musicbakery.com

Omnimusic tel. (800) 828-6664; e-mail omni@omnimusic.com; Web www.omnimusic.com

VideoHelper tel. (212) 633-7009; e-mail info@videohelper.com; Web www.videohelper.com

RICKY RESTIANO

Emmy Award–winner Brian Tarquin scores *All My Children* and writes for several sample and music libraries, including Megatrax, Sonoton, FirstCom, and Zomba.

for samplers. "They loved the idea, and the result was my original *Voices of Native America, Volume 1,* for Akai/Roland samplers. Later it was optimized as an Acid loop library.

"To do an ethnic vocal library like *Voices* was a big risk," he continues. "The response was amazing. Q Up has since put together other ethnic libraries, including my follow-up, *Voices of Native America, Volume 2.*"

Tarquin pitched his sample library idea at a NAMM show, too. "I was a fan of the Big Fish sample libraries, having used a few of their drum-loop CDs for some projects. I researched a few other libraries and saw a gap in the Big Fish roster. So at NAMM I met with them and pitched the idea for Big Fish Audio *Guitar Studio.* They asked me to prepare a short demo, and once they heard it, they loved it.

"*Guitar Studio* was unique at the time because it was live-guitar performances as 2-bar loops. I tried to cover as many styles, keys, phrases, chords, picking, and arpeggios using a variety of guitars," continues Tarquin. "They asked me to do a sequel because the first sold so well. For *Guitar Studio 2* [due out in 2004], I'm using many different amps, guitars, and mics to get a variety of sounds and tones."

Rudy Sarzo first approached Sonic Foundry at a NAMM show and expressed that he was a big fan of the company's Acid software (which is now produced by Sony). "Shortly after that, they approached me directly and asked if I would put together a bass-only library for their growing *Loops for Acid* product line. I agreed to do the first bass-only library, *Workingman's Bass.*"

DOWN TO BRASS TACKS

If you're fortunate enough to have your work accepted, you then have to get busy producing the CD. All production costs to create a finished CD—recording, talent, editing, mixing, and mastering—comes out of your pocket. Having a well-equipped project studio helps keep your costs lower, because you can do virtually everything yourself.

Turnaround times range from two to four months after acceptance. Audio CDs or digital files (WAV or AIFF) on CD-ROM are the preferred formats. Some companies prefer a completely finished project, while others will handle the final editing and mastering in-house.

Tarquin spent four months recording *Guitar Studio.* "It was hard, meticulous work. I had to keep the loops quiet and avoid breath and fret noise. I recorded it all on analog 16-track, 2-inch tape and mixed it to stereo DAT. Big Fish loaded the tracks into Pro Tools for all the editing."

It took Sarzo about six weeks to put together *Workingman's Bass*. "I tried every riff, groove, and one-shot I could think of in a variety of tempos and keys. Bass notes don't stretch as readily in Acid, so I had to cover more ground," explains Sarzo. "I recorded it all in my home studio using 4-, 5-, and 6-string Peavey Cirrus basses with Dean Markley strings through a Rupert Neve preamp into a Delta 1010 interface and directly into Acid. I used Sound Forge for editing. I kept the playing raw with very few effects so the end-users could tweak it to fit their music."

GETTING PAID

According to Spotted Eagle, "Sample libraries are usually a standard publishing deal ranging from 5 to 20 percent of net sales, paid quarterly. There may be an advance against those royalties, but usually not." Because samples and loops are sold royalty-free, there are no back-end royalties, just the sales royalty.

Music libraries, on the other hand, have many different methods of payment. Some composers may get a royalty based on sales, but that's rare. The usual deal is a one-time, up-front fee ranging from $200 to $1,000 per song. However, the composer keeps 100 percent of the writer's share of the publishing and may get a piece of the sync fees. In addition to the initial fee, you can potentially earn even more. For example, if a TV show uses one of your tracks regularly, you would see ongoing residual income (see the sidebar "The Back End").

One excellent reason to get involved with sample and music libraries is that they handle all of the promotion and sales. That means you can concentrate on your music. If you've ever just wanted to compose, record, tweak, and mix without most of the hassles of the music business, this side business may work for you. Tarquin says, "Big Fish and the other companies who purchase my music handle all of the marketing and advertising. I just compose and play."

DIY

If you want to take on a heavier workload, consider putting together your own music or sample library and promoting and selling it yourself. For a sample or loop library, record your ideas from scratch or edit them from existing performances. Make sure that you provide enough material to fill a CD and keep buyers happy.

For a single music library CD, you should have at least ten full-blown tracks. As with writing for other libraries, you can use music that you've already recorded or you can write something new. It is best to either provide a mix of popular music styles or stay in one genre. Sell the library as a nonexclusive buyout, which allows you to make money by licensing the same CD repeatedly to different buyers. If you haven't already, you should join a performance-rights organization (see the sidebar "The Back End") and tell buyers who use your library to indicate your authorship and affiliation on cue sheets. That way, you'll get paid your performance royalties.

Put together a demo with short snippets from each track or give away a few samples to encourage people to try before they buy. Market your demo on CD or from a promotional Web site. Put together the promotional material you need and start getting the word out to music buyers (video production houses, radio stations, advertising agencies, and so forth) and musicians. Finally, you need to make and sell the library. With a color printer and CD burner, you can manufacture your library CDs on demand or a few copies at a time. You could even offer your libraries as downloads.

"Libraries are a great way to get performance income," summarizes Brian Tarquin. "You may even do better than commercial radio."

Jeffrey P. Fisher's *book*, Profiting from Your Music and Sound Project Studio, *details ways in which to make more money with your gear. Find it and other resources at www.jeffreypfisher.com.*

THE BACK END

When writing for music libraries, you keep the writer's share of the publishing. Therefore, any publishing money collected by the library gets split 50-50 between the publisher (the library) and you (the writer). Because you won't sell sheet music and people won't "cover" your songs, your money comes from performance and synchronization rights.

Songs publicly performed on radio, TV, or in a restaurant or club earn royalties. These royalties are administered by three major performing rights organizations (PROs): ASCAP (www.ascap.com), BMI (www.bmi.com), and SESAC (www.sesac.com). PROs collect money by issuing both broadcast and nonbroadcast licenses. The broadcast licenses include the major television networks, public television, cable networks, local TV stations, and all radio stations. These media provide logs, called cue sheets, of all music played. Nonbroadcast licenses include orchestras, concert venues, restaurants and bars, and other public entertainment locations. Nonbroadcast performances are tracked by issuing blanket licenses. Money collected from these licenses is distributed to the song's writer(s) and publisher.

Performance royalties for music played on network TV can be substantial. Smaller stations and cable TV pay significantly less. Additional money, however, can be earned through reruns and syndication.

Although more rare, the writer may get a portion of the sync rights, too. When a song is used in conjunction with images in a commercial, film, or TV show, the song earns a royalty. Fees for synchronization rights are negotiated by the library and based on the song's use. These monies are above and beyond other royalties earned. In other words, the song still earns performance rights when a TV show also licenses the sync rights.

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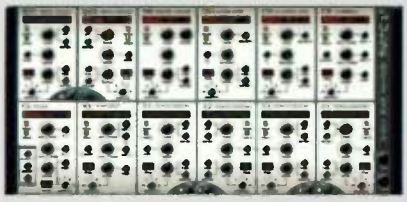

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– Computer Music, May 2003

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– Sound on Sound, June 2003

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– Keyboard, June 2003

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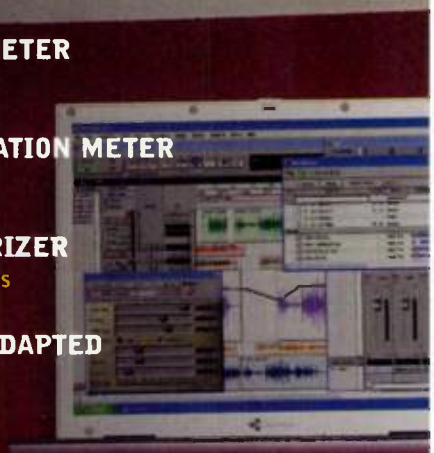

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REVIEWS

ARTURIA

CS-80V (MAC/WIN)

Modeling a true classic.

By Brian Smithers

Want to lose 200 pounds of excess weight and save more than $6,000 in the process? Compared with the original Yamaha CS-80 synthesizer, that's exactly what Arturia's CS-80V accomplishes. The CS-80 was an undisputed heavyweight in more ways than one, inspiring as much fear in roadies as awe in musicians. CS-80V is a soft synth that emulates the CS-80 in remarkable detail.

If a synth doesn't sound good, being able to tuck it under your arm in a notebook computer doesn't mean much. Fortunately, Arturia has done a good job of capturing the sonic character of the original behemoth. Additional niceties such as an arpeggiator, a modulation matrix, and a Multi mode bring the virtual instrument to a level the pre-MIDI CS-80 never reached on its own.

CS-80V is one of the most widely compatible virtual instruments I've encountered. In addition to standalone operation as a CoreAudio instrument in

FIG. 1: The layout of Arturia's CS-80V is faithful to the Yamaha CS-80, right down to the brightly colored factory Preset buttons. The arrangement of synthesis controls is intuitive, but the density of the display makes it difficult to read labels.

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- GSIF 2.0 for real time processing of incoming audio with NFX plug ins. For example, process vocals with GigaPulse™ Pro convolution without any noticeable latency.

- Real Time Surround Convolution. Sample developers can encode their instruments with acoustic space/reverb convolution and mic modeling using GigaPulse™ Pro included with GigaStudio Orchestra. You can play encoded .gig files back in any level of GigaStudio via GigaPulse™ SP. Up to seven channels of convolution is supported for surround sound applications

New Quick Edit window gives you instant access to commonly-accessed real time parameters. Perfect for moving a part down an octave, changing filter settings on a bass sound, shortening release time of a synth sound, or other common tweaks.

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GIGASTUDIO3.0

TASCAM

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Mac OS X and as an ASIO or DirectSound instrument in Windows, CS-80V supports just about every plug-in format imaginable. Under Windows, it supports VST, DXi, RTAS, and HTDM hosts; under Mac OS 9, it supports VST, MAS, RTAS, and HTDM hosts; and under Mac OS X, it supports VST, RTAS, HTDM, and Audio Units hosts.

Installation was quick and simple—for copy protection, you are required to enter only a serial number. Whenever you import a new bank, you must insert the installation disc.

CS-80V's interface, modeled after the original CS-80's, can be daunting at first. Dozens of virtual sliders, buttons, and knobs fill the front panel (see **Fig. 1**). You can simply choose Presets from a drop-down list; however, once you learn your way around the controls, you'll probably find yourself twiddling sliders and knobs endlessly.

At the bottom of the interface is a keyboard that lets you trigger notes with a mouse if your MIDI controller is out of reach. At the keyboard's left is a set of performance controls that include tremolo, portamento/glissando, sustain, delay, chorus, and pedal behavior. Above the keyboard are additional performance controls, including a ribbon controller, just like the one on the original CS-80. The arpeggiator, which is not included on the original equipment, sits at the left of the ribbon controller. The next row up offers a ring modulator, a suboscillator, two rows of Preset buttons, and an assortment of keyboard parameter controls.

The heart of the CS-80V is located in the next segment up, which contains two nearly identical rows of controls that map out the signal flow of the CS-80's synthesis engine from left to right. At the right of these controls is a Mix slider that determines the balance of the two signal paths.

At the far left of this section, the original CS-80 opened to reveal four sets of tiny sliders that corresponded to the main synthesis controls. Along with the main controls, these sliders defined the user Presets that were enabled with the white Preset buttons. CS-80V also hides controls behind a virtual flip-up panel, but it offers a modulation matrix in place of the Preset sliders. The matrix lets you map any of 12 modulation sources to any of 38 destinations. Up to ten such mappings can be stored for each Preset (see **Fig. 2**).

Above the main synthesis controls are CS-80V's Multi mode controls, with which you can assign up to eight Presets to any of four zones (see **Fig. 3**). You can detune, transpose, and pan layers, as well as define the key range of each zone. Controls for saving, loading, importing, and exporting Presets and other settings are at the top of the panel.

FIG. 2: For each Preset, CS-80V's modulation matrix allows ten mappings of control sources to synthesis parameters.

SLIP-SLIDING AWAY

Shaping CS-80V's sound is mostly a matter of tweaking sliders. It's an intuitive and highly visual approach. Dragging a slider with the right mouse button (on the PC) or while holding Shift (on the Mac) changes the scale of the slider's movement so that you can make finer adjustments. Very few of the CS-80's controls have center detents, so CS-80V makes no provision to normal a control. However, having the ability to type in exact parameter values would have been a time-saver.

The fact that the layout faithfully follows the original is a mixed blessing. Although the CS-80's configuration of controls is great on the front panel of a big old keyboard, it is difficult to see the labels of the various controls when so much is packed into a plug-in window. On a 17-inch monitor at Cakewalk Sonar's recommended 1280-by-1024-pixel resolution, I can't read the labels without leaning in for a closer view. CS-80V has a button that makes the panel disappear, leaving just the keyboard and performance controls visible. It would be useful to have another button that fills the plug-in window with just the two rows of synthesis controls. Another improvement would be to have parameter names and values displayed in larger type when you position the cursor over a control. CS-80V's user interface is one case in which being true to the appearance of the modeled instrument has a downside.

Minimum System Requirements

CS-80V

MAC: G3/500 MHz; 128 MB RAM; Mac OS 9.x or OS X 10.2

PC: Pentium II/500 MHz; 128 MB RAM; Windows 95/98/ME/2000/XP

Fortunately, memorizing CS-80V's layout is easy, so you won't be squinting for long. Every control can be assigned a MIDI controller, so serious tweakers might not look at the screen much at all. Many virtual instruments let you right-click to assign controllers, but because right-clicking is already assigned to making fine adjustments, Control-clicking (on the PC or Mac) opens a small dialog box in which you can either assign a MIDI Control Change (CC) message from a drop-down menu or enable Learn mode. To make an assignment in Learn mode, simply click on Learn and move a control.

ENERGY EFFICIENCY

Officially, the minimum CPU requirement for CS-80V is a 500 MHz Mac or PC, but I'd aim a bit higher. Although the software ran fine on my dated Celeron 1 GHz notebook, even at low latency, it pushed the CPU meter high enough that I would be skeptical of using it with a lesser processor. On my Athlon XP 2500+ desktop, performance was no issue. I've seen worse CPU hogs, but CS-80V is no lightweight.

Arturia had the foresight to allow the

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user to disable the chorus and delay effects, saving around four percent of the CPU load on my desktop computer. You can also disable the filters if you're not using them, saving a percent or two.

FIG. 3: Unlike the original, CS-80V features a Multi mode that allows for any combination of splits and layers among eight Presets. Each part can be detuned, panned, and modified independently of the other.

THE SAME, ONLY DIFFERENT

The sound of the original CS-80 is beefy, rich, and active. Between the ring modulator, suboscillator, effects, and keyboard-response controls, shaping static sounds is almost more challenging than creating sounds that blossom and develop. Arturia captured the evolving character of CS-80 sounds, and the additions of an arpeggiator and the ability to sync the LFO to MIDI are natural extensions of the original instrument's restless personality.

In the interest of science, I took CS-80V to the Audio Playground Synthesizer Museum to compare it directly with the genuine article. Because the CS-80 doesn't support MIDI, I couldn't load identical parameters into both, so making precise comparisons wasn't easy. I started with the factory Presets, which offer the presets of the original CS-80, according to Arturia. Some of them sounded similar, but none of them matched very closely. In fairness, it is difficult to say exactly how closely two CS-80s would match each other after so many years.

Using the factory Presets as a starting point, I began programming my own sounds. I stripped away everything but the raw waveforms to see how close the starting points would be, and the family resemblance got a bit stronger. As I compared filters, envelope generators, and every other parameter, I was never able to get an exact match.

You can't miss the similarities, however, between the CS-80 and its virtual counterpart. Think of Frank Sinatra at age 30 compared with age 50: each is objectively different, yet they are both unmistakably Frank. I can't say that the CS-80V sounded exactly like the CS-80 that I compared it with; there were times when I liked the sound of each better than the other. The original often had a thicker core sound than the emulation, but once you start using CS-80V's Multi mode, that advantage will be reduced if not eliminated.

PRODUCT SUMMARY

Arturia
CS-80V
software synthesizer
$249

FEATURES	4.0
EASE OF USE	4.0
QUALITY OF SOUNDS	4.5
VALUE	4.0

RATING PRODUCTS FROM 1 TO 5

PROS: Great sound. Support for many different audio engines and plug-in formats. Highly MIDI-controllable, including MIDI Learn function. Intuitive design. Multi mode allows layering and keyboard splits. Clear, detailed manual in English, French, and Japanese.

CONS: Controls are too small to read clearly. Makes moderately heavy demands on CPU.

Manufacturer
Arturia
tel. 33-438-020-555
e-mail info@arturia.com
Web www.arturia.com

Many of CS-80V's 400 or so Presets use Multi mode to good advantage. Those Presets are generally excellent and give you a good idea of the instrument's character and power. Note that switching to 96 kHz operation improved the overall sound of CS-80V, making everything a bit smoother, especially sounds with long releases.

Arturia's software emulates the Yamaha CS-80 closely enough that it's worth not having to maintain and move the original. The museum's CS-80 has not been fully restored, so it suffers from a noticeable hum—another point in favor of the virtual version. When you consider that functioning CS-80s are scarce and most musicians couldn't afford one, whether to buy CS-80V is an easy decision.

Brian Smithers *teaches at Full Sail Real World Education and Stetson University.* *Thanks to Joseph Rivers and the Audio Playground Synthesizer Museum (www.keyboardmuseum.org) for access to their CS-80.*

B I A S

PEAK 4.1 (MAC)

A perennial favorite stereo editor gets new tools and a more refined interface.

By David Rubin

After a decade of steady evolution, BIAS Peak continues to offer desktop musicians and sound designers one of the most flexible, straightforward, and feature-packed stereo editing programs for the Mac platform. EM last reviewed Peak in 2003 after the program had generated much buzz with its jazzy new Aqua-compatible user interface and its early support for OS X's Core Audio protocol.

Roughly a year later, BIAS debuted version 4, and although the current version is less revolutionary than evolutionary, it offers many noteworthy improvements. Most of the program's basic specs are still the same. For example, Peak still supports resolutions up to 32-bit and sampling rates up to a whopping 10 MHz. And the program supports an impressive and growing list of audio formats—both common and obscure—yielding wide-ranging compatibility for a variety of music, audio, multimedia, and sound-design applications. BIAS has also improved its audio engine by adding support for multithreading and multiprocessor CPUs. Further, along with its existing VST plug-in compatibility, Peak now fully supports Audio Units plug-ins.

EM thoroughly covered Peak's main features last year (see the Peak 3.11 review in the April 2003 issue), so to avoid repetition, I'll limit this review to the new tools and changes in version 4.1.

INTERFACE FINESSE

Although Peak boasts a number of under-the-hood changes, many of the program's new features are aimed at enhancing and further refining the user interface (see **Fig. 1**). For example, the Jaguar-style windows in version 3 now have the brushed-metal look of Panther and its family of applications. More importantly, the toolbar buttons now reside in a separate window that you can drag around and resize as a vertical or horizontal strip of any width or as a rectangular cluster of any size. The buttons can also be resized from a diminutive 16 × 16 pixels to an elephantine 64 × 64 pixels, which is good news for the visually impaired and anyone else who is tired of squinting at miniature icons.

Peak lets you assemble your own collection of buttons in the toolbar window, so you can have only your favorite tools appear onscreen. The program now provides buttons and reassignable keyboard shortcuts for most of its editing and processing commands. I love the flexibility and improved efficiency that the new toolbar window offers, although I wish BIAS would colorize the icons to make them a bit easier to distinguish.

The recording process in Peak is the same as it has been all along, but the earlier Aqua-influenced VU-style meters have been updated with more detailed high-contrast meters that look great. The meters still share the resizable Transport window with the transport controls and the redesigned and much clearer faux-LED Counter display. With the Transport window stretched all the way across the bottom of my 18-inch monitor, the meters span a respectable 10 inches. That, together with adjustable peak-hold settings, makes careful audio monitoring a snap. Just beneath the meters, a new Master fader lets you quickly adjust the playback volume with great precision.

Peak's Audio Document window is largely the same as in the previous version, although the overview above the main waveform display can now be

Minimum System Requirements

Peak 4.1

MAC: G4/300 MHz; 256 MB RAM (512 MB recommended); Mac OS X 10.2 or higher; hard drive with at least 18 ms average seek time; 330 MB disk space (for full installation with ImpulseVerb)

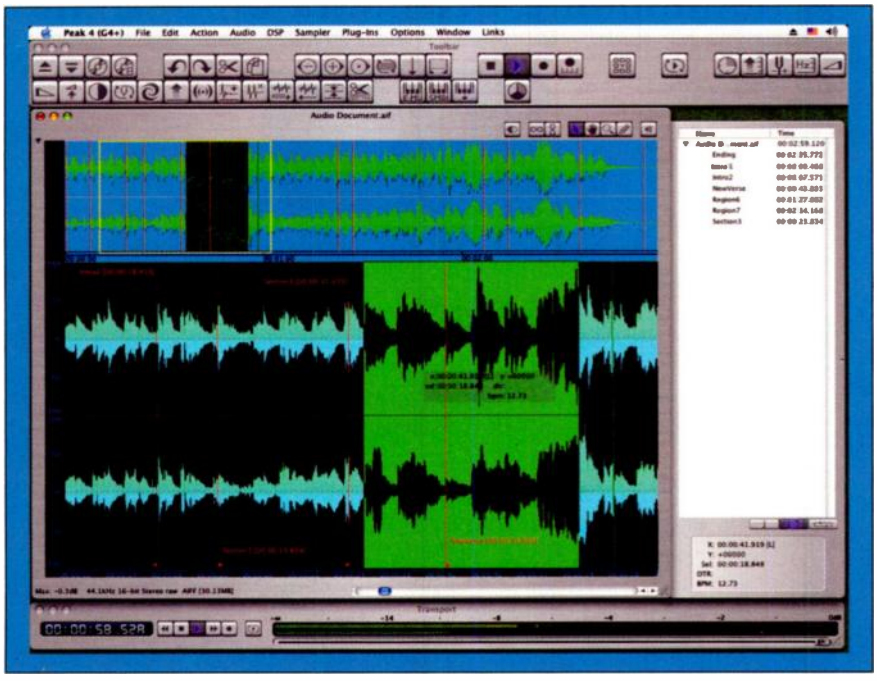

FIG. 1: BIAS Peak's new Panther-style user interface includes a slide-out drawer (on the right), an improved set of VU meters, a new customizable toolbar, and a cursor overlay with continuous readouts.

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hidden by clicking a small triangle in the upper-left corner. Peak still offers its assortment of 18 user-definable color schemes, with presets ranging from subtle to garish. As in earlier versions, I still prefer my own combinations.

The new Audio Document window differs from its predecessors by sporting an information drawer that slides out on the right side when you click a button in the upper-right corner. The drawer provides a variety of reference, looping, region, and other useful data along with a set of buttons for displaying the relevant details. At the bottom of the drawer is an x-y controller that also shows duration of the current selection, distance to the nearest marker, and beats per minute. (Peak can automatically guess the tempo of a selection.) The drawer is a nice addition that can improve your workflow in a number of projects.

A related workflow addition in the Audio Document window takes the form of an optional data field that follows the cursor as you move it around the waveform display. The lightly shaded, transparent cursor-information overlay provides continuous readouts of the same parameters as those that appear at the bottom of the slide-out drawer. The overlay lets you enlarge the waveform display to fill the screen horizontally without sacrificing the dynamic cursor information.

More often than not, however, I found that the overlay seemed to be in the way, and it was sometimes distracting when I wanted to simply eyeball an edit point. Additionally, the overlay field is frequently difficult to read—especially with certain color schemes—and its transparency allows the waveform to show through and compete visually with the readouts, which only adds to the distraction. Surprisingly, there's no toolbar button to toggle the overlay on and off; you have to use a key combination or select a menu item. I'm sure that some people will find the cursor overlay to be useful, especially for certain tasks, but I prefer using the slide-out drawer when I need a running update of important parameters.

FIG. 3: Peak 4.1 includes Sqweez, a high-quality VST compressor/limiter plug-in with 14 presets, covering a range of applications.

PLUGS AND PROCESSES

Peak's robust editing and processing tools from version 3 have been carried over to version 4 essentially unchanged. These include the program's unlimited Undo/Redo command (with random-access edit history), user-configurable fade-in/out envelopes, and ability to perform edit commands and processing during playback. However, several new features have also been added to Peak's growing feature set.

For example, the familiar Pencil tool has a new Magic Pencil mode. To eliminate a pop or an unwanted spike, simply click on the offending anomaly, and the Magic Pencil tool will smooth out the waveform automatically. This is a much simpler task than having to redraw the waveform by hand. I tested the Magic Pencil tool using a scratchy old recording and was pleased to discover that it worked very well. If cleaning up old recordings is important to you, you'll also appreciate Peak's improved waveform display that shows you all of the clicks and pops, regardless of image size.

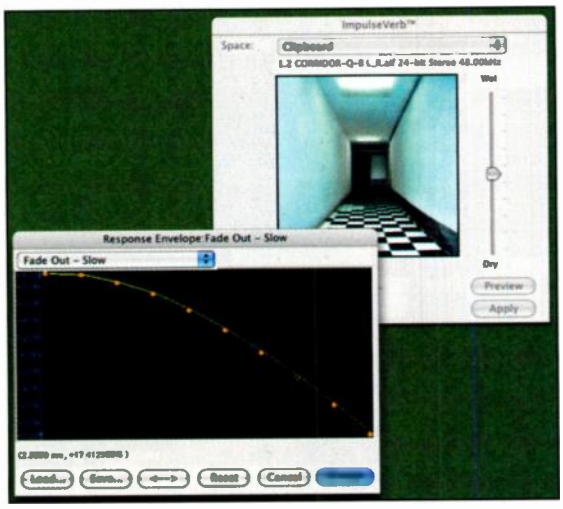

FIG. 2: Peak includes a new impulse-response reverb called ImpulseVerb. Its simple user interface belies its rich sound and powerful processing capabilities.

For most Peak devotees, however, the editing-tool enhancements are likely to be overshadowed by the program's significantly expanded plug-in support. As in version 3, Peak still supports VST plug-ins and comes bundled with Vbox SE, the flexible plug-in matrix for combining and routing effects. But with version 4, Peak has fully integrated Audio Units (AU) plug-ins, greatly expanding the available processing options. Additionally, the new Plug-ins menu offers five inserts per audio file, so you can mix and match VST and AU plug-ins as needed. Because Vbox SE appears as a single plug-in, you can create submixes of VST plug-ins and combine those with other mixes or submixes; if your computer is up to the task, the processing possibilities are staggering. As before, you can also use Vbox SE separately if you prefer to not use the inserts.

Peak has expanded its processing capabilities by adding several high-end DSP effects. My favorite is ImpulseVerb (see **Fig. 2**), an impulse-response-based reverb that leverages Peak's existing Convolution algorithm. The Altivec-optimized ImpulseVerb comes with a

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Produced by veteran sound designer Jennifer Hruska. Recorded by Emmy Award winning engineer Antonio Oliart and RIAA award winning engineer John Bono. Processed and programmed with the utmost in playability, this collection is a joy to create with.

Articulations:

Legato
Marcato Legato
Melodic Legato
Flutter Tones
Half Step Trills
Whole Step Trills
Staccato
Double Tongue "ta"
Double Tongue "ka"
Muted Legato
Muted Staccato
Muted Flutter Tones
Rips
Sforzando
Mute Sforzando
Stopped Horns
Horns Bells Up
Trombone Slides
Trombone Pedal Tones
Effects

Instruments:

French Horns
Solo Melodic French Horn
1st & 2nd chair French Horn Section
1st through 4th chair French Horn Section

Trumpets
Solo Melodic Trumpet
1st chair Trumpet
2nd & 3rd chair Trumpet Section
1st through 3rd chair Trumpet Section

Trombones
1st chair Bass Trombone
1st & 2nd chair Tenor Trombones
1st through 3rd chair Trombone Section

Tuba
C Tuba
Eb Tuba

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huge library of impulse-response files offering sampled reverb that captures a wide range of environments—from bedrooms, bathrooms, elevators, and corridors to Italian cathedrals.

ImpulseVerb's user interface is easy to use, with a drop-down menu for presets, a Preview button, and a Wet/Dry slider control. A Space Envelope button opens a separate window in which you can select or create an envelope shape that controls the reverb's length, attack, and decay characteristics. ImpulseVerb is easy to work with, and experimenting with it is a lot of fun. Best of all, it sounds terrific, with a number of presets offering a lush, full-sounding, natural ambience. I only wish there were a Bypass button for making quick before-and-after comparisons.

ImpulseVerb is not limited to using impulse-response files. Because the reverb is based on Peak's Convolve effect, ImpulseVerb has the ability to generate its effects from any file that is copied to the Clipboard, yielding an infinite palette of sound-design possibilities. For example, I used a recording of a mantle-clock chime and applied it to a recording of a nylon-string guitar. The resulting guitar track acquired metallic, ethereal sounds as if small bells were in the room and were responding to the music.

Another new DSP effect called Harmonic Rotate alters the frequency spectrum of a selected region by rotating the frequencies around a horizontal axis and assigning them to amplitudes from different parts of the selection. It's pretty hard to conceptualize, and it's even harder to predict the results. If you have the time to experiment, though, you might generate some worthwhile new sounds with this effect.

The Harmonic Rotate interface consists of an unmarked slider (to increase the effect) and two checkboxes, labeled Real and Imaginary, that apply different types of spectrum calculations. As you slowly drag the slider while in preview mode, you often pass through one or more "sweet spots" where the results are interesting and potentially usable. Although Harmonic Rotate is easy to use, it's a bit too unintuitive for my taste. Sound designers, however, may find it valuable for generating new sounds and textures.

Another of Peak's new effects is a high-quality compressor/limiter VST plug-in called Sqweez (see **Fig. 3**). It sports independent stereo input and output meters along with Threshold, Gain, Attack, Release, Ratio, and Knee controls. Auto Gain and Soft Clip modes are also provided. Sqweez's 14 presets cover a modest assortment of applications, from mastering to heavy compression.

With the introduction of Peak 4.1, BIAS has added yet another tool for processing audio files with delay, filter, modulation, and other assorted effects. Peak 4.1 includes The Sound Guy's SFX Machine LT, a "lite" version of the popular VST plug-in. SFX Machine LT includes 21 presets that have the potential to twist and mangle your audio files (in a creative way); an upgrade path to the full, 300-preset version is available. (For a review of SFX Machine, see the November 2003 issue of EM.)

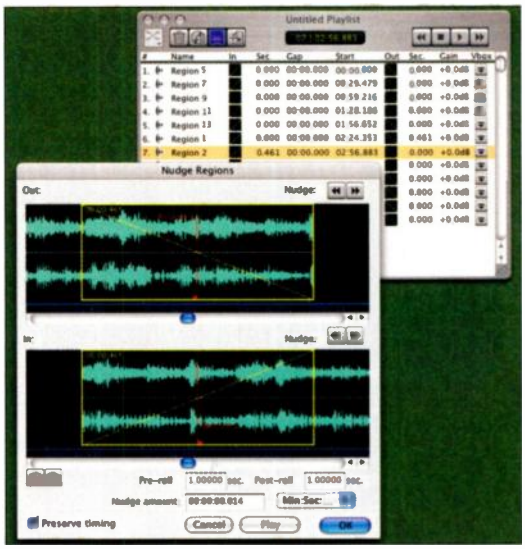

FIG. 4: Peak's powerful nondestructive Playlist window lets you apply real-time VST effects and custom fades to each track.

BURNING DESIRE

During the past several upgrades, Peak has continued to bolster its CD mastering and burning capabilities. In version 4.0, the program offered the ability to burn audio CDs directly from an audio document as well as from a collection of regions that had been assembled in the Playlist window. Peak's Playlist window remains one of its most powerful features, even if you aren't burning CDs.

The Playlist window allows you to organize regions in any order, apply user-definable crossfades before and after each track (region), specify individual gap times, adjust the gain for each track individually, and apply VST effects in real time to any track. The improved Nudge Regions dialog box lets you create custom fade-in/out curves, allowing for more control when crossfading between playlist events (see **Fig. 4**).

PRODUCT SUMMARY

BIAS
Peak 4.1 (Mac)
stereo audio-editing software
$499
Upgrade Peak $149
Upgrade Peak LE $399

FEATURES	4.0
EASE OF USE	4.0
DOCUMENTATION	3.0
VALUE	4.0

RATING PRODUCTS FROM 1 TO 5

PROS: Well-designed user interface. Supports high-resolution audio. Excellent meters. Full-featured nondestructive Playlist editing. Supports VST and AU plug-ins. Includes an excellent impulse-response-based reverb effect. Comes bundled with Vbox SE, Jam 6.0, and SFX Machine LT.

CONS: Cursor information overlay can be distracting when active. No support for external SMPTE time code. Printed manual must be purchased separately.

Manufacturer
BIAS
tel. (800) 775-BIAS or (707) 782-1866
email sales@bias-inc.com
Web www.bias-inc.com

It is disappointing that Peak still does not allow you to synchronize playlists to incoming SMPTE time code, so that you could trigger events at specific times while locked to a videotape workprint. That would make a handy post-production tool for nondestructively testing dialog, sound effects, or short music cues. Nowadays film composers and sound-effects editors can simply capture a video as a QuickTime movie and run it in sync within Peak, so this feature is not nearly as important as it once was.

The Playlist window's transport controls allow you to play the tracks (with the effects included) as if they were already on a CD. When you are satisfied with the results, you can burn the tracks directly to a disc or save everything as a Jam Image file for burning in Roxio Jam or Toast.

In fact, BIAS now bundles a fully functional version of Jam 6.0 with Peak 4.1—an excellent addition to the audio-editing software package. You can now move directly from your playlist into Jam to finalize the CD by adding track indexes, ISRC (International Standard Recording Code) data, PQ subcodes, and other important details that are necessary for mastering commercial CDs. You can also set Jam's preferences to open Peak as the external waveform editor for Jam. This feature ultimately allows you to edit seamlessly back and forth between the programs.

TAKE A PEAK

Peak's capabilities and its set of features have grown substantially since the last time that EM took a close look at the audio-editing software program. In addition to the main highlights covered in this review, some less flashy features are worth noting. For example, Peak now supports MP3 encoding rates up to 320 kbps, and the program now includes a handy Bit Usage Meter that allows you to analyze bit-depth and saturation problems.

In spite of Peak's many new features and improvements, the user interface remains attractive and accessible, and the program hasn't lost its intuitive design. I would still like to see SMPTE time-code support for the playlist, and I would like to have a printed owner's manual included in the package. (You can purchase a printed manual from BIAS for $25.) In general, though, it is difficult to complain when a high-end program like Peak adds a bunch of new tools and DSP effects and also throws in Jam 6.0 and SFX Machine LT, just to sweeten the deal. It seems abundantly clear that when it comes to Mac audio editors, the appropriately named Peak is once again at the top of its game.

EM *contributing editor* **David Rubin** *lives and works in the foothills outside of Los Angeles.*

APPLE

GARAGEBAND 1.1 (MAC)

Apple's virtual studio for the rest of us.

By Geary Yelton

If you're a Mac user, you're aware that Apple Computer has taken a keen interest in the music industry. In 2002 it acquired pro-music-software developer Emagic, and in 2003 it began selling recorded music on the iTunes Music Store Web site. Earlier this year, Apple introduced GarageBand, a virtual recording studio for Mac OS X. As part of the iLife '04 creativity suite, GarageBand is bundled with iTunes 4.2, iPhoto 4, iMovie 4, and iDVD 4—all without an increase in price. In addition, iLife '04 is included with every new Macintosh that Apple ships.

Just as Apple put video DVD production on the desktop of every Mac user who wants to make movies, GarageBand aims to put unprecedented music-production capabilities into the hands of even those who are nonmusicians. The program encompasses MIDI sequencing, loop arranging, multitrack audio recording, soft synths and samplers, and effects processing in a nonthreatening environment that anyone can enjoy.

GARAGE DOOR OPENER

You need an up-to-date, fairly powerful computer to run GarageBand. I don't have the latest Mac G5, but my setup is probably not too different from what is used in the average Mac-based project studio: a dual-processor Power Mac G4/1.0 GHz with 1.5 GB of RAM, four internal hard drives (which I defragment frequently), a 4× SuperDrive, and Mac OS X 10.3.3. My audio interface is a MOTU 2408mkII with a PCI-324 card, and my MIDI interface is an Emagic MT4.

iLife '04 took 21 minutes to install from the DVD, which upgraded prior versions of iPhoto, iMovie, and iDVD and installed GarageBand (I already had the latest version of iTunes). If you're installing iLife for the first time, expect it to take longer. Fortunately, installation ran in the background and involved no participation from me once it was underway. Before turning in this review, I updated to GarageBand 1.1.

FIG. 1: Apple GarageBand's user interface is designed so that almost any Mac user can figure it out, as befits all applications intended for mass consumption.

Minimum System Requirements

GarageBand 1.1

MAC: G4/600 MHz; 256 MB RAM; Mac OS X 10.2.6; 2 GB of hard-disk space

Although GarageBand is a surprisingly deep program (especially considering its cost), documentation is a bit skimpy. In addition to a collection of Apple Help Viewer files, it supplies 35 pages divided among four PDF files. Support pages are also available on Apple's Web site. Most of GarageBand is so easy to grasp, however, that detailed documentation is unnecessary. If you want to explore its finer points, though, you're either on your own or you'll need to purchase a book like Mary Plummer's *Apple Training Series: GarageBand* (Peachpit Press) or David Pogue's *GarageBand: The Missing Manual* (Pogue Press).

NEATER THAN MY GARAGE

Like other iLife applications, GarageBand was designed to be easy to use. If you're familiar with how sequencers are laid out, you should have no trouble finding your way around the program's user interface (see **Fig. 1**). Even if you aren't, you should be up to speed in a short amount of time.

When you create a new file, you are required to name it and specify the tempo, key, and meter in advance (you can always change those parameters later). GarageBand opens with a sampled grand piano track ready for you to play. The Track view takes up most of the main window, with controls on the left and the Timeline on the right. Each track shows the instrument's icon and name (both of which you can change), as well as the Mute, Solo, and Track Volume buttons. Each track also provides a level meter and controls for pan and volume. When you record something, data appears in the Timeline. At the top of the Timeline is a ruler that's divided by measures, and you can click and drag within the ruler to move the playhead or to define loops. Convenient keyboard shortcuts make navigation quick and easy.

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Below the tracks is a strip containing the Track Add and Track Info buttons, Loop Browser and Track Editor buttons, transport controls, time display, and the master volume fader and level meters. Brief explanations of GarageBand's various controls appear when you hold the cursor over them—a useful feature for beginners (and for reviewers trying to explain them using Apple's nomenclature).

THROWN FOR A LOOP

One feature that distinguishes GarageBand from most Mac software is its facility for working with prerecorded loops using techniques borrowed from Apple Soundtrack (see the February 2004 EM, online at www.emusician.com). Arranging loops in GarageBand is a lot like using Sony Acid. GarageBand furnishes over 1,000 Apple Loops in both audio and MIDI varieties. Each type is color coded (blue for audio, green for MIDI) and provides metadata containing keywords, tempo, key signature, and the like. You can edit the content of MIDI loops, and you can transpose and change the tempo and length of audio loops. GarageBand lets you create music in any of ten time signatures, but if you want to use the Apple Loops provided, it had better be in 4/4.

Clicking on the Loop Browser button reveals a cluster of buttons in GarageBand's lower portion, organized by instrument families on the left and genres on the right (see **Fig. 2**). Click on the control panel, and drag it upward to uncover additional choices. (An alternative view of the Loop Browser lets you display loops in hierarchical columns rather than buttons.) Clicking on any instrument button grays out all but the genre buttons, which offer styles such as Rock/Blues, World, and Acoustic, as well as moods such as Relaxed, Intense, and Cheerful. A list of Apple Loops that fit the categories you select will appear to the right of the browser buttons, each with its name, original tempo and key, and duration. Clicking on the list plays the selected loop in the current key and at the current tempo. For audio loops, you should select something close to its original key and tempo to get the best results.

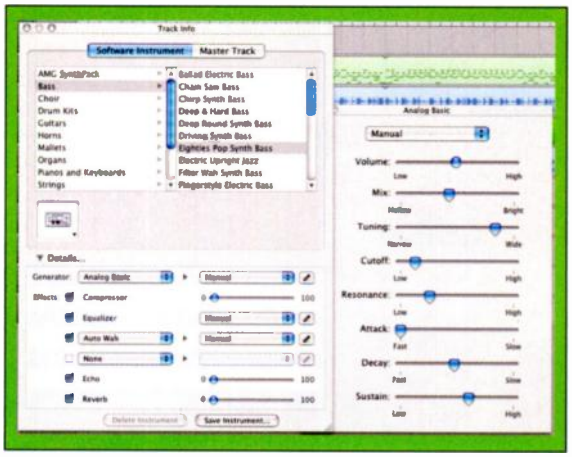

FIG. 3: Opening a Software Instrument's Track Info window reveals editable parameters.

Create a new track by dragging an Apple Loop into the Timeline, where it will snap to the nearest gridline. In the Timeline, you can transpose it, copy and paste it, drag its start and end points to shorten it, or drag its right edge to make it repeat. If it contains MIDI data, you can edit it as if you'd recorded it yourself.

The included loops cover a lot of territory, from basic instrumental riffs to complete arrangements. Unlike many loop libraries, the emphasis is not on dance-oriented grooves; a good variety of rock, jazz, orchestral, and other popular musical styles are well represented. If you want a larger library of Apple Loops, you can purchase additional content, such as Apple Jam Pack (see the sidebar "A Jam-Packed Garage") or one of several third-party offerings.

You can import loops from other sources simply by dragging files from the desktop into the Browser. If an AIFF, WAV, MP3, or unprotected AAC file contains its own metadata, GarageBand can translate it, but if the metadata doesn't exist, the Browser can't classify it. I imported Acid loops with their keywords intact, but GarageBand could neither transpose the loop nor change its tempo. Fortunately, you can use Soundtrack Loop Utility (available from http://developer.apple.com/sdk) to enter and edit Tags and Descriptors to convert any audio files into Apple Loops.

VIRTUAL OR REALITY?

GarageBand accommodates multitrack audio and MIDI recording, allowing you to record just one track at a time. When you click on the Add Track button, a dialog box lets you choose between a Real Instrument track and a Software Instrument track. Selecting Software Instrument adds a MIDI track with either a modeled soft synth or a sample-playback sound engine. Selecting Real Instrument adds an audio track with effects and dynamics processing appropriate for the instrument you select.

Clicking on a track type reveals a hierarchical list of instruments. If you click on Real Instrument and then Vocals, for instance, your choices will range from Female Basic and Gospel Choir to Male Speech and Radio Effect. If you choose Female R&B Vocals, GarageBand will add a touch of compression and EQ, some bright overdrive, and a bit more echo than reverb.

FIG. 2: GarageBand's Loop Browser displays loops by instrument family, musical style, and mood. It also lists each loop's tempo, key, and duration. You can mark the most useful loops as Favorites.

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D16XD

KORG

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Recorders shown with optional AIB-8 analog input board.

Other recorders available from Korg: D1600mkII, D1200mkII, PXR4, CR-4.

You can reassign the effects by clicking on the Info button and either selecting a different Real Instrument or changing the effects parameters. If the track is a Software Instrument, clicking on the Info button lets you edit synthesis parameters.

GarageBand's range of basic effects is respectable for any software, regardless of price. You can apply effects to any track, including Apple Loops. In addition to compression, EQ, reverb, and echo, each track provides two slots in which you can insert either one of GarageBand's 12 other native effects or an Audio Units (AU) plug-in.

GarageBand's most talked-about effect is guitar-amp modeling, and it is impressive. You can choose from four simulations: British Gain, British Clean, American Gain, and American Clean. Beyond that, the magic is in the presets, which allow you to vary preamp gain, three fixed bands of EQ, master gain, and Output Level. You also can add chorus, auto wah, and any of the other native effects. If you can play the electric guitar, GarageBand will show you a very good time (see **Web Clip 1**).

SYNTHS AND SENSIBILITY

Software Instruments let you edit their parameters, but you need to know where to look. Click on the Details button in the Track Info dialog box, choose a sound engine from within the Generator menu, and click on the pencil icon to see a list of editable parameters (see **Fig. 3**). The type of instrument you've chosen determines its selection of controls. Selecting a synth model might access sliders for waveform, filter cutoff and resonance, three or four envelope stages, and similar parameters. Sampled instruments typically provide access to volume, cutoff, attack, and release.

▼

If you can play the electric guitar, GarageBand will show you a very good time.

The assortment of Software Instruments is vast. It ranges from the basic guitar, bass, and drums to more esoteric selections such as Caribbean vibraphone. According to Apple, most of the sampled instruments have at least three Velocity layers, and some are as many as ten layers deep. Most guitars have six layers, including slides and string noise. The 200 MB, 8-layer Yamaha Grand Piano isn't great by modern sample-library standards, but it's probably as good as one you'll find in most synthesizer workstations. Considering the cost, though, it is unbeatable. (A marginally better Bösendorfer comes with Jam Pack.) All told, GarageBand provides about 1 GB of sampled instruments.

While exploring GarageBand, I noticed that the sampled Software Instruments have an EXS extension. Consequently, if you're an Emagic Logic user, EXS24 can read GarageBand instrument files.

AUDIO UNITS AND REWIRE

I had heard that GarageBand supported Audio Units, but I had no idea to what extent. I found no information in the PDF manuals, in the Help files, or on Apple's support site. I finally found the information I needed elsewhere on the Web.

AU support gave me access to some of my favorite effects and instrument plug-ins. To load an AU instrument plug-in the first time, you must add a Software Instrument track first, and then select Track Info to change it to an AU instrument. After you've done that, you can save an AU instrument along with its current preset so that it will appear in GarageBand's list of Software Instruments. Experimenting with various AU instruments, I was pleased to discover that LFOs, arpeggiators, and drum machine programs properly synced to GarageBand's tempo.

I tried using GarageBand with an enormous range of effects and instrument plug-ins. Some were native AU and some were converted using FXpansion's VST-AU Adapter. Out of dozens, I found only three that exhibited any problems at all, and those were graphic anomalies. Consequently, I was extremely impressed at GarageBand's AU compatibility.

A JAM-PACKED GARAGE

When you're ready to expand your timbral palette beyond the content supplied with iLife '04, the first add-on you'll want is Apple's Jam Pack for GarageBand ($99). Jam Pack is a 3 GB collection on DVD-ROM that furnishes more than 2,000 Apple Loops, over 100 Software Instruments, and better than 100 effects presets.

In addition to the Bösendorfer piano, new instruments include a 12-string guitar, more Eastern percussion, various synthesizers and organs, and a much wider selection of drums and basses. The loops cover an assortment of instruments and styles ranging from analog drum machines, club dance beats, and upright funk bass to jazz piano, lounge vibes, and orchestral strings. World music is particularly well represented, with loops containing non-Western instruments such as didgeridoo, santoor, oud, balalaika, saz, and even gamelan ensembles. Dozens of new acoustic guitar loops cover a lot of territory, too.

Although Jam Pack contains no actual effects plug-ins, it supplies plenty of fresh presets for GarageBand's existing effects. Of special interest are the electric guitar presets, which more than double GarageBand's complement of amp-modeling sounds.

The overwhelming bargain that is GarageBand extends to Jam Pack. It is without a doubt the best enhancement you can buy for under $100. If you enjoy using GarageBand, there's no question: get Jam Pack.

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DJ Spooky That Subliminal Kid

www.arturia.com

The latest version of GarageBand supports ReWire, but it isn't mentioned in the PDF manuals or the Help files. I did a search on Apple's Support site and found a brief document describing how to use GarageBand as a ReWire host: with GarageBand open, simply open a ReWire client and their transports will sync automatically. It doesn't get much easier than that. I tried it with Ableton Live and Propellerhead Reason, and it worked perfectly with both.

EDITING AND MIXING

Compared with more expensive audio sequencers, GarageBand's editing features are basic but functional. When you select a MIDI track and click on the Track Editor button, a familiar piano-roll display replaces the Loop Browser. A pull-down menu lets you display MIDI Note, Modulation, Pitch Bend, or Sustain data. GarageBand displays the latter three as straight lines with breakpoints. You can move the breakpoints, add new ones, or select and delete them. Although GarageBand does not allow you to edit other MIDI data, it records all MIDI Control Changes up to number 122; the only exception is MIDI Volume (CC #7), which you can enter and edit manually.

You can click and drag any note up or down to transpose it, drag it left or right to change its location, or type in a new Velocity. To change its duration, just click its right edge and drag. A Fix Timing button quantizes the selected note or region at the current grid level.

Mixdown is a straightforward affair. Each track's pan and level control lets you alter its position and balance in the mix. Clicking on a track's Track Volume button reveals a straight line to which you can add breakpoints for automating level changes. The Master Track provides volume automation, echo, reverb, EQ, compression, and one slot for insert effects. The Master Track Info dialog accesses control over echo and reverb parameters such as time, color, and volume. If you have a reverb that you'd rather use, the insert slot lets you select from your AU plug-ins.

When you have finished making your adjustments, all that is left for you to do is export your mix to an AIFF file. Just choose Export to iTunes from the File menu and you are done. If you want to save your work in MP3 or AAC format, you can do that from within iTunes.

PERFORMANCE ANXIETY

My complaints are few, and totally insignificant in light of GarageBand's price. When I first upgraded to the current version, I discovered that running GarageBand 1.1 simultaneously opened the application VirSyn Tera. I contacted Apple, which determined that I needed to remove Tera's ReWire extension to solve the problem. A more permanent solution should be available by the time you read this review.

I have more than 70 Audio Units plug-ins installed on my computer. Consequently, every time I opened GarageBand, it took a full minute and a half to appear onscreen—exactly the same length of time as Emagic 6.3.3 Platinum. Given the ephemeral nature of inspiration, it's easy to grow impatient waiting for any sequencer to open.

When I opened Preferences and clicked on the Audio/MIDI pane, I realized that GarageBand offers only two buffer sizes: large ("Maximum number of simultaneous tracks") and small ("Minimum delay when playing instruments live"). I'd like to have a bit more control over latency than that. You can, however, improve performance by specifying the maximum number of Real Instrument tracks, Software Instrument tracks, and voices per instrument.

DELUXE VALET PARKING

I have no doubt that GarageBand will fulfill its goal of making music production accessible to the masses. Furthermore, it makes a pretty good sketchpad for more experienced musicians. It's also useful for laying down grooves for improvisation in many musical styles. GarageBand is easy to learn and fun to use, and it offers enough flexibility and depth to be a truly useful songwriting tool (see **Web Clip 2**).

PRODUCT SUMMARY

Apple Computer
GarageBand 1.01
digital audio sequencer
$49 (iLife '04)

FEATURES	4.5
EASE OF USE	4.5
DOCUMENTATION	3.0
VALUE	5.0

RATING PRODUCTS FROM 1 TO 5

PROS: Easy to learn and easy to use, yet surprisingly deep. High fun factor. Excellent AU support. Loads of included content.

CONS: Can't control external MIDI instruments. Only two buffer settings. Can't import or export Standard MIDI Files. CPU hog.

Manufacturer
Apple Computer
tel. (408) 996-1010
Web www.apple.com

Apple has announced that a future version of Logic Pro will have Apple Loops functionality and the ability to open GarageBand files, so you will be able to begin projects in GarageBand and then finish the projects in a more professional environment. I wish that GarageBand could import and export Standard MIDI Files, but until it does, a third-party utility called Dent du Midi will import the individual MIDI tracks into GarageBand (www.versiontracker.com/dyn/moreinfo/macosx/22281).

Now that I've grown accustomed to using Apple Loops, I'm hooked. Throw in audio recording, virtual instruments, some nice effects, and basic editing features, and you have a winning combination. For the price, nothing comes close to GarageBand. If you use all five applications in iLife '04, it works out to just $10 per app. Even if you use only GarageBand, half a C-note is a bargain in anyone's book. And if you buy a new Mac, it's free; who wouldn't like the sound of that?

EM WEB CLIPS

Associate Editor **Geary Yelton** *has been reviewing Mac software for* EM *for more than 18 years.*

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AUDIX USA

MICROS M1245 AND M1290

The little microphones that could.

By Eli Crews

Microphones in Audix USA's Micros series are high-quality diminutive condensers that can be used for studio and stage applications. According to Audix, the Micros are the smallest condensers with integrated preamps and detachable cables currently on the market.

This review focuses on the M1245 and M1290 cardioids. Six other Micros models also are available: the M1244 cardioid, which is similar to the M1245 but has a higher maximum SPL rating and lower sensitivity; the M1244-HC, M1245-HC, and M1290-HC, which are all hypercardioids; the M1290-O, an omni model; and the M1290-S, a supercardioid.

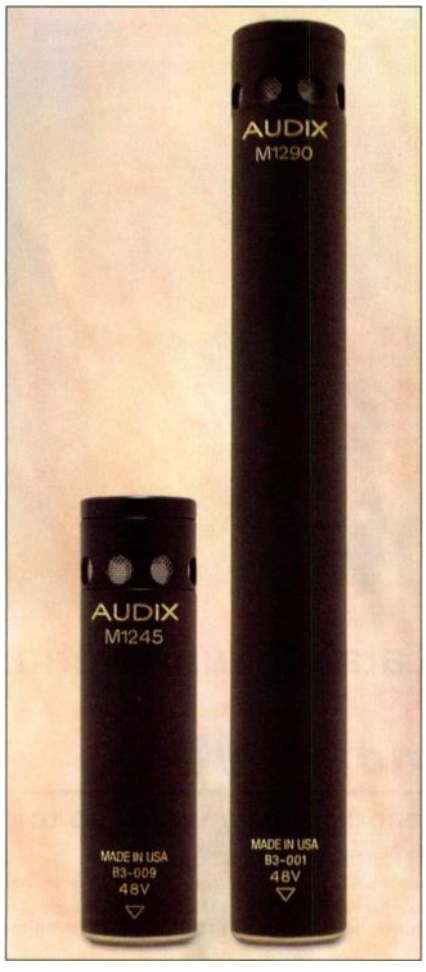


FIG. 1: Audix's M1245 (left) offers excellent sound quality and a small footprint. The M1290 (right) excels as a room mic.

THE GOODS

The M1245 and the M1290 (see **Fig. 1**) are strikingly slender—the diameter of the microphones is roughly half as big as that of a typical small-diaphragm condenser's. At 3.5 inches long, the M1290 is about the same length as many small-diaphragm condenser mics. The M1245 is only 1.7 inches long, which is impressively small.

Both mics feel solid and well made. They have a front-address design that picks up through several screen-covered holes in the housing, and each mic has a male mini-XLR connector on the other end. The mics are marked clearly with the brand name, model number, serial number, and polar pattern.

Micro series mics come standard with a rosewood case, an MC-Micro mic-stand adapter clip, a windscreen, and a 12-foot mini-XLR-to-standard-XLR cable. You can also choose from a selection of optional accessories, including clips, stands, mounts, and cables (see the sidebar "Micro Accessories").

I reviewed a pair of the M1245s and a pair of the M1290s. Although they weren't matched pairs per se, there was no discernable difference between the mics in each pair when tested on a variety of full-frequency sources. That indicates good quality control.

CAUGHT IN THE TRAPS

Because Audix is widely respected for its drum mics, I began by using the M1290s as drum overheads and the M1245s as tom mics for a session with a garage-rock drummer. The mic on the rack tom was held in place using Audix's optional Dvice-Micro clip, which has a spring-loaded mechanism for clamping onto the rim of the drum.

Clamp-on drum miking is unquestionably convenient, but getting the mic into the right position can be a challenge. Although the Dvice-Micro clip has a flexible goosenecklike shaft, you can move it just so far. You mainly control the angle at which the mic points at the drum. For the M1245 on the floor tom, I used a regular boom stand and the MC-Micro adapter clip, which enabled better control over positioning.

I supplemented the Micros with three additional mics (on the snare, on the kick drum, and in the room), and I was able to get an excellent drum sound in my studio's large, live room. The M1290s, positioned as overheads in a spaced-pair configuration, picked up the sound of the room quite well. They produced realistic imaging and fantastic detail, especially on the decay of the cymbals. With only a tiny amount of room mic in the mix, I achieved a live and open drum sound.

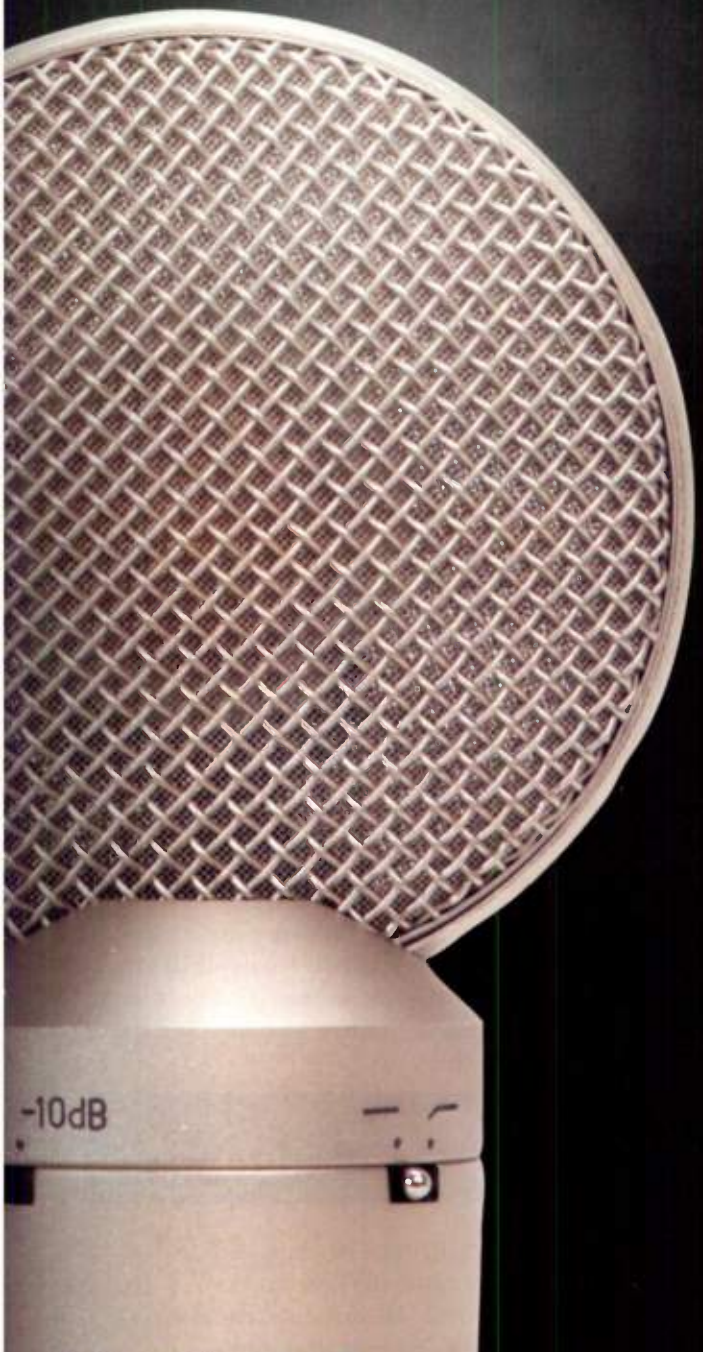
Some mics sound muddy and washy as overheads in this particular room, but the M1290s were clean in the low mids, thus avoiding the need for any EQ. (Given their excellent performance as room mics, their small footprint, and the convenience of the optional hanging mounts, the M1290s would likely be a good choice for live recording situations.) I stopped grumbling about the problems positioning the M1245s when I heard how great the toms sounded in the control room. They were round and full, with plenty of stick attack, but not boomy.

TESTING, TESTING

After the drums, I tried the Micros on a variety of other sources. First, I used the mics on a small Gallien-Kruger combo bass amp, an application for which they weren't well suited. The M1245s captured none of the bottom of the bass (because they roll off at 80 Hz), and the M1290s (which roll off at 40 Hz) offered some low end, but the overall sound was too flabby and distorted to be usable.

The Micros fared much better on an electric guitar played through a Bedrock tube amp and a 4×12 cabinet. I started with the M1290, slightly off-axis, placed approximately nine feet from the speaker. I then tried the same

The truth about microphones



Nova
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Luna
$249MSRP

Solaris
$349MSRP

It's a fact that most of the bargain-priced condenser microphones glutting today's market are made in China—many brands at the same automated factory. While M-Audio's manufacturing facility is also overseas, we utilize time-honored handcrafting techniques to create high-quality microphones from proprietary US designs and globally sourced parts. Example: While the tone-critical capsules of most inexpensive condensers are machine-manufactured from metalized plastic, we craft ours by hand from solid brass just like vintage designs with another zero in the price. For more examples of how our microphones have shattered the status quo to deliver the value that you've come to expect from M-Audio, download our free microphone guide today.

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MICRO ACCESSORIES

Audix offers a wide range of accessories for Micros-series mics. To attach a Micro to a mic stand, you can choose the SMT-Micro ($22.95), a rubber-insulated shockmount, or the Hanger-Micro ($19.95), a clear-plastic mount that hangs from the mic cable and has a hinge to facilitate precise positioning.

For attaching the Micros to percussion instruments, Audix offers two solutions. The Dclamp-Micro ($29.95) is a wing-nut-style clamp that attaches to drum or hand-percussion lugs. The Dvice-Micro ($29.95) is a spring-loaded clamp that attaches to drum rims.

You have two choices for low-profile microphone cables: a 25-foot CBL M25 ($19.95) and a 50-foot CBL M50 ($29.95). Both have a mini-XLR (male) connector at one end and a standard female XLR connector at the other end.

positioning with an M1245, an Oktava MK012 small-diaphragm condenser, and a Shure SM57 dynamic mic. The M1290 had a much nicer low end than the other mics. Sometimes it got a little too boomy in the low mids, but overall it had the best sound of the four mics at that distance.

When the mics were moved in close—on-axis and about two-and-a-half feet from the cabinet—the SM57 revealed why it is so popular for miking guitar. None of the condensers had comparable presence and definition. For the perfect guitar tone with these four mics, I would probably use the SM57 close to the cabinet and the M1290 as a distant mic to fill in the lower frequencies. The M1245 sounded very similar to the MK012 overall, with a little more grit on the more overdriven guitar sounds.

I also performed the key-jingling test to check the high-frequency response of each of the Audix mics. That revealed a natural high end that allowed the high mid and ultrahigh frequencies to be heard clearly without the harshness that some condenser mics impart. On a Chinese gong, which can have an especially harsh initial attack, the Micros sounded almost compressed, but in a nice way. The tail of

M1245 and M1290 Specifications

Element	condenser
Frequency Response	M1245: 80 Hz–20 kHz; M1290: 40 Hz–20 kHz
Polar Pattern	cardioid (M1245 also available as a hypercardioid; M1290 also available as an omni, a hypercardioid, and a supercardioid)
Connector Type	mini-XLR (adapter cable included)
Output Impedance	250 Ω
Sensitivity	14 mV/Pascal @ 1k
Equivalent Noise Level	19 dBA
Signal-to-Noise Ratio	75 dB
Maximum SPL	>138 dB
Housing	machined brass
Power	48–52V phantom power
Dimensions	M1245: 0.47" (W) × 1.70" (L) M1290: 0.47" (W) × 3.50" (L)
Weight	M1245: 0.6 oz.; M1290: 1.0 oz.

the gong, which lasts about 45 seconds, was clear to the end, with all of the nuances in resonance present—a testament to the low self-noise of the Micros.

After the gong, I tried the Micros on the bass marimba in my studio. This particular marimba is very resonant in the low mids, and it's tricky finding mics that don't accentuate that range. In a spaced-pair stereo configuration, the Micros captured the best marimba sounds I have ever recorded. Each pair had its strong points: the M1290s had slightly better ambience with a little more attack and mallet sound, and the M1245s were a little smoother in the problem area of the lower mids. Both pairs gave me gorgeous tones without any of the sharpness that I usually have to dial out when miking this instrument.

During regular sessions at the studio, I tried the M1245 and the M1290 on a variety of acoustic instruments, including trumpet, bass clarinet, and banjo. In all cases the mics provided passable results, but I ended up opting for other mics to track the instruments. Keep in mind that the mics I chose generally cost at least twice as much as the Micros, and that in the heat of a session, you usually opt to play it safe with your trusty standbys. But if I had wound up recording with the Micros in these situations, I'm confident they would have yielded usable, solid-sounding results.

SOUND INVESTMENT

Overall, I found the Audix M1290 and M1245 to be well designed and extremely useful. Not only are they excellent on drums and percussion, but they are also versatile enough for a number of other miking applications.

At close to $400 each, they aren't the cheapest mics, but they are still a great value. Whether they're your first small-diaphragm condensers or an addition to your existing collection, the Audix Micros are well worth a listen.

Eli Crews *is an engineer and musician based in Oakland, California. You can contact him through his studio's Web site, www.newimprovedrecording.com.*

PRODUCT SUMMARY

Audix

Micro M1245 and Micro M1290 cardioid condenser microphones

M1245 $379

M1290 $399

AUDIO QUALITY	4.0
VALUE	4.0

RATING PRODUCTS FROM 1 TO 5

PROS: Easy to position due to small size. Low profile makes mics attractive for live recording. M1290 exceptional at capturing room ambience. Smooth-sounding high mids. Wide range of accessories.

CONS: Not well suited (especially the M1245) for capturing bass or other low-frequency sources.

Manufacturer

Audix USA
tel. (800) 966-8261 or (503) 682-6933
e-mail info@audixusa.com
Web www.audixusa.com

JLCOOPER

CS-32 MINIDESK (MAC/WIN)

A diminutive control surface for home or away.

By Marty Cutler

Controlling a DAW with just a mouse can be a challenge. But many personal studios lack the space to house a full-size mixer or control surface. For studios in which space is at a premium, JLCooper Electronics offers the CS-32 MiniDesk, a 32-channel control surface that is small enough to sit on the palm of your hand.

ROTARY CLUB

The CS-32's channel-strip banks are divided into two groups of 16, with one bank placed above the other (see **Fig. 1**). Each channel has a miniature fader that offers a smooth, continuous resistance. Three small dual-function buttons, each not much bigger than the head of a pin, reside above each fader. The stenciled text that describes each function is tiny and extremely difficult to read (the white print on a gray background doesn't help). Fortunately, the same functions apply to each channel, making the button locations simple to memorize.

The first button above the faders toggles between track arming and muting. The button above that can select location-points or solo the track. The top button selects a track or enables the CS-32's solitary pan knob for that track. Overall, the CS-32 feels solid, but its small size necessitates that the controls be placed uncomfortably close to each other.

The angled panel above the fader banks has a two-digit Value/Mode LED, a track-null indicator, and six potentiometers. The functions of the pots depend on the software application being used: typically, one knob controls pan position, and the others control plug-in parameters or send levels. The jog wheel takes care of scrubbing and shuttle functions, and you select the mode of operation using the buttons above the wheel. Transport controls are conveniently located just above the jog wheel.

Near the transport controls are four cursor buttons. In MOTU Digital Performer 4.1, the left and right buttons control horizontal track magnification, and the upper and lower buttons control vertical magnification. In Digidesign Pro Tools, the buttons access a second bank of 32 tracks.

Above the cursors are nine user-configurable Function keys and a Shift button. The tiny red button at the top determines the operational mode for the three buttons above each of the faders.

MIDI OR USB

The CS-32 is available with MIDI ports or a USB port. I reviewed the USB version, which has its USB plug at the end of a 6-foot, nonremovable cable. (I wish it had a built-in jack that accepts a standard USB plug, because a hard-wired cable is difficult to repair or replace.) The USB-powered CS-32 must be connected directly to your computer's USB port or to a powered hub.

A ¼-inch footswitch jack sprouts from the USB cable just above the USB plug. Although this is fine for laptop use when your computer (and hence, the USB plug) is nearby, for studio use it would be more practical if the footswitch jack were on the rear panel of the CS-32. That way, the jack could be near the user, even if the computer were some distance away.

READY FOR ACTION

Concerning the CS-32's minimum system requirements: if your computer is capable enough to run the software that you want to control, it's capable of running the CS-32 software, too.

Installing drivers, called Keysets, from the CD-ROM that accompanied the CS-32 was simple. The disc provides Keysets for Ableton Live, MOTU Digital Performer, and Propellerhead Reason, but you can press the CS-32 into service for other digital audio applications by remapping functions in the provided software control panel

FIG 1: JLCooper Electronics' compact CS-32 MiniDesk holds a two-bank array of faders, each with dual-function, assignable buttons.

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IK MULTIMEDIA

MUSICIANS FIRST

(see the sidebar "Keyset Editor"). In its default state, the CS-32 works as a generic MIDI machine control device. I tested it with each of these programs using my dual-processor Mac G4/1.42 GHz, running OS X 10.2.8 and got mixed results.

You can download CS-32 software that supports Digidesign Pro Tools (Mac only), Cakewalk Sonar 2 and 3, Steinberg Nuendo, and Cubase SX and VST. JLCooper Electronics can also supply a beta Keyset for Emagic Logic, but as of this writing, it isn't posted on the company's Web site. In addition, the product's spec sheet lists support for Digigram Xtrack, Merging Technologies Pyramix, Sony Pictures, and Soundscape R.Ed (which is now Sydec Soundscape Editor), but I was unable to find software for these programs on JLCooper Electronics' Web site.

To operate the CS-32 with Digital

KEYSET EDITOR

Out of the box, the CS-32 sends MIDI Machine Control (MMC) messages. However, because different software packages require different sets of commands, JLCooper provides Keyset Editor, an application that can adapt the unit to your favorite digital audio sequencer (see **Fig. A**). The Keyset Editor turns the CS-32 into a control chameleon of sorts, adding considerable versatility to the unit overall. The GUI is a mock-up of the control surface, and the software lets you load the template for your application, verify that your controls are working properly, or create a new template (called a Keyset) if your music software is not directly supported.

FIG. A: The Keyset Editor application lets you customize the control surface and create your own templates.

From the File menu, you can create a new Keyset, Import a Keyset (you need to do this only once), or Export (save) a Keyset. The Edit menu lets you cut, copy, paste, or clear Keysets. Be sure to export any Keysets you have created or modified before you quit, as the application does not automatically prompt you to save upon quitting.

The Keysets menu provides a list of currently imported application templates. You use the Actions menu to set all controls to Native MIDI or Developer mode. The MIDI menu sets the CS-32 to the appropriate mode for the application: standard or enhanced variants of the CS-32 or CS-10 emulation, which emulates another JLCooper controller that is supported by other third-party software.

Although the CS-32 sends a set of fixed messages, the Keyset editor remaps the device's output based on your selections. Once you have edited the template, you can name and save it for later recall; the template remains in the Keysets menu, and the software will automatically configure the unit based on the application that is in your computer's front window. Once you have the software templates you need and the CS-32 is online, you do not have to load the application again.

You can edit a template while the hardware unit is running or offline. When you press a button, turn a knob, or slide a fader, the software reflects your physical actions by highlighting the equivalent graphic element. The upper-right-hand panel has three tabs. There, you can map a hardware move to a MIDI event, an ASCII keystroke (including modifier keys, such as Shift or Command), or use the Special tab for assigning a CS-32 action to Developer mode, if you want to develop an application that does not use MIDI.

A radio button beneath the MIDI tab lets you choose Native MIDI, in which a default control assignment is attached to a fader, button, or knob, or you can choose the Custom MIDI button to select from a drop-down list of MIDI messages. For example, you can assign a fader to send Channel or Polyphonic Aftertouch, or even Note-On messages. A pair of windows lets you set the value of the first and second Data byte, so you can constrain the values of your chosen data type. Another button lets you send the message over successive MIDI channels. If your system gets unruly, you could use this to assign Note-Off messages over multiple MIDI channels or just send a System Reset message. A selection box at the top right of the software lets you latch button presses so that a message assigned to it will remain in effect until you press the button a second time; otherwise, releasing the button turns the message off.

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WRH

Performer, you need to load the Keyset only once. With Reason, you have to load the Keyset and then launch the program from the provided Reason RNS template each time. Likewise, Live requires that you load its Keyset and then launch the program from a default ALS template.

When I began testing the unit with Digital Performer 4.1, I couldn't get the program to respond to any of the CS-32 controls. At MOTU's suggestion, I removed the Digital Performer Preferences file, which made the device responsive but unpredictable. For example, pressing a track-arming button muted the associated track. Then I discovered the red Mode switch, which toggles the behavior of these buttons. Unfortunately, neither the manual nor the related Digital Performer PDF document mentions the Mode switch.

I continued to have problems using the CS-32 with Digital Performer: the program quit unexpectedly when I switched to another application. I tested several driver upgrades, and although each added more stability, Digital Performer always crashed eventually. Removing the control surface driver from Digital Performer's setup menu returned Digital Performer's stability.

I also tested the CS-32 with Steinberg Cubase SX 2.01 under OS X. It took a couple of tech-support calls and some fiddling with the control panel to get the CS-32 to work properly. I ended up using the Cubase Device Manager, which lets you customize CS-32 functions, instead of using the CS-32's control panel. For example, I was able to assign an unused Locate button to bring up Cubase's Drum-Map editor, and another button to delete Continuous Controller messages. All in all, the CS-32 is well suited to handle just about any task in Cubase.

The CS-32's Reason template does not support track arming. Instead, the buttons take on their alternate role of muting tracks in the mixer. The Reason template provides 14 faders, with track 16 assigned to the Master Fader. The transport controls all worked fine, and the rotary controls in conjunction with the Pan Select button functioned the way they should. The Reason template leaves many unassigned parameters open, and I had great fun assigning and reassigning knobs and faders to sweep soft-synth parameters.

WORK IN PROGRESS

EM associate editor Geary Yelton also tested the CS-32, and his overall experience was similar to mine. One of his complaints concerned the footswitch jack. The first time he tried to remove a plug, the jack's plastic shell separated from its metal parts, stripping its threads and leaving the plug still firmly in place. After that, the only way to remove the plug was to firmly grasp the metal ring at the jack's terminus.

Ultimately, a controller is only as good as its software and the support the product receives. It is in these two areas that the CS-32 falls short: it lacks well-organized documentation, and, frankly, the technical support I received was disappointing. At the very least, the documentation should spell out all the steps required to get the CS-32 up and running, and not just most of them.

Configuring the CS-32 was often a matter of sifting through the hard-copy manual and a batch of individual PDF documents; consolidating that information would be a boon. If your experience is like mine, be prepared to contact JLCooper Electronics to get the CS-32 up and running with your system.

Once I got the unit set up properly, it worked fine—in most cases (its problematic behavior with Digital Performer remains a mystery). However, the value of the CS-32 is in its adaptability to so many software applications and its compact size, which makes it suitable for laptop-driven DAWs. Once the manufacturer sorts out the documentation, support, and driver issues, the full promise of the CS-32 will be realized.

Since leaving the Bay Area's sunny climes, EM *contributing editor* **Marty Cutler** *divides his time between teaching, doing freelance MIDI projects and gigs, and shoveling snow.*

CS-32 MiniDesk Specifications

Number of Channels	32
Faders	32 (reassignable)
Knobs	6 (reassignable)
Display	2-digit LED
Digital Ports	USB or MIDI
Other Connections	¼" footswitch
Power	USB port (external supply for MIDI version)
Dimensions	8.5" (W) × 1.5" (H) × 8.0" (D)
Weight	2.75 lb.

PRODUCT SUMMARY

JLCooper
CS-32 MiniDesk (Mac/Win) control surface
$499.95

FEATURES	3.0
EASE OF USE	2.5
DOCUMENTATION	2.0
VALUE	2.5

RATING PRODUCTS FROM 1 TO 5

PROS: Very small footprint. Supports a number of Mac and Windows DAW applications. Customizable. Fader and knob operations are smooth.

CONS: Controls are cramped. Scattered and incomplete documentation. Tiny, hard-to-read labeling. Minimal visual feedback.

Manufacturer
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R A N E

G4 QUAD GATE

An analog-digital hybrid that's the best of both worlds.

By Myles Boisen

The Washington-based Rane Corporation has been making solid, affordable audio gear for nearly 25 years. The company's well-built products are staples in countless sound-system and broadcast racks, but tend to have a lower profile in the studio market. That sleeper status may change, however, once recordists get their hands on the new G4 Quad Gate.

Admittedly, noise gates don't have the sex appeal of microphone technology, compressors, and the like. That may be because gates are (or should be) an invisible part of the process when they work as intended to control and reduce unwanted noise or acoustic leakage. In addition, gating technology—like the mousetrap—worked well when it was developed in the 1980s and has seen few refinements since.

But the audio world, or at least the part of it that's hip to the complexities of gating, may beat a path to Rane's door to find out if this company has indeed built a better noise trap. Using 24-bit digital "look-ahead" signal processing to improve triggering functions, and including four channels of every possible control parameter for gating, ducking, expansion, and external sidechain integration, the full-featured G4 is worthy of close scrutiny.

THE FACE OF THINGS

With more knobs, switches, and lights than a densely packed airplane instrument panel (see **Fig. 1**), the two-rackspace G4 has four identical sets of gating controls. Channels 1 and 2 can be stereo-linked, as can channels 3 and 4, using switches conveniently located at the top of the front panel between channel pairs.

At the top of each control section are two multicolored horizontal LED meters. The top, 11-segment meter shows the sidechain input level in decibels relative to threshold (dBr). As input level increases, LED colors change from green (for –24 to –1 dBr) to amber (for 0 dB, or threshold) to green (for +1 to +12 dBr) to red for overload.

The lower, 7-segment meter indicates the amount of gain reduction, with a red LED at the far left showing closed status, amber lights corresponding to the –24 to –1 dB levels in the row above, and a green open LED directly beneath the sidechain's threshold indicator.

Two rows of four knobs each take up the bulk of each channel section. The black and gray knobs are rubber-coated for a sure grip, and each has a broad white stripe that is easy to see against the faceplate's gray-on-black lettering. All controls are continuously adjustable over a broad range of values appropriate for studio or live sound.

On the far left, the gray Threshold pot is larger than the others and ranges from –60 to +20 dBu. Next to it are the Attack (0 to 250 ms) and Release (25 ms to 2 seconds) controls. A black Hold knob allows the user to dial in an open gate interval (0 to 3 seconds) between the specified attack and release times. This useful feature keeps the gate from flickering on and off in response to sustained dynamics of the sort that are typically produced by electric bass and keyboards.

The bottom row of black dials begins with the Depth control (80 to 0 dB). This sets the amount of gain reduction that occurs once the gate is closed, and is a handy problem solver when gating signals with a small dynamic range or high levels of background sound. To the right of the Depth control is a Ratio knob (1:1 to 8:1), which functions only when the channel is in expander mode. An expander, or downward expansion circuit, is a gentler form of noise gate. Technically, expansion works like compression in reverse, by using RMS detection to reduce the gain on an audio signal as the signal gets quieter. The result is a gradual decay that can trace the dynamics of the original signal over a long period of time. This mode is most appropriate for decay or noise control in sustained sounds produced by a keyboard, a bass guitar, or an electric guitar.

The final two knobs add useful sidechain-EQ control to the G4's generous feature set. The Low-Cut (20 Hz to 5 kHz) and High-Cut (200 Hz to 20 kHz) dials enable fine-tuning of the gate's triggering signal. This sidechain-EQ function has several potential uses, such as keeping a loud cymbal hit from opening the gate on a tom track. You'd normally accomplish such tasks by multing the signal to be gated through an outboard equalizer and then feeding it to a gate's external sidechain input.

Sidechain EQ could also be used to filter out a strong bass note that might

FIG. 1: The two-rackspace Quad Gate offers four channels of gating controls that can be linked in stereo pairs.

otherwise cause a "false trigger" of the gate applied to a flute track. Special dynamics effects and creative remix tricks can be accomplished with this sidechain control as well, and its inclusion adds value to the unit by cutting down on the time and trouble associated with making outboard sidechain connections.

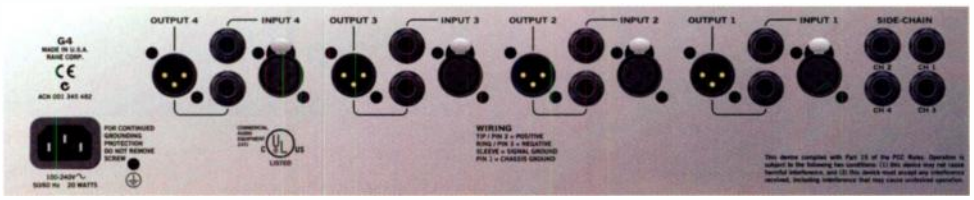

FIG. 2: The back panel of the G4 has four ¼-inch TRS sidechain inputs and XLR and balanced ¼-inch TRS inputs and outputs for each of the four channels.

At the bottom of each control section are four small toggle switches, each with its own multicolored status light. From left to right, these let you select Active or Bypass (for the gating circuit), Internal or External (sidechain-signal selection), Normal or Listen (to audition a sidechain signal), and Gate, Duck, or Expander mode. (Ducking is a process used mostly in the broadcast world—where the sidechain signal is typically an announcer's vocal—to automatically "duck," or reduce, the gain of background music. Some rock engineers use ducking to bring down the level of loud guitars or keyboards in a mix when the lead vocal is present.)

The unit does not have an AC-power switch. A small amber LED on the front panel's far right lights when the unit is receiving power.

BEHIND THE SCENES

Four sidechain inputs (¼-inch TRS) are clustered on the back panel, along with an array of XLR and ¼-inch balanced TRS inputs and outputs for each of the four channels (see **Fig. 2**). Although there is no printed indication of the reference level for the ¼-inch jacks, both input types are professional +4 dBu level, and there is no provision for –10 dBV signals. The back panel also contains a standard IEC power connector.

The G4's chassis is made entirely out of metal and has numerous ventilation holes on the top and bottom for cooling the multiple microprocessors inside. The unit gets warm to the touch and needs adequate ventilation when mounted in a rack.

The G4's exceedingly thorough manual provides textbook explanations of

G4 Specifications

Inputs	(4) XLR; (4) ¼" balanced TRS; (4) ¼" TRS sidechain
Outputs	(4) XLR; (4) ¼" balanced TRS
Maximum Input Level	+22 dBu
Maximum Output Level	+22 dBu
Frequency Response	15 Hz–20 kHz
Distortion (THD + Noise) at 0 dB Input	0.02% (20 Hz–20 kHz); 0.006% (1 kHz)
Threshold	–60 to +20 dBu
Attack Time	0–250 ms
Hold Time	0–3 sec.
Release Time	25 ms–2 sec.
Range of Attenuation	–80 dB to 0 dB
Low-Frequency Filter	20 Hz–5 kHz
High-Frequency Filter	200 Hz–20 kHz
Dimensions	2U × 5.25" (D)
Weight	7.3 lb.

all functions, as well as a useful primer on basic sound-system connection and wiring principles. One minor flaw in the documentation is that color-coding is mentioned for the purpose of deciphering oscilloscope chart displays, but all graphics are printed in black and white. Rane says that this flaw has been corrected.

The G4 uses carefully selected DSP chips to convert and process the analog audio at 24-bit depth and a sampling frequency of 48 kHz. The main advantage of digital A/D/A conversion, which is performed on the audio and the sidechain-input signals of each channel, is that look-ahead detection can be used to anticipate transient attacks in advance of the audio output. That allows for truly instantaneous triggering, so percussive attacks are never cut off or jumpy. Latency through the unit is only 1.6 ms.

SHOWTIME

The G4 Quad Gate saw a lot of use at my studio, Guerrilla Recording, during a month of tracking and mixing. On toms, kick drum, amplified instruments, and vocals, the unit was always quick to adjust and generally inaudible once set properly. I also threw some unusual challenges at the G4.

The toughest job for the G4 was on a prerecorded track of bassoonist Sara Schoenbeck, who was part of a 12-piece Anthony Braxton ensemble recorded live. Schoenbeck's instrument is especially quiet, and her custom miking system did not reject much sound from the saxes and trumpets positioned near her. After a brief period of fiddling with the Attack and Depth controls, I was able to get the G4 to work a gating miracle: attenuating the surrounding horns by 15 dB most of the time while keeping the breathy bassoon attacks tight and smooth without ever clicking or "cutting in."

The G4's immediate attack time deserves special mention. For years I've been tweaking attack times to avoid clicks or radical reshaping of transients, especially on wind instruments, so it felt like a luxury be able to set the attack at zero and leave it there to work flawlessly on a wide range of sources.

The G4 faced another intriguing trial during a soundtrack-recording session with multi-instrumentalist Mark Growden. One of Mark's motifs for the score was the sound of musical notes created by blowing on a series of tuned bottles. A click track in the headphones and the delicate and breathy nature of the bottle sounds combined to make yet another set of tough gating challenges.

As expected, attacks were smooth, and carefully adjusting the G4's depth, release, and hold settings was very effective for achieving a smooth decay with barely audible click-track bleed. I noticed no buildup of coloration or degradation in rough mixes of five or six bottle tracks recorded to analog tape using the G4.

I had two major concerns about the innovative process of A/D/A conversion used in the G4: latency-induced phase cancellation and tonal coloration. A series of studio tests allayed these worries and left me highly impressed with the unit's transparency. In normal usage—on toms or vocals, for example—the unit's latency was never a problem.

During my tests, the only time the inherent delay of the G4 electronics caused phase cancellation was when just one channel of a stereo pair was run through the gate. Of course, the need to gate one side of a drum set or piano would be rare, and processing both stereo tracks through the G4 would eliminate the anomaly.

When A/B testing stereo mixes through the G4 (circuitry active, with no gating), I listened carefully for minor timbral changes and any diminution in ambience, depth, or width that could be a by-product of A/D/A processing. The G4's converters did extremely well. Only in some big-sounding mixes with layered vocal harmonies did I hear evidence of any coloration, with a barely noticeable attenuation of high-frequency airiness. This level of fidelity shows a dedication to quality circuit design that would be laudable even in costly mastering gear.

GOLDEN GATE

The G4 Quad Gate is truly a best-of-both-worlds hybrid. With its potent combination of every imaginable analog control and top-notch digital processing, the G4 looks like the hardware gate to beat for the 21st century. Its logical layout, helpful metering, and generous range of parameters on each knob also make the G4 a delight to use. Thanks to its instantaneous attack time and look-ahead triggering, the G4 is as close as a gate can get to being a set-it-and-forget-it device.

The audio quality and implementation of Rane's innovative digital processing is also worthy of note. In fact, my only misgiving about this unit is that it doesn't offer computer-based studios access to its superb converters through digital I/O. But digital I/O is not free (yet), and it's quite remarkable that the G4's price remains competitive with the few other 4-channel gates out there. Rane tells me that a digital hardware gate is in the works. After that, perhaps the company will perfect the mousetrap as well! As it is, the G4 gets my highest recommendation. It's not only an excellent product, but it's also a tool that I intend to purchase and use regularly in my own studio.

PRODUCT SUMMARY

Rane
G4 Quad Gate
quad gate/ducker/expander
$999

FEATURES	4.5
EASE OF USE	5.0
AUDIO QUALITY	5.0
VALUE	5.0

RATING PRODUCTS FROM 1 TO 5

PROS: Complete selection of control functions for gating, ducking, and expanding. Generous range of control parameters. Top-notch 24-bit, 48 kHz internal digital processing. Look-ahead triggering for instantaneous attack time. Clear metering. Logical, easy-to-read layout. Highly informative manual. Competitive price.

CONS: No provision for –10 dBV–level operation. No digital I/O.

Manufacturer
Rane Corporation
tel. (425) 355-6000
e-mail info@rane.com
Web www.rane.com

Myles Boisen *is the head gatekeeper, janitor, and group therapist at Guerrilla Recording and The Headless Buddha Mastering Lab in Oakland, California.*

Quick Picks

APOGEE

Mini-MP

By Karen Stackpole

Apogee recently introduced an all-analog addition to its Mini-Series family, the Mini-MP ($995). This portable 2-channel preamp offers several features, including plenty of gain and a spiffy middle-side (M-S) matrix decoder mode.

All in the Family

With its sleek silver chassis, old-school toggle switches, and stylish purple pots, the ½-rackspace Mini-MP resembles its siblings, the Mini-Me and the Mini-DAC (reviewed in the March 2004 issue). Each channel offers a 48-volt phantom power switch, a selectable highpass filter with an 18 dB-per-octave slope below 80 Hz, a polarity reversal switch, and a continuously variable pot for up to 70 dB of gain. A centrally situated Output control provides an additional 6 dB of gain. Each channel sports a four-segment LED meter indicating (in VU) –20, 0, +6, and Clip. A three-position power switch can be set to Off, ST (stereo), or MS (middle-side mode). The output level knob is backlit, emitting a blue glow when the unit is on. Engaging phantom power activates a blue LED.

Each channel's Neutrik combination jack accepts both XLR and ¼-inch instrument inputs. Two electronically balanced XLR outputs are designed to mimic transformer behavior by providing self-adjustment and better balance when sending a signal to an unbalanced input. A jack labeled 6–14V DC accommodates the included lump-in-the-line power supply. According to the product literature, the Mini-MP accepts various battery and third-party power-pack options, a valuable capability for field recording.

The ½-rackspace Mini-MP mic preamp features two channels, each with phantom power, highpass filter, and phase reversal switch.

On the Job

I used the Mini-MP at my home studio and in the pro-quality studios at Ex'pression College for Digital Arts in Emeryville, California. Using both the stereo and middle-side options, I recorded electric and acoustic guitars, drum kit, hand percussion, and vocals. I tried various mics, including a pair of Schoeps CMC-6/MK4s, a Shure KSM44, a Blue Baby Bottle, a Royer R-121 ribbon mic, and several others. I also conducted controlled comparison tests with a Focusrite Green Series dual-channel preamp, a Langevin Dual Vocal Combo, and the preamps on my Mackie 1202-VLZ Pro mixer.

The Mini-MP provided clarity and realism, accurately representing the sound source and the mic capturing it. In general, high-frequency content, such as that produced by the finger taps on a dumbek or a strummed acoustic guitar, was comparatively bright but not brittle or glassy. The preamp was especially flattering to somewhat darker sounding mics like the Blue Baby Bottle and the Shure KSM44. The Mini-MP was as clean and clear as the Focusrite preamp and smoother and warmer sounding than the Mackie. The Royer R-121 ribbon mic required more gain, and cranking the level also raised the Mini-MP's noise floor. Though noticeable, the increased noise wasn't distracting.

However, when I plugged in either an Epiphone Sheridan guitar or a Yamaha bass (instruments with passive pickups), the elevated noise floor was more apparent. Apogee, which tested the review unit with a Stratocaster and found acceptable noise levels, says it has received no complaints and that the Mini-MP's higher input impedance may account for a slight noise-floor increase with certain guitars. Even with some extra noise, in instrument mode the sound of the Mini-MP was very good, though not quite as smooth and rich as that of the Langevin, a much more expensive unit.

The Mini-MP's middle-side decoding capabilities were exciting, especially for a compact unit. Recording a drum set with a pair of multipattern condensers produced expansive results. When the switch is set to M-S mode, the left channel controls the middle mic (set to cardioid pattern), and the right channel controls the stereo spread of the side mic (set to figure-8). You can route the signals from the mics through the Mini-MP's M-S matrix as you're tracking, or record the mic pair directly to separate tracks and route them through the matrix during mixdown.

Crisp, Clean, and Light

The Mini MP is an excellent preamp with great features in a compact package. Its output is clean and bright without sounding hyped or brittle, and its overall sound is smooth and realistic. While the low frequencies are not as expansive as those produced by the more expensive preamps, the sound is smooth and realistic.

The Mini-MP is unbeatable in its portability, flexibility with power supplies, and ease of use. It would be a good value for any studio looking to expand its preamp options, or a coup for the engineer on the go.

Overall EM Rating (1 through 5): 4

Apogee Electronics Corp; tel. (310) 915-1000; e-mail info@apogeedigital.com; Web www.apogeedigital.com

SOUNDTOYS

PhaseMistress (Mac)

By Nick Peck

SoundToys has introduced PhaseMistress ($249), an analog-style phase shifter plug-in for Mac-based Pro Tools systems. PhaseMistress includes RTAS, AudioSuite, and HTDM versions of the plug-in and is compatible with Mac OS 9 and OS X. An iLok hardware key is required for copy protection.

PhaseMistress provides extensive control of phase parameters and modulation sources, allowing for a wide variety of effects. The front panel has controls for wet/dry mix, center frequency, resonance

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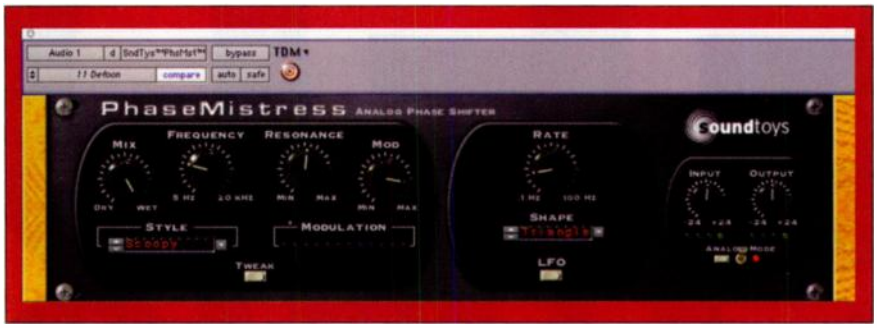

SoundToys' PhaseMistress is an easy-to-use phase-shifting plug-in that offers a comprehensive feature set.

amount, and modulation amount. The center frequency range is from 5 Hz to 20 kHz. A handy modulation meter shows the amount of modulation currently being applied to the signal.

A Style pop-up control selects the phaseshift algorithm. The styles range from emulations of vintage phasers (MXR Phase 90, DOD 201, Mu-tron Bi-Phase) to unique algorithms (WishWash, Scoopy, Primes). The factory patches are plentiful, well organized, and serve as useful starting points for dialing in a sound.

Use the Tweak button to open the Tweak window and create new styles. You can select the number of phase stages, notch and resonance phase direction, and intensity. The Modulation Tweak pane goes further, allowing you to adjust frequency modulation, resonance modulation, resonance offset modulation, and left/right offset.

Shape of Things to Come

The modulation section, which is comprehensive and well designed, offers six modulation types: LFO, envelope trigger, sample and hold, random steps, tempo-based, and an ADSR envelope. Only one modulation type may be used at a time, and each has a variety of specific parameters. For example, the LFO offers six standard waveshapes, as well as a shape editor for drawing your own. The Rhythm Modulation mode features tap tempo, selectable note values, a rhythm shape editor similar to the LFO waveshape editor, and a swing/shuffle groove control. According to SoundToys, the Rhythm Modulation, Sample and Hold, and simple LFO modes can all be synched to MIDI Clock.

Another nice feature of PhaseMistress is Analog mode, which models a tube amplifier by applying amplitude-dependent distortion and compression to the source, much as a tube circuit would. PhaseMistress's Analog mode is one of the best analog-distortion simulations I have heard in the digital domain. Although it's CPU-intensive, the improvement to the sound is worth the processing cycles. Analog mode offers seven flavors of tube modeling: Clean, Fat, Squash, Dirt, Crunch, Shred, and Pump.

Phasers On

I used PhaseMistress on a mono acoustic guitar track to create a wide, shimmery sound, with almost imperceptible animation. Using the envelope follower in conjunction with a heavy distortion setting gave a clean electric bass track a contemporary indie rock feel.

PhaseMistress can change drums radically, from adding a bit of whooshiness in the cymbals to transforming the kit into a techno gamelan. PhaseMistress also offers possibilities for lead-vocal processing, providing an alternative to the overly compressed, overly distorted, or telephone-filtered effects heard so frequently today.

Set to Stun

Within the realm of phase shifting, PhaseMistress covers a lot of ground. It's easy to use and loaded with features, but most importantly, it sounds musical in a wide variety of contexts. If you're looking to add some swirl to your sound, PhaseMistress delivers the goods.

Overall EM Rating (1 through 5): 4.5

SoundToys; tel. (802) 951-9700; e-mail sales@soundtoys.com; Web www.soundtoys.com

TRIDENT LOUDSPEAKER COMPANY

LS-101

By Mike Levine

The Trident Loudspeaker Company is a new venture for English audio-design guru John Oram. The company's first product is the LS-101 ($855 per pair), a passive monitor speaker. The LS-101s are close-field monitors designed to be used either in stereo—the application this review will focus on—or in a 5.1 configuration. (The company offers a matched system featuring five LS-101s and the LS-102 passive subwoofer.)

In and Around

The European-built LS-101 is 17 inches wide, 12 inches deep, and 7 inches high. It's encased in a heavyweight wood cabinet (the monitor weighs 20 pounds) with a cherry-wood veneer and a removable grill. It sports a pair of custom-designed 5-inch drivers and a 1-inch dome tweeter with a neodymium magnet. The rear of the cabinet has a circular bass port, as well as two pairs of gold-plated speaker terminals. The second pair of speaker terminals provides the option of setting up the LS-101s in a biwired configuration. (Biwiring, often used in audiophile systems, is a speaker-connection method in which a single amplifier uses two pairs of speaker wire per side to feed separate high- and low-frequency terminals on the monitors.)

The LS-101s, which are rated at 50W RMS at 4Ω, were designed to sit in a horizontal orientation on a console or desktop, but can also be set up vertically. The cabinets are magnetically shielded, which will keep them from causing interference when placed next to a computer monitor—a handy feature for DAW users.

Take a Listen

In my listening tests of the LS-101s, which have a frequency response rating of 30 Hz to 22 kHz, I found them to be smooth sounding, warm, and very detailed. Cymbals, tambourine, and other instruments with lots of high-end information sounded crisp and natural. Low-end reproduction was excellent, especially considering the size of the drivers. Kick drums and bass guitar were

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The LS-101 is a passive monitor that features two 5-inch drivers and a 1-inch neodymium tweeter.

full sounding, distinct, and not muddy.

Oram told me that he hopes that the LS-101s, with their wider frequency range, will be considered as replacements for the ubiquitous (and now out of production) Yamaha NS-10s. In fact, I compared the monitors to my NS-10Ms, and found that the LS-101s did sound more expansive: they had a bigger bottom end, a smoother middle and top, and a wider stereo image.

To test the LS-101s in a mixing situation, I used them to monitor a hybrid bluegrass instrumental mix featuring acoustic guitars, mandolin, dobro, and acoustic bass. I had previously mixed this piece on my NS-10s and a pair of smaller Radio Shack Minimus 7s, and was not happy with the results, especially the muddy bottom end. Not only did the LS-101s help me correct the bass EQ problems, but also they provided a clearer top end. I was able to distinguish the high frequency elements more easily and do a better job of balancing and equalizing them. The final result was much improved.

LS Is More

From the standpoint of sonic performance, the LS-101s are definitely winners. But whether the company succeeds with them commercially remains to be seen. The speakers face stiff competition in the crowded studio monitor field, and although they offer good value, they're certainly not a budget item. Another impediment the company faces is limited U.S. distribution. The easiest way to order them may be to contact Trident directly through the Oram Web site.

One thing is certain: if you're looking for quality passive monitors, the smooth-sounding LS-101s are sure to please. And for those who prefer powered speakers (an easier way to go, especially if you're thinking of using them for a 5.1 setup), the company plans to release an active version by this year's AES show in late October.

Overall EM Rating (1 through 5): 4

Trident Loudspeaker Company/Oram/Alphabet Belmont Group, Ltd. (distributor); tel. 44-1474-815-300; e-mail sales@oram.co.uk; Web www.oram.co.uk

U-HE.COM

Zebra 1.5 (Mac)

By Len Sasso

Zebra 1.5 ($199, free upgrade) is a significantly enhanced version of Urs Heckmann's unique Audio Units (AU) synthesizer plug-in. The fixed signal path has been abandoned in favor of a modular approach using a Grid that allows virtually any combination of modules to be linked across three main signal buses and an effects bus. New modulation options have been added including two 16-stage step-sequencers called MultiStages. In addition, two new effects are introduced in the Grid: a comb-filter and delay combination and a wave shaper.

Zebra's synthesis engine is built around four multiwaveform oscillators, a variable-timbre noise source, and three multimode filters. But that description belies the complexity that is possible here. In addition to the usual analog-modeled waveforms, you can use Zebra's Spectral editor to create your own additive waveforms with individual control of 127 overtones. In a nice touch, you can combine the Spectral waveform with an inverted copy of itself. You can then vary the time offset between the waveforms either manually or using Zebra's Matrix modulation. Modulating the time offset produces interesting cancellation patterns akin to comb filtering.

Zebra offers four varieties of low-pass filter as well as standard-fare resonant 12 dB-per-octave bandpass and highpass filters. Most interesting is its formant filter, which can morph across the five vowel formants (A, E, I, O, and U) and offers control of the depth of the formant effect. Both formant morphing and depth can be modulated. You can use the MultiStages to produce speechlike effects. Download **Web Clip 1** for an example.

Zebra's effects section contains three effects—delay, reverb, and a multi-effect (chorus, flange, and phase)—which can be arranged in any order and inserted in either the master output or a separate effects bus. Audio can also be routed to the send bus from Zebra's Global/Mixer module or directly in the modules Grid. One great feature of Zebra's effects is that any effect parameter can be modulated using Zebra's Matrix modulation or XY Assign features.

Got Stripes?

Zebra's oscillators, filters, noise source, and modular effects (waveshaper and comb-filter delay) are represented by graphic blocks that can be dragged to any point on the 4-column by 14-row Grid. The first three columns feed Zebra's Main, Sub 1, and Sub 2 outputs while the fourth feeds the effects bus. The outputs are mixed and enveloped in Zebra's Global/Mixer module. Setting up patches in the Grid is very simple, and there are even a couple of Scope modules that allow you to monitor the signal level at any point in the Grid.

To transfer audio between columns, Zebra provides four Line Mix modules. When one of those is activated, it can be set to route the output of the modules above it to any combination of columns. Line Mix modules can be set to merge the audio from other columns to their own column. Ring modulation of any column by

Zebra's control panel is split into upper and lower panes, each with multiple tabs. Here the upper pane shows the patch Grid as well as the individual module controls. The lower pane shows the Spectral Design window, in which you can create additive waveforms with as many as 127 overtones.

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any other is also built into the Line Mix modules.

Zebra offers three very powerful methods of modulating virtually any parameter of any module. First, there are built-in modulation controls for most synthesis parameters. Additionally, Matrix modulation allows you to apply any modulation source to any target and control the modulation amount by any other source. There are four Matrix modulation paths, and sources include three ADSR envelope generators, a ramp generator, three tempo-syncable LFOs, an assortment of MIDI messages, and the MultiStage step-sequencers previously mentioned.

XY Assign modulation uses four XY controls to affect any modulation target (which again means virtually any module parameter including the modulators' settings themselves). Each dimension can be routed to as many as eight different targets, each with its own range. This multitarget approach allows for extremely complex, yet often subtle modulation. A MIDI Learn function allows any dimension of any XY control to be assigned to any MIDI controller. MIDI Learn also works for most other Zebra controls, and happily, you can choose to have MIDI assignments apply globally or on a preset-by-preset basis. I set up the eight knobs on my M-Audio Oxygen 8 keyboard to control the eight XY dimensions, and then used the XY assignments for all MIDI control. In short, Zebra is a performer's dream synth.

Beast of a Different Stripe

There's no getting around the fact that Zebra is a complex synth. It takes a while to become a power user, but the manual is clear (if a bit terse), and a large collection of factory presets by a number of different authors provide great starting points for further tweaking. A run through the presets will also make it clear that Zebra is no ordinary sounding beast. You can download a time-limited demonstration of Zebra as well as a number of audio examples at EM WEB CLIPS www.u-he.com.

Overall EM Rating (1 through 5): 4.5

U-he.com; Web www.u-he.com

AURALEX ACOUSTICS

Aural-Xpanders

By Gino Robair

EM ★★★★★ hot pick

Adding to its respected Sound Isolation line of acoustical products, Auralex has released the Aural-Xpanders ($49.95). The 13-piece kit includes a 14-inch Xpander baffle, a pair of A-shaped Tri-Xpander baffles, four Trap-Xpander baffles, and six PlatFeet. The collection provides enough absorptive material for seven microphones and two mic stands—perfect for tracking a modest-size drum set. Auralex also offers the ISO Producer Pack ($119), which includes everything in the Aural-Xpanders kit as well as a set of MoPads Monitor Isolation Pads (reviewed in the July 2003 issue of EM) and a GRAMMA isolation riser.

Little Feats

The PlatFeet, which are designed to isolate the legs of a mic stand from a noisy floor, sparked my initial interest in the Aural-Xpanders. Most of my recording time is spent tracking concerts, and even when my mics are in a shockmount, it's easy to pick up the footsteps of an audience member walking by the mic stand. The PlatFeet seemed like a great solution for reducing this type of floor-based rumble.

Each of the pads has a 4-inch by 3-inch base, with a large notch cut into it from the top. The notch is cut inward so the extra foam wraps snugly around the stand's legs and keeps the pads secure. In the middle of the pad, on 0.5-inches of foam, is a plastic strip on which the legs of the stand rest. The strip helps protect the foam from damage over extended use and helps distribute the weight of the stand.

As I had hoped, the PlatFeet work like a dream with mic stands, and they are small enough to fit into my cable bag. Auralex also suggests using the PlatFeet on the legs of floor toms and cymbal stands to decouple them from noisy drum platforms and stages.

A Baffling Revelation

The Aural-Xpanders kit also includes pieces of foam designed for placement

The Auralex Aural-Xpanders are a collection of absorptive foam that can be used to reduce unwanted off-axis coloration for large- and small-diaphragm mics.

directly behind microphones. These mini baffles are especially useful in situations where you want to minimize 180-degree off-axis sound, but not completely isolate the mic from the resonance of the acoustical space—or block a performer's sightlines—with a gobo or large baffle.

Three sizes of baffles are included. The largest, the Xpander, is an hourglass-shaped piece of foam designed for use behind large-diaphragm mics. The outer edges are 7.5-inches wide, tapering to 5-inches in the middle. A hole in the center, with a slice leading to it on one side, allows you to easily attach the baffle to a mic stand. Exactly where you place the baffle behind the mic determines the balance of wanted and unwanted reflections, and the simple design of the Xpander lets you make adjustments easily.

The Tri-Xpanders are half the size of the larger Xpander and can be used with smaller mics. Their convenient size and shape also makes them perfect for isolating a pair of mics from each other (for example, when you're recording a singing acoustic guitarist and you want to minimize vocal bleed into the guitar mic and vice versa). The Tri-Xpanders also have a hole in the middle and easily attach to a mic stand.

While using these larger pieces on a session with horn players and a vocalist, I was concerned that the musicians (particularly the singer) would be distracted by having a foam wedge more or less in their face. However, no one minded, and the foam was a big help in filtering out unwanted coloration.

The asymmetrical Trap-Xpanders wrap around a small diaphragm mic and are designed for use on drum and percussion setups in which you want to lower the bleed of adjacent instruments. The foam pieces are 6 inches long and 1 ½ inches at the thickest point (with ridges cut into one side), and they taper from 3 ⅛ inches wide at one end to 2 ⅛ inches at the other end.

Placed around a snare drum mic, the Trap-Xpander can noticeably reduce hi-hat coloration in the snare mic. Positioning the foam is the key to success. With four Trap-Xpanders included in the kit, you have enough foam to isolate a snare and three toms. In a live session, I added a Tri-Xpander behind the bass drum mic to effectively lower the bleed from the horn players in the room.

Cut It Out

I'm always looking for ways to further isolate mics in a multimic setup without using up what little space I have. In these situations, the absorptive goodies in the Aural-Xpanders kit efficiently help to reduce problems due to off-axis coloration. And the PlatFeet are great for recording shows. At this price, the Aural-Xpanders are irresistible. ●

Overall EM Rating (1 through 5): 5

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

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(continued next page...)

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ADVERTISER INDEX

ADVERTISER	INTERNET	PAGE
Access	www.access-music.de	52
ADK Microphones	www.adkmic.com	109
Akai (Analog Filter)	www.akaipro.com	69
Akai (VST Plug-Ins)	www.akaipro.com	66
Apogee (AD-16X)	www.apogeedigital.com	118
Apogee (DA-16X)	www.apogeedigital.com	119
Arturia	www.arturia.com	103
AudioMIDI	www.audiomidi.com	123
Audix	www.audixusa.com	99
Auralex	www.auralex.com	55
B&H Photo-Video	www.bhphotovideo.com	89
B&H Photo-Video	www.bhphotovideo.com	125
Bag End	www.bagend.com	126
BIAS	www.bias-inc.com	113
Big Fish Audio	www.bigfishaudio.com	81
Cakewalk (Project 5)	www.cakewalk.com	83
Cakewalk (Sonar 3)	www.cakewalk.com	35
Carvin	www.carvin.com	56
Disc Makers	www.discmakers.com	68
Echo Digital Audio	www.echoaudio.com	147
Edirol	www.edirol.com	49
emagic	www.emagic.de	21
E-mu Systems (Digital Audio Systems)	www.emu.com	46—47
E-mu Systems (Emulator X)	www.emu.com	93
Europadisk	www.europadisk.com	91
Event Electronics	www.event1.com	65
Focusrite	www.focusrite.com	11
Full Compass Systems	www.fullcompass.com	121
Full Sail	www.fullsail.com	50
Garritan Orchestral Strings	www.garritan.com	73
Genelec	www.genelec.com	19
Graph Tech Guitar Labs	www.graphtech.com	127
IK Multimedia	www.sampletank.com	111
Ilio (Spectrasonics)	www.spectrasonics.net	25
Independent Music World Series	www.discmakers.com/imws	131
Jasmine Music Technology	www.JasmineMusic.net	10
Korg (D32XD/D16XD)	www.korg.com	101
Korg (Legacy Collection)	www.korg.com	3
Korg (Triton Extreme)	www.korg.com	2
Korg (Triton Le)	www.korg.com	29
Korg (USA Online Store)	www.korg.com	108
KRK Systems	www.krksys.com	27
Kurzweil	www.kurzweilmusicsystems.com	9
Lexicon	www.lexicon.com	67
Mackie	www.mackie.com	15
Mark of the Unicorn	www.motu.com	148
M-Audio (FireWire 1814)	www.m-audio.com	59
M-Audio (Mic Family)	www.m-audio.com	107
Muse Research	www.museresearch.com	54
Musician's Friend	www.musiciansfriend.com	105
Musicians Institute	www.mi.edu	79
Perfect Pitch	www.perfectpitch.com	128—129
PMI Audio Group	www.pmiaudio.com	97
Recording Workshop	www.recordingworkshop.com	127
Rode Microphones	www.rodemicrophones.com	71
Roland (Fantom-X)	www.rolandus.com	4-5
Roland (GR-20)	www.rolandus.com	61
RUDE! Audio	www.rude-audio.com	80
Shure	www.shure.com	41
Sibelius Software	www.sibelius.com	17
Sonic Implants	www.sonicimplants.com	95
SPL-USA	www.spl-usa.com	51
Steinberg	www.steinberg.net	13
Sweetwater Sound	www.sweetwater.com	23
Sweetwater Sound (Digidesign)	www.sweetwater.com	84—85
Sweetwater Sound (GearFest 2004)	www.sweetwater.com	43
Sweetwater Sound (MOTU Studio)	www.sweetwater.com	142—143
Sweetwater Sound (MOTU Studio)	www.sweetwater.com	144—145
Sweetwater Sound (PreSonus)	www.sweetwater.com	115
Synthax.com (RME)	www.synthax.com	75
Synthax.com (Samplitude 7)	www.synthax.com	117
Tascam (Frontier Computer Recording)	www.tascam.com	39
Tascam (GigaStudio 3.0)	www.tascam.com	87
Taxi	www.taxi.com	77
Universal Audio	www.uaudio.com	33
Waves (Enigma)	www.waves.com	55
Waves (Linear Phase Multiband)	www.waves.com	53
Waves (Processing Software and Hardware)	www.waves.com	57
Waves (Renaissance Compressor)	www.waves.com	51
West LA Music	www.westlamusic.com	53
Yamaha (MG-Series)	www.yamaha.com	31
Yamaha (01X)	www.yamaha.com	63
Zaolla	www.zaolla.com	90

ELECTRONIC MUSICIAN MARKETPLACE ADS

AEA	www.wesdooley.com	135
Apogee Digital	www.apogeedigital.com	134
BenchMark Media Systems	www.BenchmarkMedia.com	133
Clearsonic	www.clearsonic.com	132
Crystal Clear Sound	www.crystalclearcds.com	133
ELS Productions	www.elsproductions.com	132
Europadisk, LLC	www.europadisk.com	135
Extreme Isolation Headphones	www.extremeheadphones.com	135
Lonely Records	www.lonelyrecords.com	132
Media Services	www.mediaomaha.com	134
MIDI Warehouse.com	www.MIDIWarehouse.com	135
PcAudioLabs	www.pcAudioLabs.com	135
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Shreve Audio	www.shreveaudio.com	132, 133 & 134

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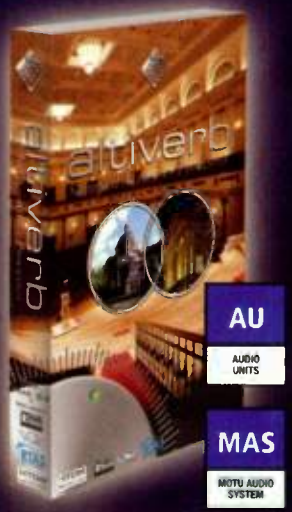

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By Larry the O

Every Which Way to Sunday

The idea of multitasking has become quite prominent in recent years. The term was coined for computers, but multitasking was with us long before the word existed. The concept is that, given a number of tasks to be completed, it is often preferable, and sometimes necessary, to execute them simultaneously.

However, in the world of personal computers, a single processor is capable of doing only one thing at a time (pipelining aside). The most common solution is preemptive multitasking, a microprocessor version of "sleight of hand," in which the machine consecutively executes a small piece of each task, rotating through the jobs so that all tasks are effectively performed in parallel. Make this process fast enough, and it appears to the user that the machine is doing the tasks simultaneously.

How did the concept exist before digital computers? Multitasking is an essential part of our personal and work lives. If you doubt this, just ask anyone who has had to take care of children while cooking a meal or doing other work. The ever-accelerating pace of life has placed increased emphasis on multitasking in recent years, and it was a no-brainer that the term would come to be applied to people shortly after it was coined for computers.

Although there are many levels at which multitasking can be viewed, the concept of human multitasking is perhaps most clearly illustrated by the challenges that those people who are in management positions have. After all, management is largely the wrangling of multiple simultaneous processes. While concert sound engineers must be aware of and, in turn, balance many things, they are focused on the one task of making the sound good. In contrast, the production or tour manager must worry about every aspect of the show from the sound to the hall's union help, the catering, and the condition of the truck drivers.

Multitasking is an increasingly essential skill, but there is an important difference between the multitasking that is done by a person and the multitasking done by a computer. If the software is well written (a big "if"), multitasking by a computer differs from executing tasks serially only in the time that it takes to complete a task; there is no difference in the quality of the work done.

Human beings, however, are not quite so simple as that. Of course, some people thrive on juggling a lot of things at once, but I have seen constant multitasking strain other people's ability to focus sufficiently on any of the tasks being handled, so much so that multiplicity almost cripples their ability to work. Results can include distraction, sloppiness, loss of perspective, anxiety, and numerous other undesirable side effects.

This is simply one more example of how necessary it is to have balance in one's life. For me, playing music, especially practicing, provides the kind of intense focus that can counterbalance the demands of multitasking. For someone else, it might be wiring a patch bay or painting a house.

As a child, my family had a neighbor who was an actor. Returning home after a performance, he often found release playing with Lincoln Logs. At the time, I thought that that was odd behavior for an adult, but it was explained to me by my parents that people with complex work also often needed to participate in simple play, and that somehow made sense to me.

Unfortunately, the downsides of multitasking are insidious in their manifestation, rather like quicksand: you may not realize what you've stepped in until you are already sinking. Fortunately, being able to recognize the pressures of multitasking is a powerful first step toward regaining balance, which, ultimately, gives better odds for escape than quicksand might.

Duality and paradox are ever present in our lives, and multitasking versus focus is yet another example of that. There is nothing inherently bad about multitasking; the pitfall to be avoided is losing one's balance.

Larry the O *is accustomed to spending hours alone in windowless rooms, during which time he fondles a small gold ring and mutters "My Precious!" a lot.*

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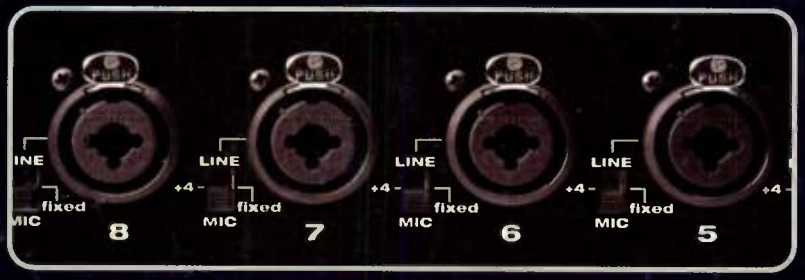

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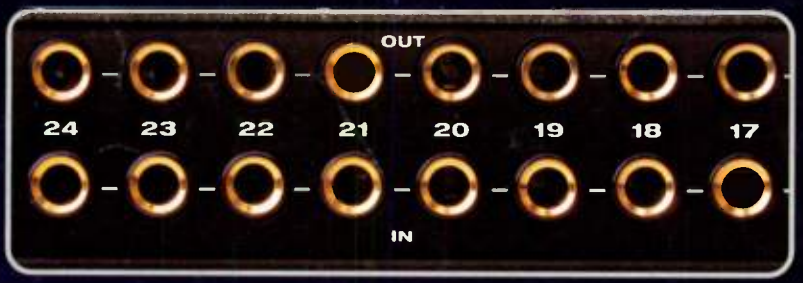

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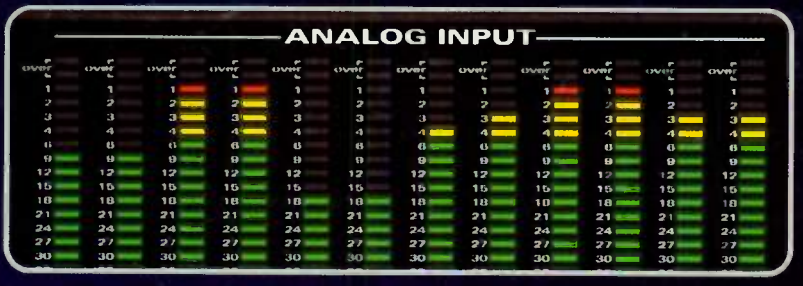

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