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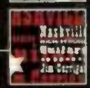
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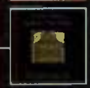
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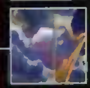
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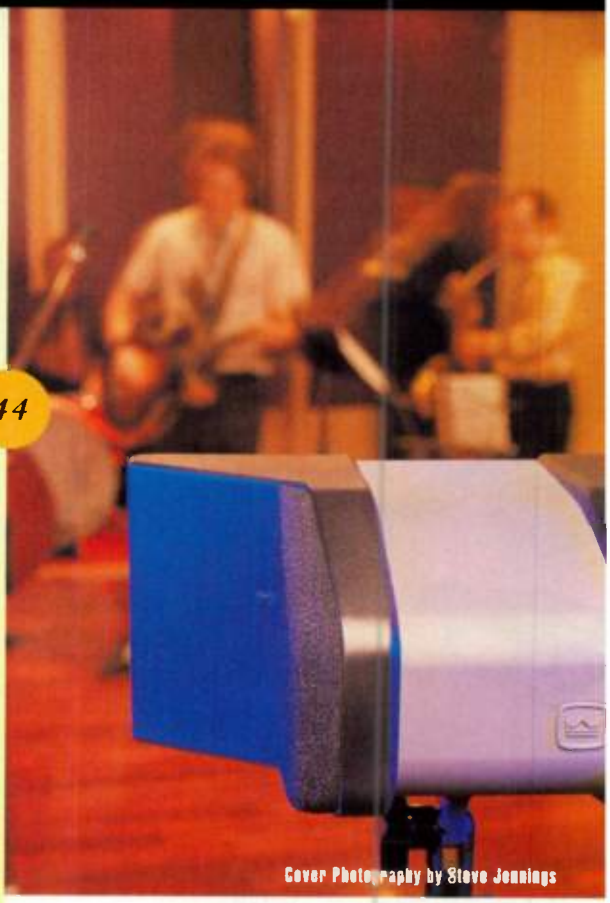

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Cover Photography by Steve Jennings

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DEPARTMENTS

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COLUMNS

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REVIEWS

We Play Requests

ANTHONY PIDGEON

This month we're delighted to honor one of the most frequent requests we've received in the past couple of years: a comparison of selected software synthesizers with the vintage hardware instruments they emulate. I, too, have long been interested in this subject, but we bided our time before jumping in.

Why did we wait? Because we wanted to have just the right combination of interesting software; access to well-maintained, classic hardware; the correct environment for meaningful tests; and the right author. Thanks to author Brian Smithers and to owner of the Audio Playground Synthesizer Museum, Joseph Rivers—and, of course, thanks to some amazing software-development efforts—the pieces fell in place in recent months, and we were able to research the story the way we wanted to (see "Is It Real or Is It Emulated" on p. 35).

Admittedly, comparing hardware with software emulations has its limitations. For starters, no two vintage analog instruments sound exactly alike. Sure, in a broad sense, a given model has its own identifiable sound. But put two Hammond B-3s or two Sequential Prophet-5 Rev 3s side by side, and you can usually hear subtle—and sometimes not-so-subtle—differences. An analog instrument's sound can be affected by many variables, some of which are difficult to identify, such as variations in a trace on a circuit board, components with the same value that don't operate exactly to spec, or a unit that is unusually temperature-sensitive. Furthermore, an analog instrument's sound depends on how you program it, or in the case of electromechanical instruments, how you adjust the mechanical parts.

Before moving forward with the story, therefore, we consulted with several experts in vintage electronic instruments, including long-time service technician and former **EM** "Service Clinic" column author Alan Gary Campbell, and Dave Smith, cofounder of Sequential Circuits (of Prophet-5 fame) and most recently the designer of the Evolver synth (reviewed on p. 110 of this issue). As Smithers writes, "ultimately, we realized that accounting for all possible variables was simply impossible; we needed to get over ourselves and be practical." He's right, of course, but with the understanding that we would attempt to account for variables that were at least partly within our control.

We also applied the Steve O Rat's Ass test. (I explained this test in my May 2002 column "Contemplating a Rat's Rear.") In this case, we asked if, assuming the software versions sound great and come quite close to the sound of the hardware, who gives a rat's ass whether they sound identical to the hardware? My guess is that few projects require perfect emulations; in most cases, a great soft synth that captures the essence of the hardware's sound is sufficient, if not more than sufficient. As you will see, the soft synths we tested certainly achieve that, and many offer greater flexibility, if not features, than the original hardware.

Thanks to all of you who wrote us requesting this story and others like it; I am confident that you will enjoy the results.

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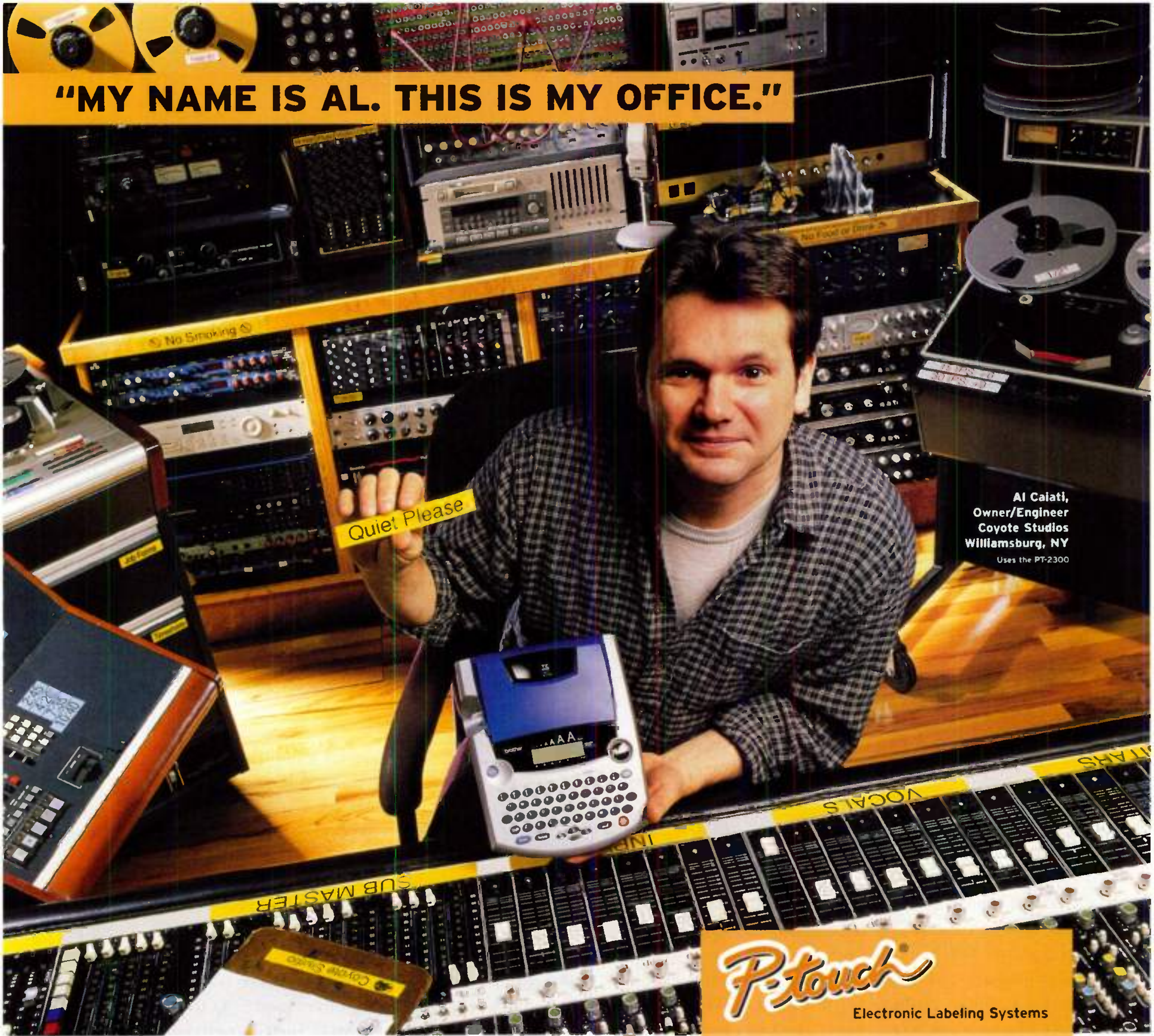

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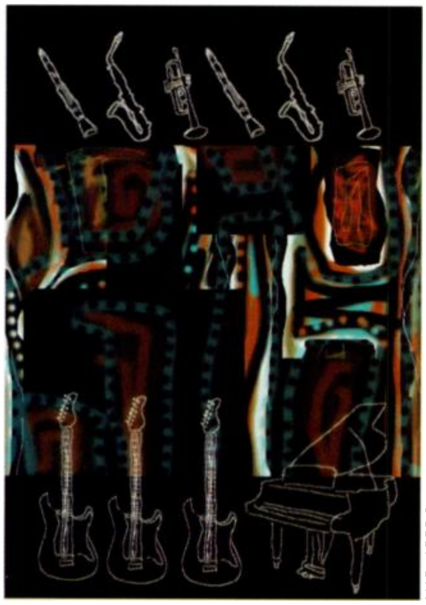


MAE LAROBIS

WILD THINGS

I just finished reading the article "Going Wild" by Gino Robair in the April 2003 issue of **EM**. Mr. Robair did a fine job of covering the ins and outs of field recording, which is very underrated. Reading his article opened my eyes, and I am completely fascinated. I have a new appreciation for people like Bernie Krause and Ruth Happel.

About one month ago, Los Angeles was subject to some pretty ugly rainstorms, so I decided to set up a pair of microphones and record some of that ugliness. The results were anything but ugly. The rain was very pleasing and interesting to listen to (except when an airplane flew over about five minutes into the recording).

Steve Oppenheimer sums it up best in his April "First Take" column when he says, "Sometimes we just need to get away." I applaud **EM**, Mr. Robair, and Bernie Krause for providing a means of escape from such horrible and stressful times in our world. What could be better then to escape through recording?

Charles Eck
via e-mail

GOTCHA

Even though it was coincidentally the morning of April 1st when I sat down to look through **EM**'s April 2003 issue, I admit that I blithely read through the complete "Tech Page: Fuzzy MIDI" column, nodding my head in general and naive agreement. It was only as I was about to turn the page that I pronounced General Mai Dai's name out loud and realized that I had been snookered. Good work on a joke well done.

I appreciate your publication's lightheartedness. I also reference Mr. Oppenheimer's "bluesy" editorial ("First Take: One More Thing to Sing the Blues About") about keeping a healthy perspective. Thanks for the fresh (and "fuzzy") air.

Mark Adamczyk
via e-mail

LOOPHOLES

Todd Souvignier's article "Closing the Loop" in the April 2003 issue is quite good. He clearly states its purpose in the opening paragraphs, and I have no quarrel with the article. However, the article and the excellent products he discusses offer a very narrow and limited view of looping in musical composition.

Looping began almost as soon as tape recorders were used for music. Listen, for example, to the loop studies from the 1950s on the Folkways recordings, *Sounds of New Music*. In those days, a loop was literally that: a loop of analog tape. Souvignier refers in several places to shooting for the goal of "one note per slice." That is accurate, and it is a good point in the context of the kind of music and products that he discusses. However, confining loops to transient sounds of short duration artificially limits the concept of looping. For example, long loops of sounds lasting many seconds (even minutes) that have a Gaussian profile can be layered to create evolving compositions.

Souvignier's article is excellent for its stated purpose, but I hope that **EM** readers will understand that there is more to looping in musical composition than simply using loop sequencers such as Ableton Live, Propellerhead ReCycle, and Sonic Foundry Acid while making assumptions about notes and measures.

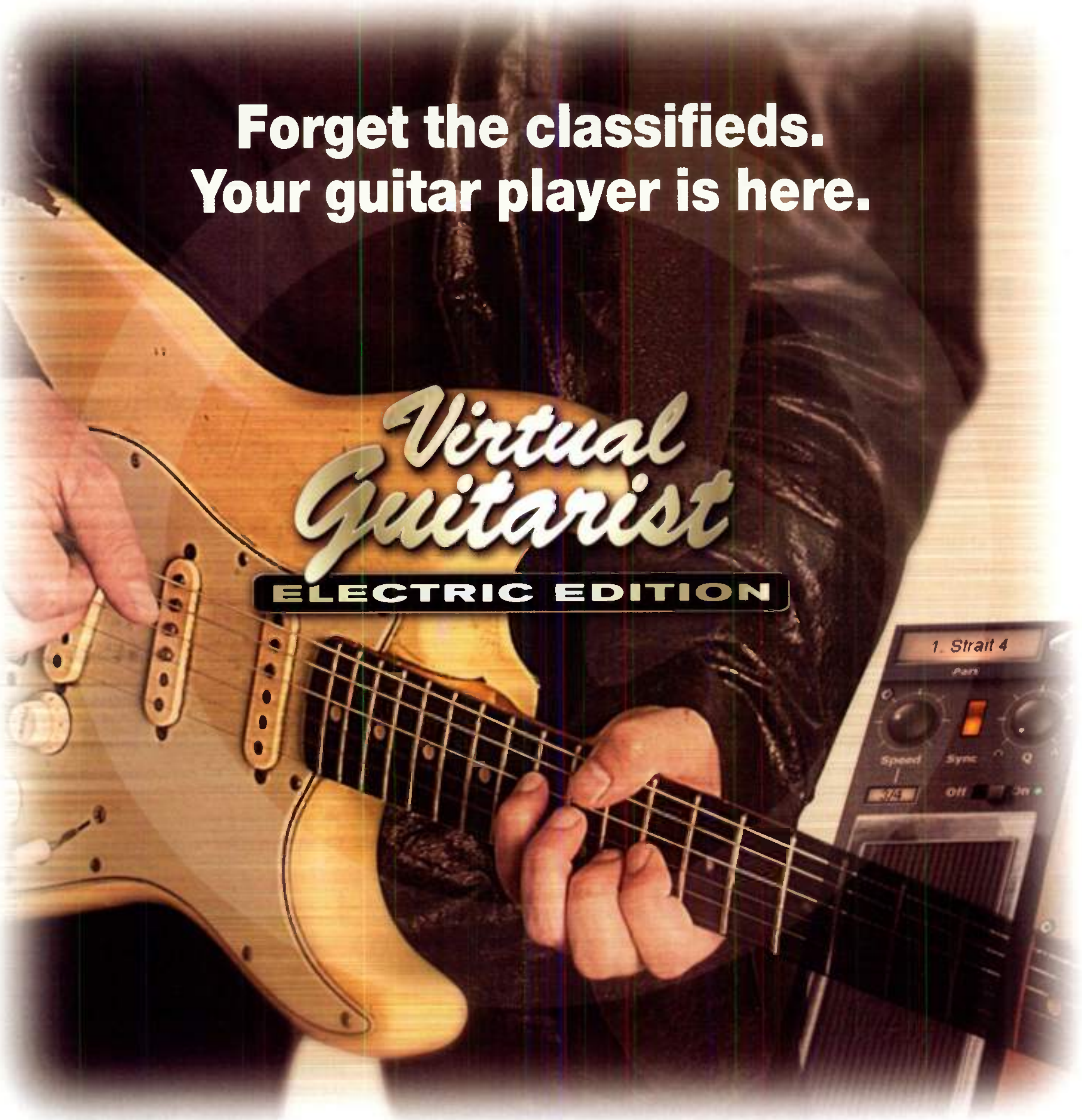
David Mooney
via e-mail

BRASS TACKS

I thoroughly enjoyed Rob Shrock's feature about brass libraries ("In Search of . . . The Ultimate Brass Library," March 2003). I have never gone wrong following his soundware recommendations.

However, Shrock overlooked one of the best brass collections, which has been out for some time: Tascam's *Bigga Orchestral Brass* for GigaStudio. It's the only currently available library (that I am aware of) that incorporates separate instrument samples for, say, First and Second Trombone, or First and Second Trumpet. It's also the most reasonably priced collection. Some of the other libraries that Shrock mentioned, such as EastWest's *Quantum Leap Brass*, are getting a bit "long in the tooth" and seem overpriced compared with the offerings from Tascam and Project SAM. I have steered clear of these older collections for that very reason.

I am also a little surprised that there is no mention in the article of the "lite" versions of the Dan Dean brass libraries. Those are also a fine value. Another curious omission was Best Service's Xsample sound library, disc 3, which sounds very good—better than



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Listen at www.steinbergusa.net - US: 818.678.5100 - Canada: 416.789.7100 some of the libraries that Shrock discussed enthusiastically.

All in all, it was a very good feature that could have been truly useful if the above omissions had been included.

Philip N. Chance
via e-mail

Author Rob Shrock replies: Philip—I partially dropped the ball on this one. Bigga Orchestral Brass *was submitted for consideration, but I was not initially impressed with the collection. When I narrowed my list of products down to the "best-of" stack, it didn't make the cut.*

I felt that its instrument sounds did not include a sufficient number of articulations for a collection of this size—four discs covering solo trumpet, trombone, French horn, and tuba in Giga format. The library consists primarily of three dynamic levels of sustains and staccato samples. Some nice specialty articulations like mutes, flutter tongues, and crescendos are provided, but I felt that the basic sustain sounds weren't flexible enough and offered little in the way of creative controller programming other than several keyswitched programs.

More importantly, I wouldn't include this library on my "must-have" list, which is an important consideration for EM*'s highly subjective "In Search Of" series of articles. I probably would not use the* Bigga Orchestral Brass *samples on one of my own real-world, commercial projects simply because I prefer other libraries.*

However, I must admit to some oversight on my part. During the review process, I usually go back at a later time and load up some of the libraries that almost made the cut so I can reevaluate them after I've lived with the other libraries for awhile. By doing so, I often gain a new perspective that changes my opinion of a library. In this case, I neglected to revisit Bigga Orchestral Brass, *and consequently, I did not mention it in the article.*

I stand by my opinions, but in hindsight, Bigga Orchestral Brass *deserved a mention specifically for the reason that you point out: the library is noteworthy for offering separate First and Second instruments to address the doubling issues that I described in the sidebar to the article. The library's doubling capability should certainly be applauded, and I agree that it should have been mentioned.*

While writing the article, I was unaware of the "lite" version of Dan Dean's Brass Ensembles *library. Perhaps it was not yet shipping at that time. Best Service's* Xsample, *volume 3, was not submitted for evaluation, although the distributor did provide other titles.*

GOOD REFERENCES?

I enjoyed Larry the O's review of five midpriced, powered studio reference monitors ("Sound Judgment," February 2003). I own Dynaudio Acoustics BM6As. In the article, you said that the vocals sounded further back in the BM6A mix than in some of the other mixes. I wonder if you found that to be true when you played the mixes on other systems? If so, how much of an adjustment did you make? I haven't done any final mixes on these monitors yet, but I like them so far.

Keep up the good work. I love your "Final Mix" columns.

Allen Kaye
Fresh Tracks Studio

Contributing editor Larry the O replies: Allen—Thanks for the kind words about "Final Mix." I assume that you are asking whether the mix that I did using the BM6As and subsequently checked on other systems had vocals sitting back in the mix. The answer is no, just the opposite. I made no accommodations for the character of the monitor; I made it sound as good as I could.

An excess or deficiency of some sort in a monitor often results in a mix that exhibits the opposite characteristic because during the mix, the mixing engineer compensates according to what he or she is hearing. I did not listen to each mix on each one of the monitors that I tested; rather, I listened more often on speakers with which I was familiar. The vocals were somewhat more out front in the BM6A mix compared with most of the other systems because I had brought them up while mixing until they sounded loud enough.

The BM6As' treatment of the vocals created an audible problem that wasn't pronounced. Note that the tendency for the vocals to sit back in the BM6As' mix was more noticeable because I was comparing the BM6As with a number of other systems.

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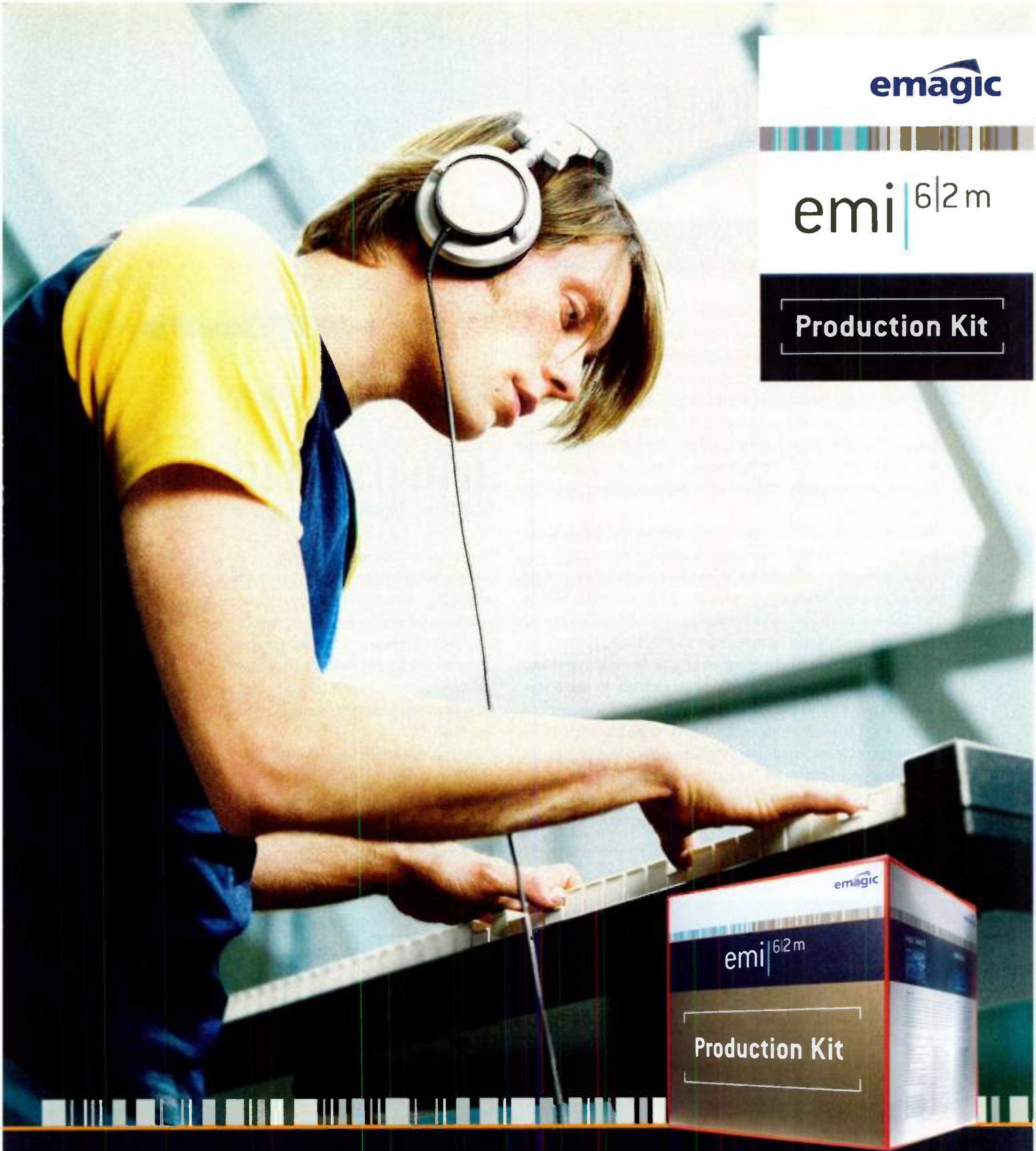

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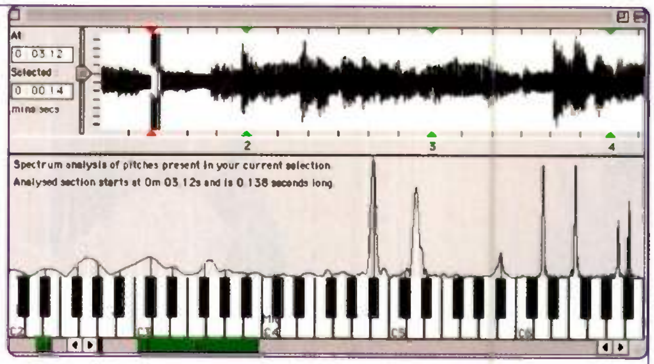
Download of the Month

By Len Sasso

If you subscribe to the time-honored tradition of learning a piece of music by ear but you don't have perfect pitch, Transcribe (Mac/Win shareware, $40) from Seventh String Software can give you a real leg up on the process. Transcribe performs three basic functions: it displays a spectral analysis of a selected bit of audio—a chord, for example—overlaid on a piano keyboard; it provides DSP functions for picking an individual instrument out of a mix; and it offers independent time-stretching and pitch-shifting functions. Transcribe's purpose is to help you transcribe or learn from an audio clip by playing along and deconstructing chords.

You start the process by loading a sound file; the program supports WAV, AIFF, and MP3 formats. Once you've opened the file, you can change its speed, place markers in it, and transpose it as you would in any sample editor. The recommended procedure is to place section, measure, and beat markers in the portion of the sound file that you want to analyze. Although markers have no effect in Transcribe's performance, they help you find your way around the file. As the file is looping, you can quickly tap them in on the computer's S, M, and B keys.

Transcribe offers two tools for improving its spectral analysis: control of the mix (Transcribe always analyzes a mono mix of the sound file) and equalization. In particular, the Stereo Mix dialog lets you balance the two sides of a stereo file and invert or delay one channel. You can often use those controls to focus in on the instrument you're trying to analyze. For example, if a vocal is mixed in the center of a stereo file, you can reduce or eliminate it by inverting one channel.



If you've used pitch-recognition software, you know that pitch analysis is a difficult technical task, especially when dealing with polyphonic material. With Transcribe, you still need to use your ears and brain, but its visual and DSP aids make the task much easier. Transcribe doesn't generate MIDI or score files, but it is a great aid in scoring by hand or with your favorite notation software. If you like to learn by playing along with audio examples, Transcribe is a must-have download.

For jazz-oriented musicians, the Seventh String Web site (www.seventhstring.demon.co.uk) also offers a free, searchable index of more than 40 jazz Fake Books as well as an interesting selection of big-band charts.

Key Changes

By Marty Cutler

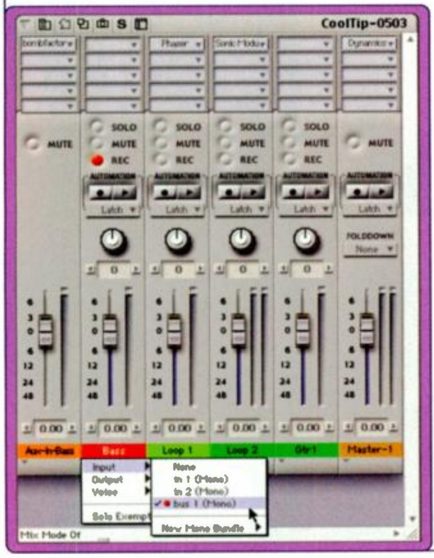
VirSyn (www.virsyn.com) is shipping the Mac OS X and Audio Unit versions of its TERA software synthesizer. Registered users can download the update from the VirSyn Web site . . . In order to entice customers to check out its VST effects and plug-ins, new company **ConcreteFX** (www.concretefx.com) offers DebaserFree. The plug-in is a pared-down version of Debaser, a multi-effects processor . . . **Cakewalk** (www.cakewalk.com) has announced that its Sonar digital audio sequencer will support Digidesign hardware with Digidesign's new ASIO drivers for Windows XP. Registered owners of Sonar version 2 software can get the new drivers at Cakewalk's Web site . . . **M-Audio** (www.m-audio.com) is bundling Live Delta with its Delta-series digital audio cards. Live Delta is Ableton's entry-level version of its Live loop-sequencing program. You can upgrade Live Delta to the full version of Live 2 to gain Live's more advanced features, such as MIDI sync and ReWire support . . . **Sonic Network** (www.sonicimplants.com) now offers *Symphonic Strings Mini*, an abbreviated, more affordable version of its *Symphonic String Collection*. The *Mini* version comes on six CD-ROMs and is available in Giga and SoundFont formats . . . The **Clavia** (www.clavia.se) Nord Electro 2 now ships with a close-miked, stereo-sampled, 9-foot Malmsjö acoustic grand piano. The new samples are also available as a free upgrade to existing Nord Electro owners. According to Clavia, the new sampled piano offers a "vast amount of Velocity levels." You can download the new piano at Clavia's Web site or request a CD-R containing the upgrade . . . **Wave Arts** (www.wavearts.com) has announced MAS support for its line of plug-ins. New MAS versions include the company's MasterVerb, TrackPlug, and WaveSurround plug-ins.

Cool Tip of the Month

The EM Cool Tip of the Month is presented courtesy of Cool Breeze Systems.

Tracking with Plug-Ins

You need to lay down a bass part, and you would like to use a plug-in to compress the track. Do you realize that when you record a track in most digital audio workstation software, the plug-in inserts are post-hard disk? For example, if you create a track in MOTU Digital Performer 3 (DP3) and insert a compressor, you are not compressing to hard disk when you record. When you listen back, the unaffected bass part streams from the hard drive and through the plug-in on its way to your audio interface.



If you're familiar with analog mixing consoles, it's like inserting a compressor in the monitor section versus the input section. The advantage is that you can edit the plug-in parameters later. A work-around is available, however, and with a sufficiently powerful computer and the right settings, latency isn't an issue.

1. First, in order to minimize monitor latency, adjust the Samples Per Buffer in DP3's Configure Hardware Drivers window from the default of 1,024 to 256 (or lower if possible). The lowest value that you can set will depend on what your hardware configuration is.

2. Next, open the Mixing Board from the Windows menu (Shift + M). Add an aux track (Control + Command + A), an audio track (Shift + Command + A), and a master track (Control + Command + M) from the Mixing Board's mini-menu.

3. Insert a compressor plug-in on the aux track. Configure its input to the hardware input that your bass is plugged into, and set the output to bus 1.

4. On the audio track, set the input to bus 1 and the output to out 1–2, and then arm the track.

5. Adjust the compressor to your taste and record a bass part.

In this configuration, the bass part is first being compressed and then routed to the mono track and recorded. Some lesser systems may experience latency, but with newer, faster computers, that will be less of an issue.

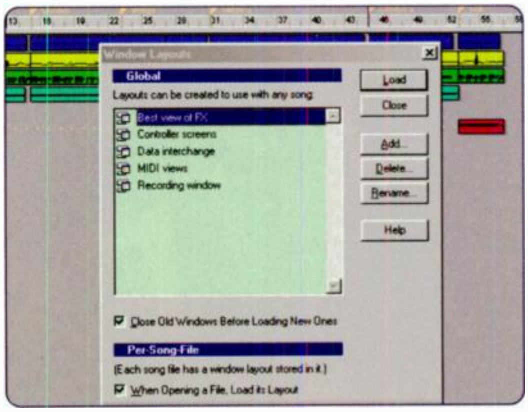
—Steve Albanese

Be sure to check out the streaming movie tutorial of this procedure. Log on to www.emusician.com/cooltip to take part in this online adventure. Also, if you dare, take the quiz to review what you've learned!

TIPS

Customizing the Sonar Interface

Creating custom Window Layouts is a great way to configure Sonar's user interface for specific tasks. For example, perhaps you prefer to see certain windows while recording but different windows while editing and arranging. Simply configure any views the way you want them, then go to View and choose Layouts, click on Add, and provide a descriptive name for the Layout. You can now access any saved Layout from the Window Layouts dialog. For even quicker access, you can assign Window Layouts to Key Bindings, which allows you to instantly launch a Window Layout with the press of a key. Go to Options and select Key Bindings, select a key combination, and then bind it to any of the saved Window Layouts.



You can also create a custom toolbar to select Window Layouts or any other command in Sonar. Begin by creating a new blank StudioWare panel, and then insert a button for each toolbar command you desire. To configure the buttons, enable Design mode and double-click on a button. Make sure Spring-loaded is checked. Next, set the Primary Action to None and set the Return Action to Binding. You can then choose from any of Sonar's available Key Bindings. If you want your custom toolbar always displayed on top of other windows, click on the upper left corner of the StudioWare view and select Enable Floating.

—Morton Saether, Cakewalk

Son of PRAM

Veteran Macintosh users have long been accustomed to occasionally resetting their computer's parameter RAM, otherwise known as zapping the PRAM. This is accomplished in Mac OS X just as it was in previous versions: by starting your Mac and then immediately holding down the Command, Option, P, and R keys until you hear the computer start a second time.

Resetting the PRAM is a first line of defense when your Mac exhibits unusual or eccentric behavior. PRAM (or, in recent PowerBooks, NVRAM) is similar to a PC's CMOS memory. It contains various system settings that your computer accesses during boot-up and are held in nonvolatile memory by battery backup. The stored data includes settings for

For more EM Cool Tips, visit our Web site at www.emusician.com.

FANTOM-S88

The New Fantom-S, with 16 Velocity-Sensitive Sample Pads.

The Roland Fantom-S Workstation, comes with 16 touch-sensitive pads and innovative audio features like new SkipBack Sampling and realtime timestretching at your fingertips, as a result, you can build rhythm tracks, trigger samples and patterns, and produce


PAD BANK
14
15
16
INS
DEL
BS
10
11
12
SPACE
6
7
8
PQR
STU
VWX
2
3
4
DEF
Some call it
the seamless integration
of audio and MIDI.
We call it the Fantom-S.

Rev Up

By Marty Cutler

Sound Quest

Version 9.0 of Midi Quest (Win, $199; upgrade from v. 8, $79; upgrade from v. 2 through 7, $99) offers a revamped editing engine and new user-interface features. The standard version of the editor-librarian supports 600 instruments from 26 different manufacturers.

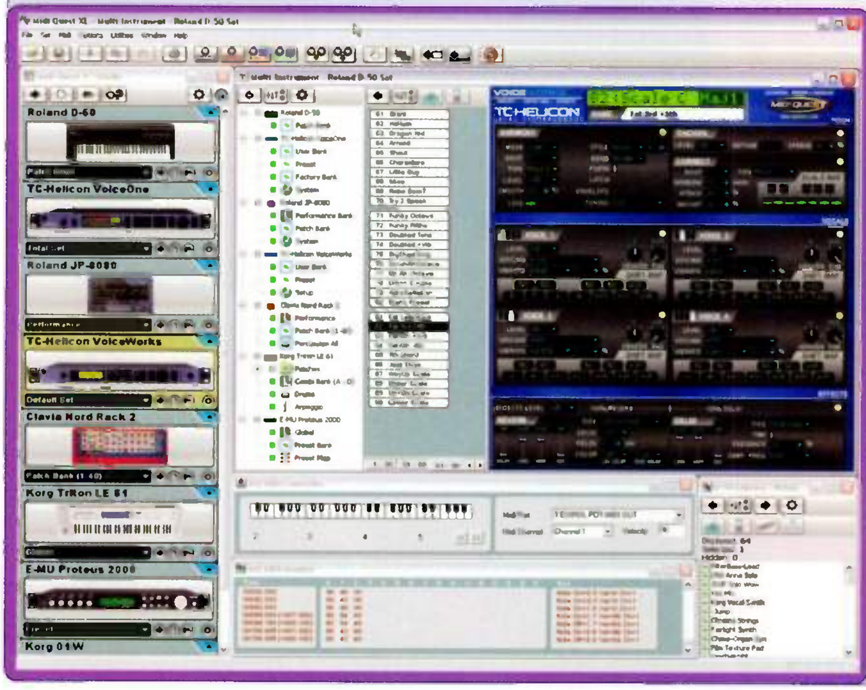
The Studio window now features a high-quality picture of each installed instrument; if that consumes too much screen space, the instrument display is collapsible. You can configure settings and load and send SysEx data by pressing a single button. You can also store, organize, and edit all the settings for each instrument in one Set window, which features a tree-structure list of all banks and patches, organized by instrument.

Among the patch-editing enhancements, the Smart Controller feature assigns continuous controllers to selected parameters, so you can edit patches from any MIDI keyboard or control surface. You can automatically select controls on mouse-over, and if you're using an Intellimouse wheel, you can select and edit a parameter without clicking.

Midi Quest can appear as a VST plug-in for supporting parameter automation, allowing you to edit, record, and play back all patch edits from any VST-compatible sequencer. MIDI Quest also supports the recently developed Open Plug-in Technology (OPT), which enables MIDI Quest to function as a plug-in within OPT-compatible sequencers, such as Cakewalk Sonar.

Sound Quest also offers MIDI Quest XL (Win, $299; upgrade, $149), which promises enhanced interactivity with a variety of sequencers. The program supports multiple patch-naming formats so that new sounds appear in third-party MIDI software. In addition, it automatically updates patch names when it sends a new bank to an instrument.

Midi Quest requires at least a 486 CPU with 8 MB of RAM and Windows 95, 98, 2000, ME, NT, or XP. Sound Quest Inc.; tel.



TIPS

your startup disk, time zone, speaker volume, DVD region, and so on. Zapping the PRAM returns those settings to their default values. Afterward, you may want to open your System Preferences to ensure that everything is copacetic. If you're having trouble with your network or monitor settings, however, zapping the PRAM no longer has any effect in OS X.

—*Geary Yelton*

Nonstandard MIDI Files

Standard MIDI Files (SMFs) are supposed to be universally readable, but in fact, they're not always 100 percent cross-platform compatible. Most of the MIDI files floating around the Internet were created on and formatted for PCs (typically with an MID extension) rather than Macs.

If you're using an earlier version of MOTU Digital Performer, you should be able to open an SMF by choosing Show All Readable Files from Performer's Open File dialog box. (The latest version automatically looks for all readable files.) If your target file doesn't show up, however, don't despair. MOTU provides a handy little utility called MIDIfile Converter. You'll find it in the Extras folder inside the main Performer folder.

Just double-click on the program icon and point MIDIfile Converter to the file of your choice. The program quickly and easily converts SMFs from PC to Mac format by changing the Type and Creator data so that Digital Performer (or other Mac programs) can properly recognize the file as an SMF. It should then show up as expected in the Open File dialog box.

—*David Rubin*

Hang On to Your Canvas

I have never sold my Roland SC-55 Sound Canvas, despite owning several other synthesizers with

(800) 667-3998 or (250) 478-4337; e-mail sales@squest.com; Web www.squest.com.

Anwida Soft

DX Modulation Pack 2.0 (Win, $99) is a new version of Anwida Soft's bundle of six DirectX plug-ins. The pack includes phaser, flanger, chorus, tremolo, delay, and modulated resonant lowpass and highpass filters. According to the manufacturer, the new version offers improved sound quality, more control parameters, and a revised graphical interface. You can now edit parameter values by typing them in, and you can freely assign continuous controllers to control parameters.

Many new presets are included, and the built-in Preset Manager helps organize them. You can even choose specific presets to load at startup. The upgrade also fixes a few bugs, including one that produced CPU spikes when the software was used with Cakewalk Sonar on a Pentium 4 computer.

DX Modulation Pack 2.0 requires a 200 MHz Pentium, 32 MB of RAM, a DirectX-compatible host, and Windows 95, 2000, ME, NT4, or XP. You will also need Microsoft's DirectX Media 5.2 or higher. Anwida Soft; e-mail info@anwida.com; Web www.anwida.com.

Yamaha

Firmware for Motif 6, 7, and 8 synthesizers is now at version 1.7. A free download is available at Yamaha's Web site; the new version includes file-management enhancements in addition to programming improvements.

In prior versions, you could load or save Performances only in groups of 128; now you can load individual Performances from the Motif's internal memory. When converting Pattern chains to a Song, an optional menu is available for copying Program Changes. You can also copy Pattern Mixing data. The Motif can now serve as a master keyboard for controlling up to four zones in the Song and Pattern modes (even when the tone-generator block is set to Multitimbral mode), which allows you to create live performance setups with external synths. The Motif's remote-control functions now include support for Cakewalk Sonar and the Yamaha SQ01 sequencer.

The Song/Pattern Mixing mode's INIT display offers a General MIDI mode. Initializing with the GM box checked on specific parts will assign those selections to the GM bank, which is convenient when you want to play a Standard MIDI File that doesn't contain a GM On message.

In addition, the new version provides preset plug-in Voices for the PLG150-DR and PLG150-PC, so you can play the various drum and percussion sounds as if they were part of the Motif. You can even use the sounds with the Motif's dynamic rhythm-related arpeggio settings. Yamaha Corporation of America; tel. (714) 522-9011; e-mail info@yamaha.com; Web www.yamaha.com or www.yamahasynth.com.

General MIDI (GM) sounds, superior samples, and greater polyphony. Because most commercial Standard MIDI Files use 24-note polyphony as a yardstick for the maximum polyphony of a GM sequence, the Sound Canvas is a useful tool when I need to deliver sequences in the GM format. If I start hearing notes drop out, I know that I've exceeded my polyphonic allowance.

I have also developed GM sound sets for manufacturers of other synths, and the SC-55 came in handy for comparisons. Because the instrument was probably the first to define the GM standard, it is useful for balancing relative patch volumes and replicating the original envelope generator settings when developing a GM sound set. Sadly, the SC-55 is no longer in production, but you can easily find one at your favorite Web auctioneer for a song.

—Marty Cutler

Drivers Education

If you are a Windows XP user with a Digidesign interface and you want to use the interface with software that requires ASIO drivers (such as Ableton Live or Propellerhead Reason), now you can with Digidesign ASIO Driver (Beta 5.3.3b2) (www.digidesign.com/download/asio). The company notes that if you plan to also use Pro Tools with your Digidesign interface, you must load it on your computer before loading Digidesign ASIO Driver.

Mac users who want to use a Digi 001 interface with programs requiring an ASIO driver can get one from the Steinberg Web site (http://service.steinberg.net/download.nsf/23d2391ee0a43f2fc12563af00270e95/e4692628073c4223c12568e900506482?OpenDocument).

—Gino Robair

WHAT'S NEW

By Marty Cutler

MBHO MBNM 440 CLS

MBHO offers the MBNM 440 CLS ($439), a small-diaphragm condenser mic for recording choir, percussion, and miscellaneous acoustic instruments. The mic's 3.35-inch body houses an integrated amplifier and the MBNM 440 cardioid capsule. This gives the mic an extremely short signal path and low phantom-power consumption, with the ability to accept voltages from 22V to 48V.

The MBNM 440 CLS delivers a frequency response of 40 Hz to 20 kHz with a signal-to-noise ratio of 66 dB. Maximum SPL is rated at 126 dB at 1 kΩ. The mic has a nonreflective black finish and features a –10 dB pad and a defeatable highpass filter. Matched pairs are available ($878) and include two free mic clips with metal threads. MTC (Music Trade Center) (distributor); tel. (718) 963-2777; e-mail info@mbho.de; Web www.mbho.de.

CUTTERMUSIC REVITAR

Revitar (Win, $49) from CutterMusic is a VST Instrument that uses physical modeling to generate a wide range of guitar sounds. Revitar models the position, velocity, and momentum of 120 points along each guitar string, at a resolution of 44.1 kHz, to reproduce guitar characteristics such as sympathetic resonance, slap effects, pluck position, and vibrato. The manufacturer claims that no two notes will sound exactly alike.

Revitar is four-note polyphonic and lets you play entire chords with a single keystroke. You can choose from seven types of chords as well as select the rate and direction of the strumming. For precise modeling of melodic playing, a mono switch allows you to play notes on the same string without getting overlapping decays.

All parameters accept MIDI continuous controllers, including adjustments for bass, treble, and sympathetic resonance. You can control note duration with modeled bridge and string damping and choose models of round- or flat-wound strings. Other controls let you adjust plucking position, guitar-pickup position, and fret height.

To use Revitar, you will need a Pentium III/700 MHz; Windows 95, 98, 2000, ME, NT, or XP; and 64 MB of RAM. CutterMusic; e-mail support@cuttermusic.com; Web www.cuttermusic.com.

ROLAND GI-20

Roland's latest pitch-to-MIDI converter for guitar is the GI-20 ($395). The GI-20 detects the signal from any electric guitar or bass equipped with a divided pickup, such as Roland's GK-2A or GK-2B ($275; $200 when purchased with the GI-20). The unit will also accept input from MIDI-ready guitars with a built-in 13-pin connector.

The GI-20 sports a built-in USB MIDI port in addition to the customary MIDI In and MIDI Out ports. The unit offers a ¼-inch analog output for accessing the guitar's audio signals. Other connectors include a 13-pin input jack and ¼-inch TRS jacks for a footswitch and an expression pedal.

The converter gives you 50 user locations to store settings and includes a Play Feel feature, which improves tracking by tailoring converter response to playing styles such as fingerpicking. Other programmable features include transposition and string-muting settings for each string. You can transpose MIDI output for each string in semitone increments. The GI-20 supports and stores Bank Select as well as Program Change messages for each patch. You can engage the global transpose functions using an optional footswitch.

Editing is simple with the GI-20's front-panel parameter-selection knob, increment and decrement buttons, Write button, and 2-character LCD. The converter also provides a multifunction tuner and string-select switch. Roland Corp. U.S.; tel. (323) 890-3700; Web www.rolandus.com.

STEINBERG GROOVE AGENT

From the programming team that developed Virtual Guitarist comes Groove Agent (Mac/Win, $249), a VST Instrument plug-in that lets you quickly create percussion tracks by matching musical styles with sampled drum kits. It simplifies the groove-creation process and provides instant inspiration for anyone who needs help sequencing realistic drum parts.

At the heart of Groove Agent's drum-machine-style user interface is a slider that lets you choose from 50 years of musical styles, including examples of Latin, soul, rock, and electronica beats. Another slider alters the level of musical complexity in real time. You can adjust individual instrument parameters such as volume, tuning, and Velocity response and then save snapshots for immediate recall. Instruments can be routed to any of four stereo outputs.

Groove Agent provides 24-bit samples, including four complete drum kits, electronic drums, and various percussion instruments. In addition to internal sounds, Groove Agent can play other virtual instruments or even external MIDI synths and samplers.

On the PC, Groove Agent requires a Pentium III/400 MHz and Windows 2000 or XP. On the Mac, you'll need a G3/366 MHz and Mac OS 9 or OS X 10.2. Both platforms need at least 256 MB of RAM, 300 MB of disk space, and a VST 2.0–compatible host. Steinberg North America; tel. (818) 678-5100; e-mail info@steinberg.net; Web www.us.steinberg.net.

BEHRINGER TUBE ULTRAGAIN MIC100

The Tube Ultragain Mic100 ($49.99) from Behringer is a tube preamp designed for studio, live, and hard-disk recording applications. The unit features a hand-selected 12AX7 vacuum tube combined with the company's proprietary UTC technology, which helps distribute tube-warmth characteristics at all volume levels. Behringer says the Tube Ultragain Mic100 has low-noise and low-distortion properties and lends punch to percussion instruments and signals that are rich in upper harmonics. The device can also be used as a DI box.

The Tube Ultragain Mic100's microphone preamp circuitry uses discrete components that the company claims helps to produce a highly transparent sound. The preamp offers a built-in limiter, a phase-reverse button, defeatable 48V phantom power, a switchable 20 dB pad, and an LED meter. The preamp includes two top-panel knobs: one boosts the input from +26 dB to +60 dB, and the other adjusts the output from unity gain to +10 dB. The rear panel offers balanced ¼-inch and XLR inputs and outputs. The device is powered by an external supply. Behringer USA; tel. (425) 672-0816; e-mail sales@behringer.com; Web www.behringer.com.

VIRSYN CUBE

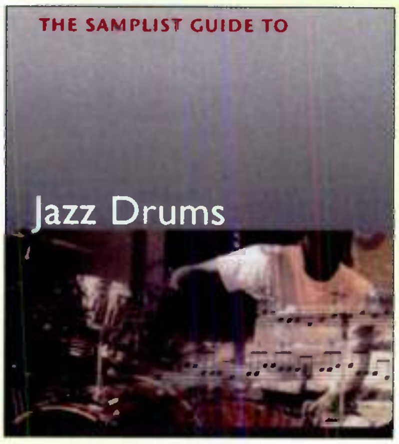
VirSyn's Cube (Mac/Win, $249) is a software additive synthesizer that operates as a standalone synth or as a VSTi, DirectX, and Audio Unit plug-in. Cube lets you combine harmonic and inharmonic spectra, and it delivers eight multitimbral parts, 512-note polyphony, and 512 partials per voice. Patches consist of up to four additive sources, and you can add so-called Arbitrary noise modulation to each partial.

Cube offers eight independent outputs and eight virtual racks of effects, including reverb, modulation-type effects, distortion, echo, and delay. The synth sports three 64-stage envelope generators that can sync to tempo. Likewise, the instrument's programmable arpeggiator and two LFOs can also lock to bpm.

Real-time adjustments with MIDI continuous controllers enable you to morph between sources as well as banks of filters with arbitrary shapes. A Learn function lets you assign your choice of continuous controllers to desired parameters. Macro parameters, which allow you to quickly change the parameters of all partials, include Brightness; Partial Stretch, which incrementally detunes the frequency ratios between partials; and Ensemble, which offers a natural, detuned chorusing effect between sources. All parameters of Cube can be adjusted in real time.

PC users will need a Pentium III/600 MHz, 128 MB of RAM, and Windows 98, 2000, ME, or XP. Mac users will need a G4/400 MHz, 128 MB of RAM, and Mac OS 9 or OS X 10.2. VirSyn Software Synthesizer; tel. 49-7240-202-956; e-mail info@virsyn.com; Web www.virsyn.com.

SOUND ADVICE



POWERFX

Swedish soundware developer PowerFX has introduced the first two CDs in its *Samplist Guide* series. *The Samplist Guide to Jazz Drums* (WAV, $49.95) features performances by respected drummer Ralph Peterson. Encompassing jazz from its New Orleans roots to recent styles, the disc provides drum and cymbal hits, patterns, rolls, and fills as well as mallet and brush patterns. As a bonus, Windows Media clips show Peterson demonstrating the various styles and discussing their history.

Fans of retro gear will want to check out *The Samplist Guide to Vintage Drum Machines* (WAV, $49.95). The disc contains single hits and loops from 13 analog and digital drum machines such as the LinnDrum LM-1, E-mu SP-12, and Roland TR- and CR-series. It includes samples from what might be the first drum machine in existence—the Wurlitzer Sideman, circa 1957. The disc also supplies Windows Media clips of each of the featured drum machines. PowerFX/EastWest (distributor); tel. (800) 833-8339 or (310) 271-6969; e-mail sales@eastwestsounds.com; Web www.soundsonline.com.

SCARBEE

Scarbee's *RSP 73* (Giga, $199) is a two-CD set dedicated to the Fender Rhodes Stage Piano MK I. *RSP 73* takes advantage of Tascam GigaSampler's streaming capabilities by delivering unlooped, naturally decaying samples.

The instrument was recorded at 24-bit resolution without effects, saturation, or compression. The engineers sampled directly from the instrument's harp, thereby eliminating the front-panel passive tone control to avoid excess noise in the recording.

The 73 keys were sampled at 12 Velocities, each with its own release sample. The release sample captures the sound of the damper bouncing against the tine as the key is released, which is an artifact of the original instrument.

The package includes reduced versions of the instrument, consisting of sample maps of the white keys that

stretch to cover the flat and sharp keys. These versions provide a choice of 4, 8, and 12 Velocity-layer instruments. You also get a filtered version of the piano to accommodate preferences for a more muted sound. Scarbee/EastWest (distributor); tel. (800) 969-9449 or (310) 271-6969; e-mail sales@eastwestsounds.com; Web www.soundsonline.com.

PG MUSIC

World Fretboards (Mac/Win, $29) is a set of Band-in-a-Box styles that covers a wide variety of genres, including new age, Hawaiian, pop, jazz-pop, and ethnic. Many of these styles utilize the Intelligent Chord Fretboard features in version 12 of Band-in-a-Box, which can illustrate correct chord voicings for ukulele, banjo, mandolin, and bass.

Alternative Contemporary Rock (Mac/Win, $29) features 20 new styles, including alternative, contemporary rock, pop, hip-hop, and jazz-inflected trip hop. *Alternative Contemporary Rock* requires Band-in-a-Box version 10 (Mac/Win) to play back these styles. PG Music; tel. (250) 475-2874; e-mail info@pgmusic.com; Web www.pgmusic.com.

BIG FISH AUDIO

Phat Beats From the Box (audio/Acidized WAV, $99.95) is a sample collection made entirely from the human voice. The two-disc collection taps some of New York City's hottest human beatbox talent, including the renowned Kenny Muhammad (aka the Human Orchestra), Omri Anghel (aka the Human Beat), Baba, and Hesher. *Phat Beats From the Box* offers full mixes, individual hits, and breakdowns. Individual instruments include mouth-made bass drums, snares, cymbals, and effects. The tempos range from 75 bpm to 147 bpm. Disc one is a 49-track audio CD. Disc two offers the same material on disc one as 24-bit, 44.1 kHz Acidized WAV files, and adds an extra folder called Single Sounds Bank. Big Fish Audio; tel. (800) 717-FISH or (818) 768-6115; e-mail info@bigfishaudio.com; Web www.bigfishaudio.com.

VIENNA SYMPHONIC LIBRARY

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The Performance Set 110,000 **Performance Elements** on DVD-ROM, innovatively mapped and enhanced by the revolutionary **Performance Tool**, a suite of MIDI utilities that provide incredible authenticity and ease of use. With the aid of this custom software, developed exclusively by the Vienna Symphonic Library, you can create fluid, realistically articulated passages like legato, portamento or glissando. Imagine the authenticity of pre-recorded phrases with the flexibility of multi-samples! An orchestral sampling breakthrough! **48 GB, $1,490**

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EQ Blue Ribbon winner
Musikmesse International Press Award
Game Audio Network Award

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The Demo Cube with over 650 MB of orchestral sample data, MIDI templates, tutorials and audio demos is available for just $10 directly from ILIO. Call 800-747-4546 or visit www.ilio.com.

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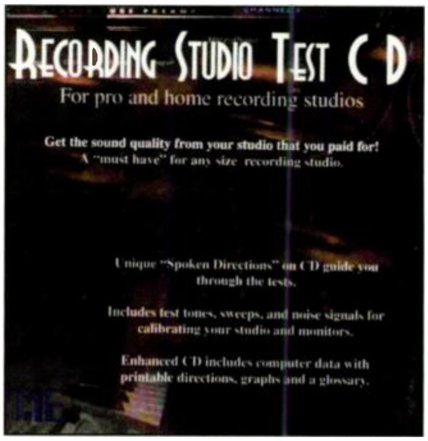
TME RECORDING STUDIO TEST CD

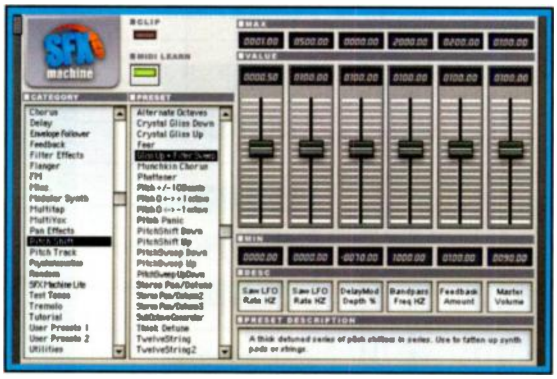
TME has introduced its *Recording Studio Test CD* ($14.95), an enhanced compact disc that features a variety of test tones for fine-tuning the gear in your studio. The disc plays on an ordinary CD player and includes step-by-step directions that walk you through the tests for calibrating and setting up your system. The data portion of the disc contains data files with printable instructions, graphs, and a glossary of audio terms.

Most of the tests require simple adjustments. The first 19 audio tracks introduce and describe how to perform the tests. The remaining 60 tracks offer reference tones for identifying speaker channels, testing speaker phase, calibrating VU meters, testing signal-to-noise ratios, checking for dropouts in DAWs, tuning monitors to the room, locating rattles, and checking frequency response. The disc can also be used to check the performance of outboard gear by testing compressor ratios and gate and limiter thresholds, and to plot EQ band response. You can also use the CD to set up P.A. systems as well as home and car stereo systems. TME; Web www.tmenet.com.



THE SOUND GUY SFX MACHINE RT

SFX Machine RT (Mac/Win, $150) is an updated real-time version of the popular SFX Machine program, an **EM** Editors' Choice Award winner in 1998. The software has been retooled as a plug-in for Mac OS X Audio Unit and Mac and Windows VST hosts. The plug-in includes hundreds of presets ranging from standard effects such as chorus, flange, echo, tremolo, vibrato, AM, FM, and triggered wah to unusual effects such as Sitar Drone, Pitch-Tracked Pan, and MultiVox. You can modify the presets to create your own. SFX Machine RT retains the simple user interface of the older version, with effects categories and presets in left-hand windows and processor controls on the right. SFX Machine RT features a MIDI Learn function, which lets you automate presets from an external controller.



The minimum system requirements for SFX Machine RT are Mac OS 9.2 or OS X 10.2, a Mac G3, and 5 MB of RAM above the host's requirements. Windows users will need a Pentium III, Windows 95, and 5 MB of RAM beyond the host's requirements. The program comes bundled with the non-real-time Premiere-format plug-in, version 1.56. The Sound Guy; e-mail sfx@sfxmachine.com; Web www.sfxmachine.com.

M-AUDIO OZONE

M-Audio's Ozone ($399.99) combines a MIDI keyboard controller, mic preamp, and USB audio and MIDI interface. The unit occupies a footprint no bigger than a laptop computer and can fit into a single 19-inch rackspace.

Like M-Audio's Oxygen8, the Ozone offers a 25-note, full-size keyboard, along with pitch-bend and modulation wheels. It also sports a Data-Entry slider and eight assignable MIDI control knobs. The MIDI/Select button lets you assign the Data-Entry slider by pressing notes on the keyboard that are highlighted with target parameter names, or you can select controls using keys flagged for numerical data entry. With its USB MIDI connection and two MIDI Out ports, the Ozone serves as a 16-channel MIDI interface: one Out port passes data from the computer, while the other sends the Ozone's direct MIDI output.

The USB port features the same 24-bit, 96 kHz audio section found in M-Audio's USB Duo, Quattro, and Audiophile interfaces. The audio inputs include an XLR microphone jack with a built-in preamp and phantom power, a balanced ¼-inch TRS input for guitar and bass, and unbalanced ¼-inch stereo aux inputs for line-level sound sources. The audio input can be routed to the computer or directly to the Ozone's audio outputs. M-Audio claims that the Ozone provides zero-latency direct monitoring of its inputs. The Ozone also includes unbalanced ¼-inch analog outputs as well as a ¼-inch stereo headphone jack.

M-Audio offers the optional Oxygen Tank ($39.99), a padded nylon backpack specifically designed to carry an Ozone, Oxygen8, or laptop computer; and the Studio Pack ($89.99), which can tote an Ozone or Oxygen8, a laptop computer, and accessories. M-Audio; tel. (800) 969-6434 or (626) 445-2842; e-mail info@m-audio.com; Web www.m-audio.com.

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PRO FILE

By Matt Gallagher

New Moon

Guitarist Michael Goldwasser (aka Michael G) is the musical core of Easy Star Records, an indie record label in New York City specializing in reggae. Goldwasser produces, arranges, and writes music for Easy Star's artists. He also serves as a recording engineer and is the label's principal session musician.

Dub Side of the Moon is Easy Star's most recent release and the brainchild of label vice president Lem Oppenheimer, a Pink Floyd fan as a teenager who later immersed himself in reggae. In 1999 Oppenheimer re-imagined *Dark Side of the Moon* as a reggae album and persuaded his label partners to develop his vision. "I enlisted Ticklah [longtime musical collaborator Victor Axelrod] to coproduce," Goldwasser says. "We worked out some bare-bones arrangements on sequencers to feel it out."

Easy Star creates a reggae version of Pink Floyd's classic album.

The producers decided that *Dub Side* should line up with *Dark Side* bar for bar. "We had to write reggae arrangements for all of the songs while figuring out how to keep the flow of the original [album]," says Goldwasser. "Reggae only works at certain tempos. We tried to be true to the original tempos, but on some songs, we had to change them." Goldwasser and Axelrod recruited drummer Patrick Dougher and bassist Victor Rice to complete their rhythm section. They also sought out guest musicians from New York's reggae and drum 'n' bass scenes.

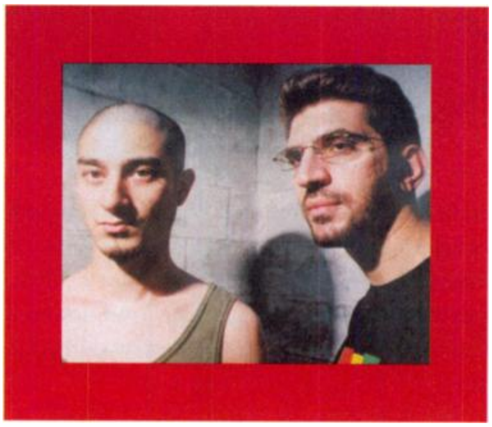
Dub Side has an analog foundation. The All-Stars recorded basic rhythm tracks without EQ or compression to 1-inch tape on a 16-track Tascam 8516 in a basement studio. They later added horns, vocals, percussion, guitars, synths, and pianos. "Most overdubs were done at Ticklah's studio," Goldwasser says. "We transferred all the 16-track stuff to ADAT. We were going for an analog sound using analog and digital equipment," he says. Their outboard equipment included an Aphex Model 109, a Lexicon MPX 100, a Line 6 Echo Pro, and a homemade tube compressor. Goldwasser feels that a Roland Space Echo produces delay effects best suited for reggae. "That's what was used back in the '70s," Goldwasser says. "That's how the dub sound came to be. The final tracks were noisier than we wanted, but it's a trade-off we had to make."

The tracks "Money" and "Us and Them" feature sequenced drum parts that feel live. "Due to technical problems, we had to re-create some drum parts that we recorded live by sampling our own sounds [with an E-mu ESI 4000] and sequencing them on an old version of [Steinberg] Cubase Audio XT," Goldwasser says. "The original drums were recorded onto 1-inch tape, so we were just sampling those."

Out of necessity, "the entire thing was mixed in a little less than a month," Goldwasser says. "We mixed to DAT and to a 2-track, ¼-inch machine: an Otari MX-5050." They decided to master from the analog-tape mix. "The DAT version sounded bright and clean, but the tape version sounded so much warmer, like the reggae that I love."

The mixing process was critical to ensuring the success of *Dub Side*'s mission. "There are certain standards we had to meet in terms of making the arrangements work," Goldwasser says. To preserve the album's continuity, he says, "we had to think things through before we did anything. We messed up a few times, and it was a good example of how you have to be prepared before doing a session."

For more info, contact Easy Star Records; P.O. Box 602, Midtown Station, New York, New York, 10018; tel. (212) 736-2160; e-mail easystar@easystar.com; Web www.easystar.com.



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***Dub Side of the Moon*/Easy Star All-Stars**

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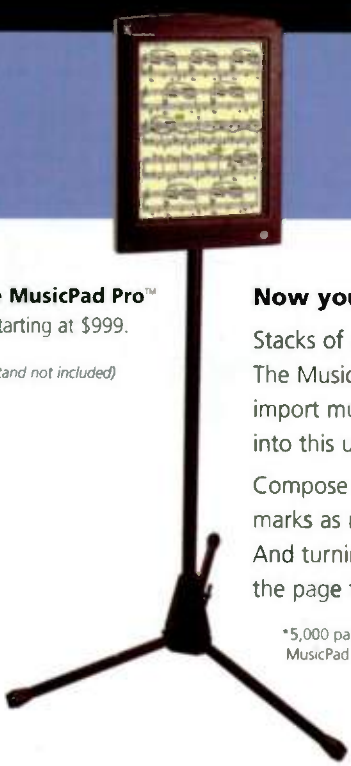

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Is It Real or Is It Emulated?

Photo by Brian Smithers

Will the real instruments please stand up?

By Brian Smithers

It isn't widely known, but I invented virtual instruments. I don't like to brag about it, but I remember the moment of inspiration well. It was winter of 1986 at the Tralfamadore Café in Buffalo, New York, as my bandmates and I sweated, strained, and cursed our way up the back stairway, scraping the walls, bruising our knuckles, and laboring under the weight of a Hammond B-3. Ninety-nine percent perspiration, indeed.

Well, perhaps "invented" is a bit of an exaggeration, especially because my Texas Instruments TI-99/4A wasn't really up to the task of real-time emulative synthesis. Still, my anguished cry of "There has *got* to be a better way!" resonates through that stairwell to this day.

Seventeen years and countless hernias later, the age of software synthesizers is upon us. In fact, it's in full bloom, and major stars such as Herbie Hancock are touring with virtual substitutes for their signature keyboards. You can now find software emulations of everything from classic analog synthesizers to electromechanical keyboards, drum machines, and more.

I decided it was time to put some of these impostors to the test by comparing them with the instruments that inspired them. I put software and hardware side by side and played them "blind" for a panel of experts with critical ears (see the sidebar "Meet the Panel of Experts"). The results were enlightening.

ANALOG HEAVEN

The scene for the showdown was the Audio Playground Synthesizer Museum (www.keyboardmuseum.com) in Orlando, Florida. Founder and curator

Is It Real or Is It Emulated?

Joseph Rivers has assembled what is arguably the world's biggest and baddest collection of keyboards, synthesizers, and drum machines, along with an enormous collection of service manuals, product literature, and other paraphernalia. Think of the most obscure instrument that you've ever encountered; he probably has two of them and can tell you where to find a replacement for the widget that always snaps off in transit.

Clearly, acquiring the reference instruments wasn't going to be a problem. Figuring out how to test them fairly, however, was. At the top of our long list of questions was how to judge the characteristic sound of an instrument that hasn't been manufactured in a decade or more. Should we use a "Mark I" or a "Mark II"? The original version or the "improved" version? For that matter, how much does a 20-year-old instrument sound the way it did originally, and how much did one instrument sound like the next one off the assembly line?

FIG. 2: Native Instruments' FM7 produces excellent Yamaha DX7 emulations for a number of sounds. It combines admirable sonic detail with good Velocity response.

Ultimately, we realized that accounting for all possible variables was simply impossible; we needed to get over ourselves and be practical. We set up the computers and the classics side by side in the Vintage Room of the museum (see **Fig. 1**), where most of the test instruments were already hanging on the walls. We then adjusted the hardware and software instruments until they were as close to each other as possible and played them for the panel of experts listening in the control room (which, by the way, features 80 synthesizers and more than 1,000 MIDI channels).

The virtual instruments resided in either a Mac G4/733 MHz running a Digidesign Digi 001 or in a Toshiba Celeron 1 GHz notebook with an RME Hammerfall DSP CardBus interface connected to a Multiface I/O box. We split the MIDI output of a controller keyboard to the two computers to ensure consistent MIDI response. Soft and hard instruments all went into the same mixer and were bused from there to the control room. It's worth noting that latency was not an issue with any of the virtual instruments. In fact, we were well into the tests before it even occurred to anyone to think about it.

All of the virtual instruments feature stereo outputs, because none of the hardware instruments did; however, we

MEET THE PANEL OF EXPERTS

BRIAN SMITHERS

The panel of experts from left to right: Sam Zambito, Per Danielsson, Andy Hagerman, and Joseph Rivers.

Keyboardist **Per Danielsson** teaches jazz studies at the University of South Florida in Tampa and has two CDs that are both in the final stages of production: one featuring his trio, and the other featuring a collaboration with drummer Danny Gottlieb.

Andrew Hagerman (www.singularityarts.com) is a bass player, a composer, an ILDA Award–winning producer of LASER shows, and an associate course director of advanced audio workstations at Full Sail Real World Education.

Engineer, producer, and synthesizer authority **Joseph Rivers** operates the Audio Playground studio and its world-famous Synthesizer Museum and has worked as a consultant and sound designer for major synthesizer manufacturers.

Trumpeter **Sam Zambito** (www.samzam.com) is one of the first and most prominent proponents of Nyle Steiner's Electronic Valve Instrument and has done sound design and programming for the likes of Michael Brecker and Bob Mintzer.

Is It Real or Is It Emulated?

listened to everything in mono. We also made a point of disabling any built-in effects so we could focus on the raw sounds. Keep in mind that to many users, those features are reasons to prefer the virtual versions, but our priority was to evaluate the accuracy of the emulations.

I was in charge of tweaking and playing, and Rivers sat with the panel, monitoring their observations and recording the examples for posterity. The panelists had only a general idea of the instruments they were to hear, and I identified the instruments by letter or number. We set up a talkback mic, and occasionally, the panelists would ask to hear an example again or to hear a different musical figure played. In rare cases, they made fun of my wrong notes.

IN THIS CORNER . . .

Our first bout was between a Yamaha DX7 and Native Instruments' FM7 (see **Fig. 2**). We thought that would be the easiest place to start, because we could match the factory presets that Native Instruments ships with the FM7 against cartridges of DX7 factory patches. In some cases, the match was startling, but in others it was hardly a match at all. Getting to the source of the differences would have been an interesting but time-consuming endeavor, so we declared the glass to be half full (more than half, actually) and singled out the best matches for comparison.

I played a range of sounds, including an electric piano, a synth lead, a harpsichord, and a flute. First I played the FM7 (test A) and then the DX7 (test B). Although there was some discussion over the proper use of the term *warm* (musicians!), the panelists all felt that the DX7 had more midrange presence, especially in the electric-piano sound. In contrast, the FM7 had more harmonic activity in the sound and was somewhat brighter.

Electronic Valve Instrument guru Sam Zambito summed it up this way: "I concluded that A was the hardware instrument; but I was hoping it was the virtual instrument, because I liked the sonic detail." That theme was repeated throughout the day as panelists inevitably considered which sounds they would like to have in their personal arsenals. In general, they were pleased that the virtual instruments had sufficient character.

The FM7 has a digital slider that lets the user reduce the bit depth of its output to match that of the original instrument's output. According to Native Instruments, "Unlike all the other features of the DX sound engine, which the FM7 emulates exactly, the quantization noise characteristics are only approximated." At a setting of zero, there is no bit reduction, and at a setting in the 10 to 15 range, the noise is said to be close to that of the DX7. Armed with that bit of information, I made some adjustments and played the electric piano again. At a setting of 14, the panelists felt the sounds were much more closely matched, suggesting that the digital slider does a good job of making the emulation more convincing.

I also noticed that the FM7 was responsive to key Velocity. That's one of the things I liked most when I first checked out the instrument. The responsiveness not only makes it easy to play colorfully, but it also makes it easier to distinguish the real from the virtual. To compensate, I set up the MIDI Out of the DX7 to trigger the FM7 and selected DX7-style Velocity response (0 to 100 instead of 0 to 127) in the FM7's preferences. That made the electric piano sound almost indistinguishable between the original and software instruments.

At its best, the FM7 was capable of near-perfect emulation of the DX7. Where our panel heard differences, they generally preferred the FM7's brighter, richer sound. Typical was Rivers's comment that the FM7 often "sounded brighter and better to me for today's recordings."

AND WEIGHING IN AT . . .

Despite the begging and pleading of roadies everywhere, the behemoth Hammond B-3 has found its way into practically every genre of popular music during the past four decades. It has also been sampled and simulated in a variety of ways in an attempt to capture the famous sound without the lugging. We pitted a real B-3 against two contenders: Native Instruments' B4 (see **Fig. 3**) and Emagic's EVB3.

The very things that organists love about the B-3—the nearly infinite

FIG. 3: Native Instruments' B4 offers a reasonably authentic emulation of a Hammond B-3 and was complimented for its clear sound.

Is It Real or Is It Emulated?

combinations of stops, percussion, and vibrato, as well as the always cool Leslie rotary speaker cabinet—make accurate comparisons almost impossible. Rather than spending a solid week testing every variable, we opted to limit the subjective testing to single manual sounds without the more exotic extras.

The first test was with EQ, percussion, vibrato, and other parameters disabled and all the stops fully out. With that setting and after carefully matching volumes, I played some scales, arpeggios, and simple melodies on each of the three instruments. The first thing that jumped out was that with identical settings, the EVB3 was noticeably brighter than the other two. I compensated by trimming some of the higher stops back a bit, and although that helped somewhat, the EVB3 continued to stand out as a generally brighter sound. Qualitatively, the panelists didn't feel that was necessarily a bad thing but merely a difference that made it identifiable.

The B4 came close to the B-3 at the same settings, but there was something about the warmth of the real B-3 that made it easy to pick out. As I adjusted the stops to reshape the timbre, matching the sounds as closely as I could, nobody was fooled. The B4 got compliments for its clear sound, with the B-3 being regarded as the darkest of the three.

Rivers asked me to play a series of single notes on each organ to judge the attacks. "Because the tonewheels of a real B-3 are constantly turning, the attack of a note can come at any point in the waveform, lending variety to the articulation of the instrument," he says. That was yet another subtle characteristic that separated the real instrument from the software versions.

Throwing pure subjectivity aside, we explored some of the presets on the B4 and EVB3 to see how they sounded at their best. Even on wildly different settings, it was easy to identify each organ. They all had their own characteristic sounds that shone through regardless of their specific parameters. The EVB3 got high marks for its grit and edge, whereas the B4 continued to sound a little more authentic.

FIG. 4: Applied Acoustics' Lounge Lizard EP-1 offers a fine emulation of a vintage Fender Rhodes.

DUSTY RHODES

I got sneaky on the next comparison by throwing in *two* Rhodes electric pianos to compare with Emagic's EVP88 and Applied Acoustics' Lounge Lizard EP-1 (see **Fig. 4**). That caused our esteemed panelists to doubt themselves as they tried to identify which instruments were originals. Interestingly, they weren't so confused that they failed to identify at least one of the real pianos as being nonvirtual. Rivers identified the Rhodes 73 as the real one and then, assuming that only one was an original, declared the 88-key Stage Piano Mark I to be "the best virtual instrument I've ever heard."

The other interesting thing about the electric-piano comparison is that even though they could confidently pick out the original ones, nobody cared. "I'd be happy playing any of those pianos," says Per Danielsson. The other panelists were similarly pleased with the virtual pianos.

Rivers asked for a crescendoing series of single notes on each piano, and we noted that the real ones exhibited more bark as the notes were struck harder. Zambito observed that the real pianos also had more tine distortion, which resulted in a more varied attack compared with the clean and consistent attacks of the virtuals.

Meanwhile, behind the curtain, I had some observations about playing the four pianos. The first thing I noticed was how mushy the "ancient" keyboards of the two Rhodes instruments were. Even though I was playing a light semiweighted MIDI controller, the virtual pianos were more comfortable to play. Had the panel members heard the resounding clunk of the Stage Piano's damper pedal, they would have had even less doubt about which one was hardware. Score one for virtual instruments.

However, when I played the Lounge Lizard with the damper pedal down, we noticed that the instrument's note-allocation implementation could lead you into trouble. I played an arpeggio with the pedal down, and when I reached the maximum polyphony, the first notes gave way to the new notes, as you would expect. Then I played an arpeggio and repeated the top note several times; the same thing happened. Apparently, each

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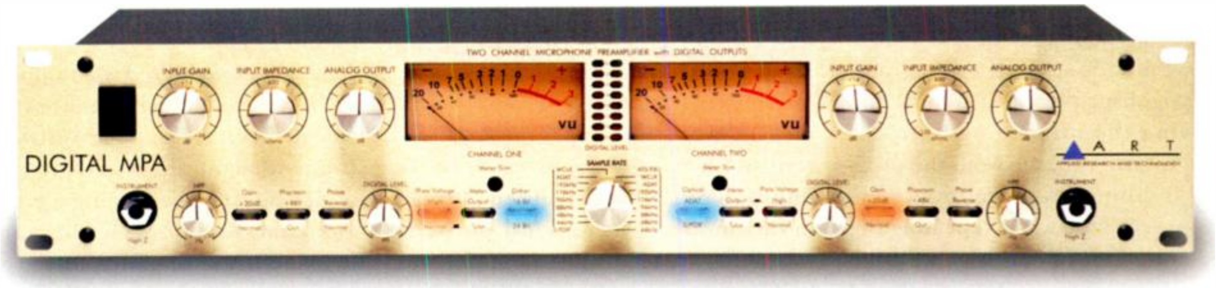

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time you strike a repeated note, it eats up a note of polyphony, eventually causing the lower notes of the arpeggio to disappear one by one, leaving only the repeated note.

We compared that with the EVP88's behavior and found that the EVP88 handles that situation much more gracefully, recognizing the repeated note and preserving the other notes. On the Lounge Lizard, you can avoid the problem by holding down the lower keys with your left hand while repeating the upper note. Out of curiosity, we tried the same experiment with the FM7 and found that it sacrificed the oldest notes regardless of whether you held the keys down by hand or just used the damper pedal. We all agreed that was the least attractive behavior. Score one for hardware and the EVP88.

NOT FOR PROPHET

One of the most challenging comparisons (for me, anyway) was the Sequential Circuits Prophet-5 against the Native Instruments Pro-52. For those two, I had to choose some Pro-52 patches and then attempt to match them by hand on the Prophet-5. I started by visually matching the positions of the numerous knobs and then adjusted them by ear until I had a good match. Fortunately, the Prophet-5 does have user memory, so I could save several patches and switch between them without delay for the demo.

The Prophet-5's keyboard is not Velocity sensitive, so I had to be careful that I didn't give away the Pro-52. My MIDI controller had a fixed Velocity mode, but it made the Pro-52 sound dull and dark. Apparently, the keyboard sent out a fixed Velocity value of 64 rather than 127, triggering the Pro-52 in the middle of its response curve. It was much easier to just hammer the keyboard to ensure I sent all 127s. That technique matched the Prophet-5 much more closely.

The Prophet-5's keyboard also had some bad keys, so I had to play everything in keys such as F-sharp and D-flat, taking advantage of the less-used black keys. Score another one for virtual instruments or at least for instruments less than a zillion years old!

The consensus on this matchup was the opposite of the DX7/FM7 pairing. The panel found the Prophet-5 sounds to be brighter, bigger, and more open than the Pro-52. As composer and bassist Andy Hagerman said, "It wasn't that the virtual synth sounded bad. But I was hoping for it to be the more interesting one, because I'm not terribly interested in buying and maintaining an antique keyboard. Unfortunately, the classic has a deeper, more interesting sound." Danielsson noted, however, that the Pro-52's attack was smoother than the Prophet-5's and that its sound was nice and solid.

The panel wondered what would happen if we started turning knobs on the two, suspecting that the software instrument might reveal its digital nature in such a test by producing a grainy or "zippered" response. On the contrary, the panelists were quite pleased at the smoothness of the virtual filter sweeps. The only giveaway was that the sweeps were less even because I was controlling them with a mouse instead of a knob. All of the Pro-52's parameters respond to MIDI controllers, so getting around that limitation would be simple with an appropriate keyboard or control surface.

INTERNET, MEET CLAVINET

Our final contestant was Emagic's EVD6, an emulation of a Hohner D6 Clavinet. When we were setting up, we noticed that the real Clav had an extraordinary click on the release of each key. The click is a normal part of the Clavinet sound, but this was huge. Rivers suggested a couple of ways to hide the click so it wouldn't be so obvious that it was the real deal. As soon as he left the room, I dug into the interface of the EVD6 to see if I could make the virtual Clav as funky as the physical one. Sure enough, by maxing out the Click parameter (see **Fig. 5**), I was able to get a satisfying release.

By the time we got around to the Clav test, only Zambito was left from our original panel of three, so he and Rivers had to endure my rendition of "Do You Believe in Love?" by themselves. Both were quite certain that the first instrument I played was the original D6, and

FIG. 5: The interface of Emagic's EVD6 lets you make the Hohner D6 Clavinet emulation as funky as the original, much cleaner than the original, or even far afield from the original.

in fact, they were correct. When I played the EVD6 for them, however, they shared a moment of uncertainty because of the realism of the click. In the end, it was primarily the evenness of the EVD6's click that made it sound too well behaved. (In my enthusiasm for the magnitude of the click, I overlooked the "random" slider right below it.) Still, the realism of the sound elicited the only "Wow!" of the afternoon.

We noticed that the D6 rang a lot longer on sustained chords, and in about ten seconds, I had adjusted the sustain time of the EVD6 to match. As we dug deeper into its interface, we discovered that it not only duplicates the damper slider of the Clavinet and its multiple pickups but also goes way beyond the sonic possibilities of the original.

FINAL THOUGHTS

None of us was terribly surprised to find that we could consistently identify the original instruments. Given the differences in the electronics between the instruments and the computer output stages alone, we had expected to see some telltale signs. Nevertheless, in most cases, we were able to adjust the virtual instruments into a close—occasionally even remarkable—likeness of their namesakes.

We set up the computers and the classics side by side.

Each of the virtual instruments that we tested does a good job of emulating its hardware equivalent. As Hagerman says, "I doubt even geezers like us would be able to pick out these differences in anything other than this A/B comparison." More importantly, the panel of experts felt that the software synths were viable instruments in their own right, often going beyond the emulative aspects to extend the sonic legacies of their forebears.

Although for the purposes of our tests, we were primarily concerned with the accuracy of the emulations, the virtual instruments also offer many conveniences not found in the originals, such as unlimited patches, computer-based editing, and MIDI control of things that only a technician could change in hardware. There is indeed a better way, and musicians and roadies alike have a genuine reason to celebrate.

Brian Smithers *is course director of advanced audio workstations at Full Sail Real World Education in Winter Park, Florida.* *Thanks to Andy Hagerman, Sam Zambito, Per Danielsson, and Joseph Rivers for their invaluable assistance.*

More than the Sum

Given their exceptional capabilities and ease of use, I have long puzzled over the comparative lack of attention stereo microphones get. The oversight can't be blamed on a dearth of products—no fewer than 17 microphone companies offer stereo mics, and more than two dozen different models are currently available.

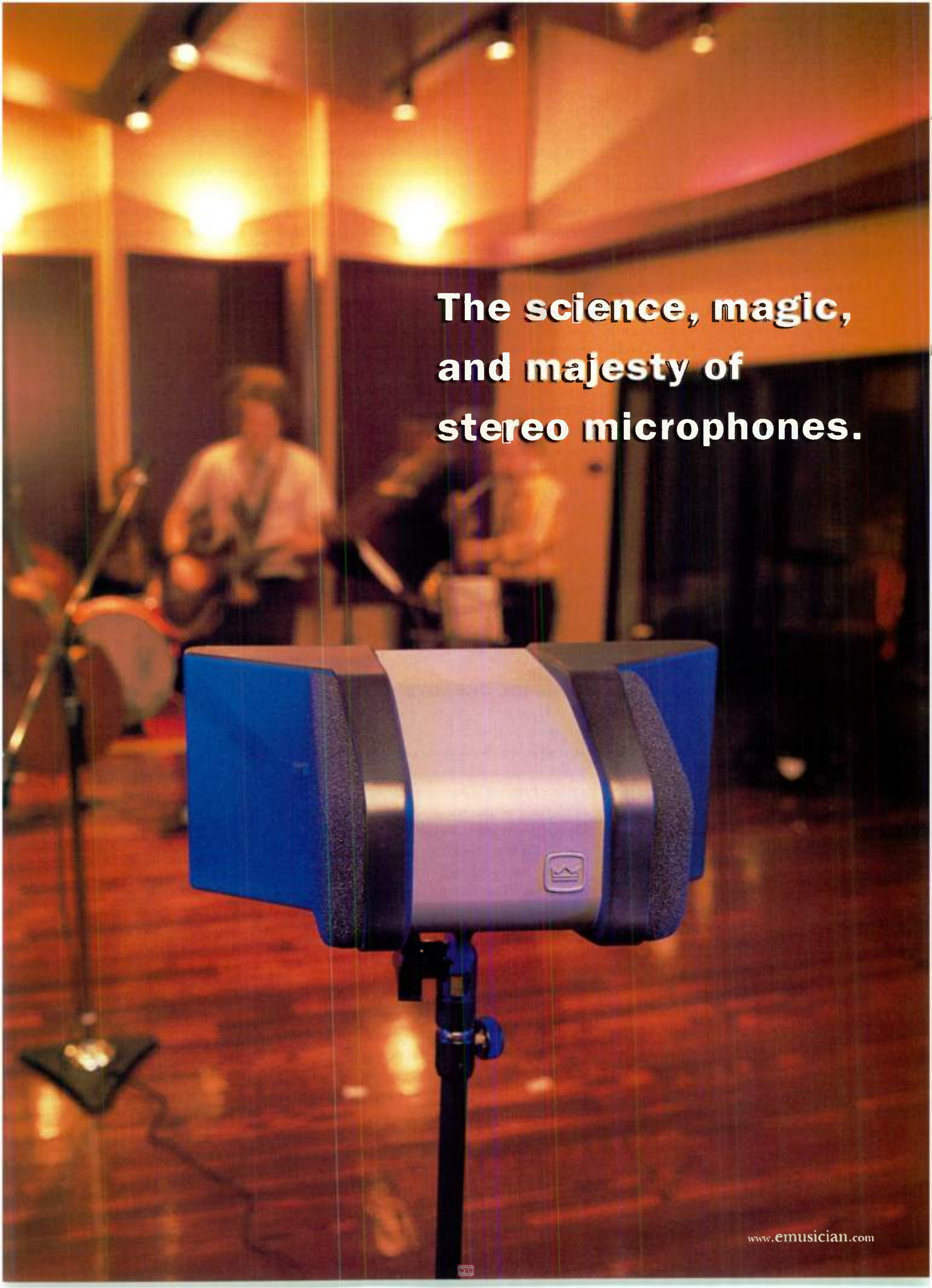
Maybe one reason stereo mics get overlooked, especially among personal-studio recordists, is a general perception that they're expensive—perhaps due to the fact that a couple of well-known models cost upwards of five grand each! Then there's the issue of variety: stereo mics come in a surprising, sometimes bewildering range of shapes, sizes, and types, reflecting a number of different approaches to capturing stereo sound. As is true of all mics, some that are particularly well suited for certain duties are often not so well suited for others. Whatever the case, the basic problem is unfamiliarity—it seems a lot of folks simply aren't aware of how useful, affordable, and downright wonderful stereo microphones can be.

By Brian Knave

This article is intended to help remedy that situation. It culminates in an evaluation and comparison of six of the most affordable stereo mics out there: the Audio-Technica AT825, beyerdynamic MCE 82, Crown SASS-P MKII, MBHO MBNM 622 E PZ, Rode NT4, and Sennheiser MKE 44P. Priced below $1,000 each, these six mics cover a range of applications at prices personal-studio and location recordists can handle. Before getting into the evaluations, I'll discuss stereo microphones in general, covering the various types and how they work, the ways in which dedicated stereo mics are superior to stereo pairs of mics (and vice versa), and how, when, and where to use stereo mics. If you are considering buying a stereo microphone, or simply want to know more about them, then step through this curtain.

Photo by Steve Jennings

The science, magic, and majesty of stereo microphones.



More than the Sum

STAGE LEFT, STAGE RIGHT

I'll start with a riddle: when does one plus one equal three? That might sound like a trick question, but actually it helps elucidate the common, if seemingly magical, phenomenon of stereo sound. As former NBC engineer Randy Hoffner famously put it, "Stereo does not equal mono times two." Consider a setup for stereo recording and playback: two mono microphones send separate mono signals to separate mono loudspeakers. Yet the result for the listener is *three-dimensional* sound (or some semblance of it, depending on the quality of the recording, placement of speakers, and so on). Indeed, that's why it's called stereo—the Greek word *stereos* means "solid" in the sense of comprising three spatial dimensions: breadth, depth, and height. Stereo, then, is more than the sum of two mono signals—a kind of synergistic leap occurs between the pair of speakers; one plus one equals three dimensions.

Of course, a stereo-recording rig isn't necessary to experience the phenomenon of stereo—the apparatus sitting atop your shoulders will do fine. Just as human vision is stereoscopic (different information from each eye is required to accurately gauge depth), human hearing is stereophonic. Try this experiment: when listening to a band or orchestra (ideally with the musicians spread out across a wide stage), close your eyes, turn your head so one ear is aimed toward the music, and shut off the other ear with your hand. Listen that way for a while. You will notice that it is difficult, if not impossible, to discern the location of individual instruments; the music seems to emanate from a single point or area. Moreover, the depth of the instruments, or distance they sit back from the front of the stage, is obscured.

Essentially, by covering one ear, you've changed your head from a stereophonic to a monophonic receiver. Now uncover your ear, turn your head to face the music (with eyes still closed), and listen again. Thanks to subtly different information reaching each ear (more on this later), you can now hear the three-dimensionality of the space and mentally pinpoint the location of individual instruments on the stage.

A stereo microphone performs an analogous feat. Its two "ears," aimed in different directions, capture two different "views" of the sound. Conveyed to two transducers (whether headphones or loudspeakers), those two views combine to re-create for the listener a sense of the original space and the music (or whatever) occurring within it. Stereo mics, then, are basically little "hearing heads," devices that approximate, to varying degrees, what human heads do in the act of hearing. That's no small feat. To the extent that stereo hearing is the human head's peak auditory achievement, stereo mics are crowning achievements in recording technology—in these devices culminate centuries of discovery in the sciences of sound, hearing, and recording.

ONE BODY, ONE LOVE

Of course, two separate microphones, properly positioned, can provide results that are virtually identical to what you get from a stereo mic. And certainly a pair of mics is more versatile in terms of placement. But the great thing about stereo mics is how easy they are to use—sort of like your head. The two capsules are already sonically matched and joined together in proper alignment on a single body, ensuring quick setup and problem-free stereo recording. For those with no experience setting up stereo pairs, that might not sound like a big deal. But if you regularly record in stereo with two mics, you know what a production it can be getting the mics positioned just right, especially up high over a drum kit or in other out-of-the-way places. Then all it takes is someone accidentally kicking a stand or tripping over a cable to knock the whole thing out of whack.

Contrast that with a stereo mic. Just set the thing up, aim it at the sound source, and you're pretty much ready to roll. For me, a gigging musician and recordist dedicated not only to playing but also to writing and producing music, stereo mics are a godsend. I can take my stereo mic and portable DAT recorder to gigs, for example, and with that bare-minimum setup document my playing—great for learning purposes—in high fidelity. I can also turn out pro-quality recordings culled from best takes. In the studio, a good stereo mic can serve up a complete drum-kit sound fast, which means that when inspiration strikes, I'm less apt to get bogged down by the gear and forget the groove I was meaning to lay down. Likewise, if a friend comes over with his guitar to play me a new song, I can be set up to record it in a matter of

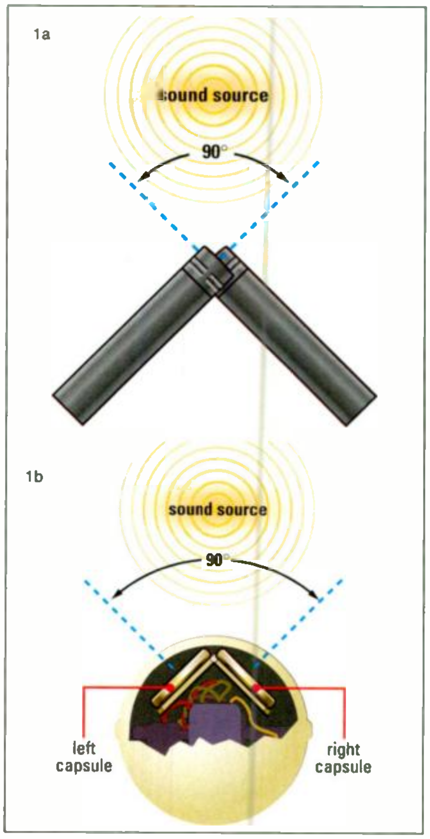


FIG. 1: The XY-coincident stereo-recording technique requires that the two capsules be positioned as closely together as possible. When using separate microphones, the best solution is an over-and-under configuration (Fig. 1a). Dedicated stereo mics, on the other hand, allow for the option of positioning the caps side by side (Fig. 1b).

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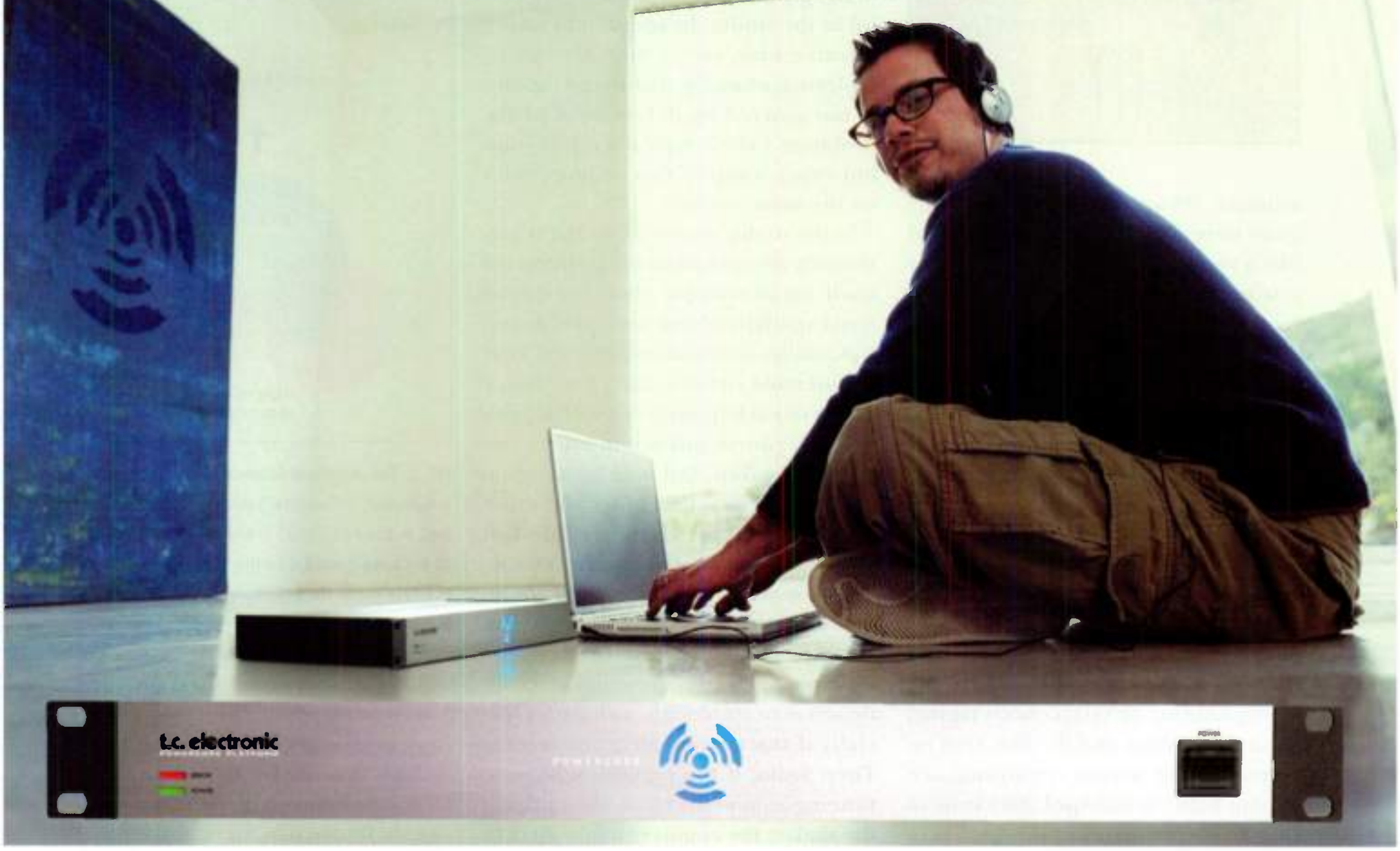

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More than the Sum

minutes. What's more, given a really good stereo mic, the result can sound like a record already, complete with a you-are-there realism that only a stereo recording can convey.

In short, a stereo mic is the quickest path to sonic realism. For those whose goal is *re-creative recording*—recording intended to convey actual sonic events—a stereo mic is the ideal tool for the job.

PICTURE THIS

Occasions that commonly call for a re-creative approach to recording (as opposed to a creative one, in which the goal is to alter the source sound) include most on-location jobs: electronic news gathering (ENG), concert taping, nature recording, and the like. (For information on nature recording, see "Going Wild" in the April 2003 issue of **EM**.) In such situations, the goal is to capture the sonic event as it happened, with minimal enhancement of the original acoustics.

Many styles of music—jazz, blues, classical, and rock, to name a few—also rely heavily on re-creative recording,

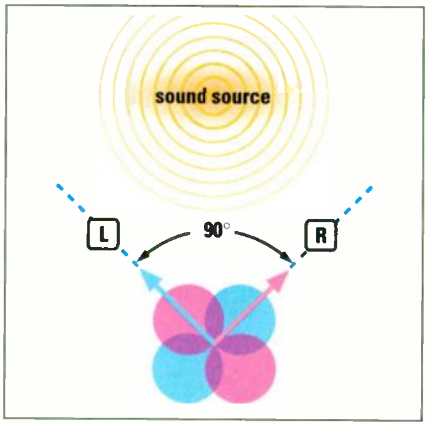

FIG. 2: A Blumlein pair specifies two figure-8 mics or capsules positioned so that the pickup axis of one is aligned with the null axis of the other. This stereo technique picks up sound equally, albeit in opposite polarity, from the front and rear.

making stereo microphones very useful in the studio. In addition to saving valuable time, stereo mics are largely foolproof, ensuring that stereo capture is not marred by deleterious phase problems (which typically result from improper setup of two or more mics on the same source).

In the studio, stereo recording is particularly advantageous for instruments made up of multiple elements spread apart spatially: drum sets, percussion rigs, mallet instruments, and any kind of ensemble (strings, a horn section, a group of backup singers, or what have you). Of course, any sound source can be stereo-miked, and even many smaller instruments sound fuller and more natural in a mix when recorded in stereo. For that reason, engineers sometimes use stereo-miking to highlight instruments that are central to a mix—an acoustic guitar in a country song, for example. Conversely, it would make little sense to stereo-mic a shaker, especially if track count were a concern. Then again, if the percussionist were running around the room while playing the shaker, the engineer might elect to record the part in stereo to convey that movement to the listener.

That brings us to another consideration in determining whether to record something in stereo: motion. Any element that moves across the soundstage—someone opening a door and walking across a room, for example—is an excellent candidate for stereo-miking. The effect can be very dramatic and is in a different league from the simulated movement that can be created by dynamically panning a mono signal during a mix.

Another application perfectly suited for stereo mics is capturing natural reverb, as in a recording space ("room miking"), echo chamber, or reverb tank. Reverberation, after all, is a stereophonic phenomenon—in the real world it comes from all around the listener. You may have noticed that if you sum a stereo mix to mono, the reverb largely disappears. That's because with only one channel reproducing the signals, the reverberations must emanate from the same place as the direct

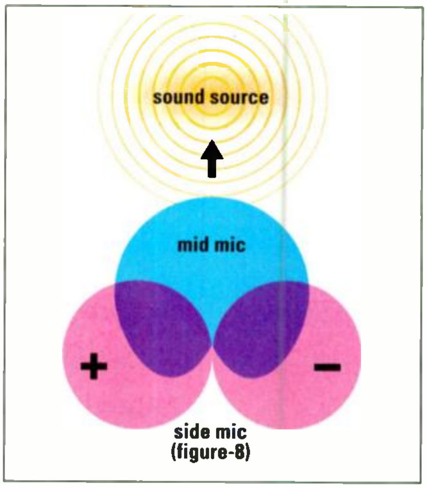

FIG. 3: The middle-side stereo technique employs a directional ("middle") mic aimed at the source and a bidirectional ("side") mic positioned to pick up sound coming from the sides. The signals must be properly decoded to create the final stereo image.

signal—a totally unnatural situation.

A stereo field, on the other hand, allows more space around the signal, making it easier for the listener to differentiate between direct and reflected signals. It is, in part, by "calculating" the difference between the arrival of the direct signal and that of its multiple reflections as they bounce off the walls and around the room that the brain is able to determine general dimensions and sonic qualities of a space. Two channels—and both ears—are required to make that calculation. That's why classic echo chambers are typically equipped with stereo mics or pairs of mics. (For information about creating your own echo chamber and/or reverb tank, see "Recording Musician: Keepin' It Real" in the February 2003 issue of **EM**.)

CUES TO THE KINGDOM

We've established that it takes two ears (or mic capsules) to register the minutely differing cues that allow the brain to determine spatial information from a given sound field. Now let's consider the main cues: intensity and phase.

Intensity refers to amplitude, or sound energy per unit of area. Differences in intensity result from the simple fact that the closer you get to a sound source, the louder it sounds, and vice versa. That

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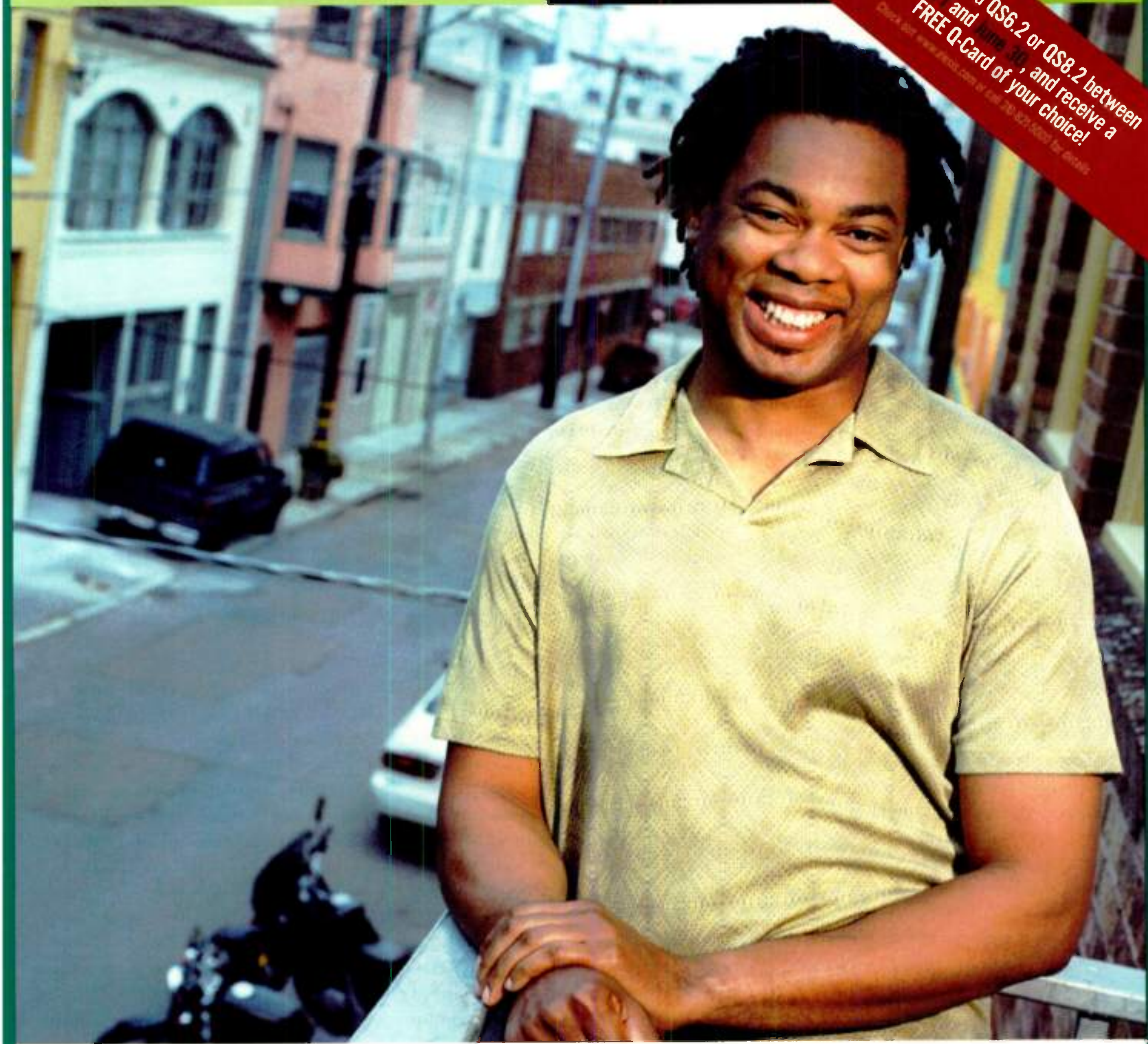

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is, the brain interprets louder sounds as being closer than quieter ones.

Phase refers to the timing of the signals. Differences in phase result from the signals arriving at the two ears (or mic capsules) at different times. This results in some degree of phase distortion or incoherence. (Conversely, signals that arrive at the same time are said to be *phase coherent.*) As we'll see, some phase distortion in stereo recordings can actually be perceived as an enhancement—it's what creates a sense of "air" around the sound sources. Beyond a certain point, though, its effects begin to degrade the sound.

To localize sounds, the ear and brain system processes those two cues in differing amounts, depending on the frequency of the incoming sound. Low frequencies have big sound waves that go right around the head, and thus are perceived omnidirectionally, as if coming from all directions. High frequencies, because the smaller sound waves are effectively blocked by the head, are heard very directionally—a small wave coming from one side reaches mostly one ear, so the main thing registered between the ears is an intensity difference. And midrange frequencies are localized using a combination of timing and intensity cues.

FIG. 4: The petite Schoeps CMXY 4V, which we used as a benchmark for the evaluations, is one of the sweetest concessions ever made to XY-coincident stereo recording. Its unique gear mechanism allows variable control over the XY pickup angle.

BY COINCIDENCE

As noted earlier, the variety of shapes, sizes, and capsule configurations among stereo mics reflects (no pun intended) a sizable range of techniques devised to capture stereo sound. That range can be divided into two broad categories: coincident and noncoincident. (Those designations apply equally to stereo mics and stereo pairs of mics, of course.) Basically, coincident means the two capsules are positioned close together and noncoincident means they are spaced apart.

In coincident techniques, the two capsules are positioned as close together as is physically possible. In the case of an end-address stereo pair (two small-diaphragm condensers, for example), this is typically done with one capsule directly above the other (see **Fig. 1**), so as to accommodate the mic bodies. A stereo mic, on the other hand, permits the option of positioning the capsules side by side. Either way, the capsules are angled symmetrically on either side of the midpoint of the sound source so that each picks up one side, or half, of the stereo image. Due to the close proximity of the capsules, incoming signals arrive nearly simultaneously at the two diaphragms, largely avoiding any phase distortion. This ensures not only a stable, mono-compatible image, but also *angular fidelity* to the source sounds. Angular fidelity means the sounds are positioned accurately across the reproduced soundstage, true to their positions in the original sound field.

Because coincident techniques preserve only intensity cues, they are often called *intensity stereo.* With intensity stereo, the character of the stereo image is determined by three things: the choice of polar pattern, the angle between the two diaphragms, and the position of the two capsules relative to the sound source. Time-of-arrival differences are not part of the equation.

BRIAN KNAVE

FIG. 5: This shot of Ex'pression Center's Studer control room shows the Studer D950 M2 console, Dynaudio Acoustics BM15A close-field monitors, and state-of-the-art Meyer X10 main monitor system (soffit-mounted).

Common coincident techniques include XY, middle-side (M-S), and Blumlein. XY seems to get the most use, both in terms of stereo pairs and dedicated stereo mics. Indeed, four of the six microphones evaluated in this article have XY-configured capsules. In an XY setup, two directional capsules (cardioid, supercardioid, or hypercardioid) are angled somewhere between 60 and 135 degrees apart. In most lower-priced stereo mics, the angle is fixed, typically at 90 or 110 degrees. However, some costlier stereo mics with XY capabilities provide variable control of the angle. Thanks to the directional pickup of the capsules, XY is an excellent choice when you need to minimize sound—audience noise, for example—coming from behind the caps.

The Blumlein technique, named after audio pioneer and inventor Alan Dower Blumlein, specifies two bidirectional (figure-8) microphones or capsules positioned so that their principal axes are at 90-degree angles to one another (see **Fig. 2**). In that configuration, the main pickup axis of each mic or capsule is precisely aligned with the null axis of the other. The result is very accurate stereo

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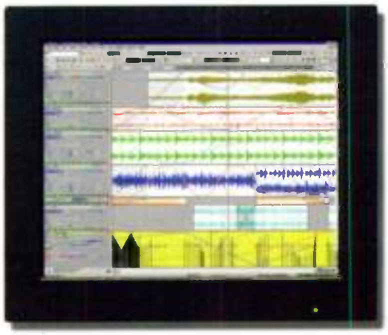

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representation—the most accurate, many believe—of the original stereo sound field for sounds arriving from the front. Of course, the Blumlein array also picks up sound equally from the rear (albeit in opposite polarity). This can be an advantage or disadvantage, depending on the situation. In the case of a great-sounding performance in a great-sounding space, a Blumlein setup might be just the ticket; but if the space is acoustically undesirable—overly reflective, too noisy, or what have you—the Blumlein array would be a poor choice, given its faithful reproduction of total-room acoustics.

The middle-side technique, unlike other coincident-capsule setups, makes use of two different polar patterns. Traditionally, the "middle" mic or capsule is directional (usually cardioid) and the "side" mic or cap is bidirectional. The mid mic is aimed at the source, or 0-degree axis; the side mic is positioned vertically coincident with the mid mic such that its null plane is aligned with the principal axis of the mid mic (see **Fig. 3**). That way, the side mic picks up information mainly from the left and right and largely rejects sound coming from the front and rear. Because the two sides of the bidirectional mic are aimed in opposite directions, they produce opposite-polarity signals, essentially dividing the room into distinct halves—polar opposites, actually.


BRIAN KNAVE

FIG. 6: Ex'pression Center Sound Arts instructor Eli Crews (left) confers with student André La Velle during one of many stereo-mic listening sessions.

One advantage of M-S recording is that you can adjust the width of the stereo spread after the recording is completed—a real boon in situations where the side mic picks up more extraneous noise than you had counted on. This is done simply by changing levels of the two channels relative to one another: increasing the level of the mid signal tightens the stereo image and makes the source seem closer; increasing the level of the side signal makes the source seem more distant.

To produce a stereo image, the mid and side signals first must be "decoded" in a special sum-and-difference matrix. Basically, the mid and *positive* side signals are combined to create a new (directional) signal aimed in one direction (typically left), and the mid and *negative* side signals are combined to create a new (directional) signal aimed in the opposite direction. That is, the left channel equals mid *plus* side (M+S) and the right channel equals mid *minus* side (M–S). Together, these two new signals form the final stereo image. Engineers who specialize in M-S recordings often use dedicated sum-and-difference modules, which guarantee proper matrixing. Certain stereo M-S mics, such as the Shure VP88 and Pearl Labs MS 2, also provide built-in sum-and-difference matrixing. (For more information on M-S recording, as well as stereo recording in general, see "Double Your Pleasure" in the June 2000 issue of **EM**.)

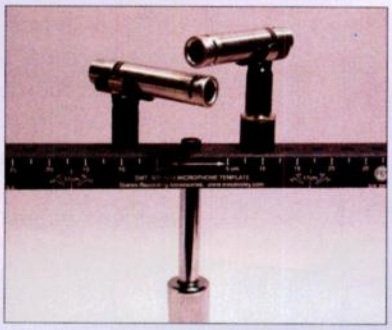
STEREO HEAD

Noncoincident techniques, by virtue of employing two capsules spaced apart, are able to capture both intensity and timing cues—just like the human head with its noncoincident ears. As a result, they tend to sound more open than coincident techniques, providing an increased sense of "space" or "air" around the performers. That additional sense of openness results from phase anomalies introduced by time-of-arrival differences between the capsules. That is, the phase distortion, which might be heard as "comb filtering" or "phasiness" when the signals are summed to mono,

DREAMING AS ONE

If you already own a matched pair of microphones, you can enjoy some of the convenience and quick setup of a dedicated stereo mic by employing a device called a microphone-array positioner. After attaching and correctly orienting the mics on the positioner, the whole assembly can be moved as a unit, maintaining capsule alignment and thus stereo integrity while you search for the sweet spot.

Shown here is the Audio Engineering Associates (AEA) Stereo Microphone Template ($145). The Stereo Microphone Template can be stand-mounted vertically or horizontally, hung from the wall or ceiling, or even held by hand. Included with each SMT is a copy of AEA's helpful Audio Engineering Society review paper "Basic Stereo Perspectives." (For more information, go to www.wesdooley.com.)



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is heard as an enhancement to the stereo field. But again, there are limits on how much phase distortion the ear will accept before it begins to perceive the sound as unfavorable.

The head is the inspiration for most noncoincident recording techniques—obvious by the fact that they employ capsules spaced six and a half or so inches apart, which is the average width of human heads. Indeed, the spectrum of noncoincident techniques might be viewed as a continuum ranging from those most like the human head to those least like it. At one extreme (the one least like the head) are widely spaced pairs, also called A/B stereo, which in some cases are positioned several feet or even yards apart. (I won't delve into those here, given that the focus of this article is stereo mics.) Most noncoincident techniques, however, take their lead from the head, at least in terms of capsule spacing; indeed, they are often described as near-coincident rather than noncoincident.

If we rank the near-coincident techniques in order from most like the human head to least like it, binaural comes first, followed by quasi-binaural, and then "closely spaced pairs" (which may or may not comprise a single microphone). Most readers are familiar with binaural recording, which typically involves the use of a dummy head fitted with omnidirectional capsules inside the ear canals. An even closer-to-reality approach is to employ a headset binaural recording system, such as the Sennheiser MKE 2002 or Soundman OKM II K, which permits the user to wear the mic capsules (they generally attach at the entrance of the ear canal) and thus generate directional cues with his or her own head. (For a review of the Soundman OKM II K, see the January 2002 issue of **EM**.)

But though binaural comes closest to replicating the audio functioning of the human head and results in recordings with an uncanny sense of realism, the recordings translate poorly to stereo speakers due to crosstalk interference—the listener's right ear hears unwanted sound from the left speaker and the left ear hears unwanted sound from the right speaker. The crosstalk degrades both stereo imaging and frequency response. To hear binaural recordings properly reproduced, the listener must wear headphones—a somewhat cumbersome requirement that probably accounts for the relative unpopularity and general unavailability of binaural stereo recordings.

Various approaches have been devised to get around this unfortunate aspect of binaural recording. Those that employ some type of barrier between the mic capsules are described as quasi-binaural; those without a barrier are generically called spaced pairs (or closely spaced pairs). In either case, the goal is recordings that capture both intensity and timing cues, thus providing a more spacious sound, yet still translate well to loudspeaker playback both in stereo and mono.

Two of the stereo mics evaluated for this article—the Crown SASS-P MKII and the MBHO MBNM 622 E PZ—qualify as quasi-binaural. The Crown Stereo Ambient Sampling System (SASS) employs two boundary-layer or Pressure-Zone Microphones (PZMs) positioned 6.7 inches apart on either side of a large plastic housing. (The original, SASS-B mic could be retrofitted with higher-quality, omnidirectional Brüel & Kjær mics, but due to low demand that option is no longer offered.)

The MBHO MBNM 622 is a variation on the Optimal Stereo Signal (OSS)

STEVE JENNINGS

FIG. 7: Recording the Garth Steel Klippert band "unplugged" required positioning the performers so the mics would capture an essentially finished stereo mix. From left to right are David Cooper, Bill Noertker, Dan Nelson, Garth Klippert, Carroll Ashby, and Tom Griesser. The mic is Audio-Technica's AT825.

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More than the Sum

technique invented by Jürg Jecklin of the Swiss Broadcasting Corporation. Jecklin's stereo-miking system, better known as the "Jecklin disk," specifies a pair of omni mics spaced 6.5 inches apart and separated by an acoustically opaque baffle (the disk) 11.02 inches in diameter. The MBNM 622 employs two boundary-layer caps (boundary-layer mics are naturally omnidirectional) mounted on either side of a metal plate (which serves as a boundary for the caps) and separated by a foam-rubber-covered half disk.

Two closely spaced pair techniques also warrant mention. The ORTF array, named after the French national broadcasting agency *(Office de Radiodiffusion Télévision Française)* that developed it, specifies two cardioid mics spaced 6.69 inches apart and angled outward at 110 degrees—sort of like two ears without the head in between. Widely used in Europe, the ORTF configuration generates just the right amount of comb filtering (predominantly in the 1 kHz region) to create a sense of air around the source, but without noticeably degrading the sound. ORTF typically produces an evenly spread stereo image with good localization, and the resulting signals sound good both in stereo and mono playback.

The NOS configuration, named after the Netherlands Broadcasting Foundation *(Nederlandsche Omroep Stichting)* where it was developed, employs two cardioid mics angled at 90 degrees and spaced 11.81 inches apart—considerably wider than the head. With this arrangement, comb-filter effects appear a couple of octaves down from the ORTF's, at around 250 Hz. Though this compromises mono compatibility of the signals, the spacious stereo field captured by NOS-configured mics translates quite well through stereo loudspeakers.

Both ORTF and NOS techniques are commonly configured with separate microphones. To reduce hassle, a convenient device called a microphone-array positioner can be used to hold the properly configured mics in place, allowing them to be positioned as a unit (see the sidebar "Dreaming as One"). Also notable is a tidy ORTF stereo mic from Schoeps, the MSTC 64g, which comprises a matched pair of cardioid caps mounted on either end of a handy T-bar-shaped body (see www.schoeps.de/E/mstc64.html).

IT'S ALL RELATIVE

In the evaluations and comparison tests of the six stereo mics, I was not so much looking for a "winner" as for a close understanding and appreciation of each microphone. A strictly apples-to-apples comparison wasn't an option anyway, given that four of the mics are XY stereo and the other two are quasi-binaural. But that disparity doesn't diminish the utility of the testing. The crux of the matter, after all, is to reveal each mic's "personality"—its sonic disposition, its strong suits and weak, its creature comforts or lack thereof. The comparison aspect simply provides context and perspective.

BRIAN KNAVE

FIG. 8: Classical guitarist Suzie Metzler, shown here with the Audio-Technica AT825 stereo mic, performed selections from John Dowland, George Gershwin, and Juan Serrano.

Speaking of perspective, I strongly felt the need for a benchmark microphone. The purpose of a benchmark mic is to establish a reliable reference so that during listening tests, you can readily switch to the benchmark tracks for a reality check. The ear, after all, is quick to acclimate to the way a given mic portrays sound sources, and it soon starts to hear that portrayal as "right"—at least until it hears a different portrayal that sounds "more right." Having a benchmark keeps things honest by helping listeners not fall prey to the easy position of overestimating a performance heard in isolation.

After much research, I selected the highly regarded Schoeps CMXY 4V ($3,475) as the benchmark mic. Like four of the mics in the test group, the CMXY 4V is an XY stereo mic employing two small-diaphragm cardioid condenser capsules (see **Fig. 4**). The CMXY's capsules are arranged side by side rather than top to bottom, allowing the two capsules to be rotated equally in opposite directions by means of an ingenious gear mechanism that keeps them perfectly aligned throughout a combined 360-degree arc. A simple twist of either gear adjusts the angle of the two capsules relative to one another. In addition, the capsule bodies and gear mechanism are mounted atop a smoothly rotating cylinder with nearly 240 degrees of travel, further simplifying positioning of this elegant and remarkably unobtrusive stereo mic.

BRIAN KNAVE

FIG. 9: I set up a wide array of drums, cymbals, and traps in order to challenge the stereo mics both timbrally and spatially.

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METHOD TO THE MADNESS

In devising tests for the six mics, I sought applications that not only would help me gauge each mic's capabilities throughout a range of performance criteria—spectral accuracy, transparency, transient response, angular fidelity, stereo realism, and so on—but also would represent typical uses for stereo mics, particularly among personal-studio recordists and/or nonprofessional concert archivists (better known as tapers). I set up three recording dates: one for a large band in a large room, another for a classical guitarist in a large room, and the third for a drum set miked from above in a small room. I also spent time with each mic individually, both in the studio recording to ADATs and outdoors recording to a Tascam DA-P1 portable DAT recorder.

I had the good fortune of getting to record the large band in the Studer Room at the wonderful Ex'pression Center for New Media in Emeryville, California. There I had the pleasure of working with two very capable volunteers whose assistance proved invaluable: Ex'pression Center Sound Arts instructor (and **EM** reviewer) Eli Crews, who operated the Studer D950 M2 console (see **Fig. 5**), and Ex'pression Center Sound Arts student André La Velle, who, like me, is something of a microphone fanatic (see **Fig. 6**). We used the Studer's fine mic preamps and 24-bit converters and routed all signals to Digidesign Pro Tools. The Studer Room features Dynaudio Acoustics BM15A (active) close-field monitors, which I am very fond of, and the mains are Meyer X10, a remarkable monitor that, I was told, employs built-in microphones that measure and adjust

FIG. 10: The Audio-Technica AT825's frequency response is predisposed to turn out well-balanced stereo recordings in a wide range of applications.

speaker response in real time according to the needs of the room.

Kind enough to perform for the tests was Garth Steel Klippert's band (www.garthsteelklippert.com; see **Fig. 7**), a six-piece original-music ensemble from the Bay Area featuring Klippert on guitar and vocals, Bill Noertker on contrabass, David Cooper on marimba, Carroll Ashby on trombone, Tom Griesser on baritone saxophone, and Dan Nelson on drums. Klippert's group nicely fulfilled a couple of criteria I had deemed critical for the tests. One was that the group be able to play acoustically, with no amplification. My reasoning for that was twofold. One, it would force us to create the final stereo "mix" by positioning the instruments strategically around the room at different distances from the microphone; that way, during playback I could judge how faithfully each mic represented the instrument positions (as well as the space between). And two, I would be able to hear how each mic reproduced the pure, unadulterated timbres of the instruments. The other criterion was that the band contain vocals, horns or strings (or both), acoustic guitar, an upright acoustic bass, and a drum kit with cymbals. I wanted to make sure I hit the mics with a range of timbres, including both low and high strings (upright bass, acoustic guitar) and low and high percussive sounds (bass drum, cymbals).

Ex'pression Center also kindly granted me use of the Studer Room for the classical guitar session. That time, however,

FIG. 11: The beyerdynamic MCE 82 has forward high mids and a somewhat aggressive sound overall. Its light weight, low handling noise, and intuitive operation make it a natural for on-location duties.

I worked alone, recording directly to DAT using the DA-P1's mic preamps and, when necessary, Blue Blueberry mic cables. Oakland-based classical guitarist Suzie Metzler generously offered her musical services (see **Fig. 8** and www .geocities.com/suziemetzler). She played a wonderful-sounding hand-built Takamine EP 90. I asked Metzler, who specializes in baroque, renaissance, and flamenco music, to choose selections that would utilize the full frequency content and dynamic range of the instrument, while simultaneously covering a range of playing styles and techniques. She selected three short pieces: a 16th-century work by John Dowland titled "An Heart That's Broken and Contrite," "Rhapsody in Blue" by George Gershwin, and a flamenco-styled piece titled "Posades" by one of Metzler's former—and favorite—instructors, Juan Serrano.

I did the drum-overhead tests at my own studio, recording to ADATs through a Manley Langevin Dual Vocal Combo preamp and Blue Kiwi mic cables. I set up the drums for maximum sonic variety and a wide lateral spread (see **Fig. 9**). For the first test, I played to a simple folk-rock track I had recorded the week before that didn't yet have drums. For the second, I wrote a rhythm chart with a slow funk groove featuring drum breaks that utilized, in a prescribed sequence, the different sounds from one end of the kit to the other. I played the chart to a click track so all the parts lined up. That way, during the listening tests, I could jump back and forth between the mic signals and hear the same basic figures.

I took pains to ensure consistent microphone positioning during each test session. That was fairly straightforward with the four XY mics, but sometimes the two quasi-binaural units required I take a different tack. In the end, though, I chose to err on the side of what sounded best, in the interests of getting to know each mic with its best foot forward.

Signal levels and preamp gain were handled a couple of different ways. In the Studer Room we kept the juice equal from one mic to the next and compensated for level differences only later, during the listening sessions. That way, the Pro Tools tracks provided an easy visual indication of how hot (or not) each mic was. For the classical guitar and drum sessions, though, I spent a good amount of time precisely matching levels and noting gain settings—a different way of formulating the same equation.

Following the three sessions, I digitally transferred the Pro Tools and DAT tracks into MOTU Digital Performer, lined them up, and bounced them to fresh ADAT tape in the same order as the tracks I had already recorded to ADAT. Once all the tracks were side by side on tape, I used mute buttons on my analog mixer to jump back and forth between channels—an easier, if more old-fashioned, way than clicking with a mouse.

NOW INTRODUCING

I'll describe each mic physically before getting to the evaluations. You can tell a lot about a mic's intended use just from its design, features, and packaging and get a good idea whether it's appropriate for your needs.

Audio-Technica AT825. Known in some circles as the "Grateful Dead mic" due to its long-standing popularity among Deadhead tapers, the Audio-Technica AT825 (see **Fig. 10**) is also popular for ENG and other field-recording duties. Hefty without being heavy, the mic has a cylindrical metal body with a matte gray finish and is topped with a hardened-mesh grille basket. The rectangular shape of the basket makes for easy mic orientation in low-light situations, and a silk-screened L/R logo identifies left-right orientation. Beneath the logo is a small recessed switch for engaging a 150 Hz highpass filter.

The mic can be powered externally by phantom power or internally by 1.5V AA battery—the top part of the cylindrical body unscrews and slides down over the lower part to reveal the battery compartment. The battery is automatically bypassed when phantom power is present—a helpful touch. (Audio-Technica also offers the $399 AT822, a battery-operation-only stereo mic with unbalanced, ¼-inch connectors that otherwise is identical to the AT825.)

The AT825 employs two miniature "electret" condenser cardioid elements. The capsules are configured side by side in a fixed 110-degree angle.

The AT825 comes with a 16.5-foot shielded cable with Cannon connectors and color-coded sleeves on the preamp end (red indicates right) and a 5-pin XLRM-type connector on the mic end. Also included is a robust mic clip (with metal threads!), a foam windscreen, a protective vinyl zipper pouch, and even a AA battery, all neatly packaged in a sturdy cardboard box.

Beyerdynamic MCE 82. The least expensive mic of the bunch, the beyerdynamic MCE 82 (see **Fig. 11**) has a long, cylindrical body that gets progressively thicker from bottom to top. The basket's metal mesh is a bit flimsy—a hard push with the thumb will dent it. An external metal "rib" bisects the basket, both strengthening it and providing a helpful visual cue to left-right orientation. A rough-coat, matte gray finish provides a nice, nonslip feel, and the mic is light—less than half a pound—making it a good pick for lengthy handheld sessions. (Though too late for inclusion in this article, beyerdynamic recently released another affordable XY stereo mic, the $429

FIG. 12: The Crown SASS-P MKII excels at capturing broad, well-balanced, and phase-coherent images of total-room acoustics.

MCE 72, which has electret condenser elements angled at 120 degrees.)

Like the AT825, the MCE 82 can be powered externally by phantom power (8 to 52V) or internally by 1.5V AA battery. The bottom half of the mic body unscrews and can be slipped off to reveal the battery compartment. A recessed switch directly beneath the grille basket engages a 100 Hz highpass filter. A second, raised switch just below that controls battery powering via three positions: On, Off, and B.C. (battery check). When the switch is in B.C. position, a red LED between the switches lights to indicate full battery capacity—a handy reminder for fieldwork. (Careful, though—leaving the switch in B.C. position will quickly drain the battery.) Both switches are conveniently scored, and even the recessed switch can readily be moved by thumbnail. The raised switch is helpful when positioning or hand holding the mic in low- or no-light situations—you can feel the switch with your thumb, which makes for easy left-right orientation. There's also a white silk-screened L/R logo for orientation purposes.

The MCE 82 employs miniature electret condenser cardioid capsules angled 90 degrees apart in a side-by-side configuration. The elements are elastic-suspension mounted, which helps reduce handling noise.

The 8.8-foot shielded cable that comes with the MCE 82 features Neutrik connectors and has color-coded sleeves on the preamp end and a 5-pin XLRM-type connector on the mic end. No mic clip or windscreen is provided, but the mic does come packaged in a protective nylon zipper pouch.

Crown SASS-P MKII. The Crown SASS-P MKII (see **Fig. 12**) looks more like a vacuum cleaner attachment than a microphone. But bulky as the thing appears, its relatively light weight and clearly marked features make for easy setup and use.

The back of the SASS-P provides two standard XLR connectors, labeled A and B, and two 9V battery compartments that slide out like little drawers. Between the XLR connectors, which feature gold-plated posts, is a large knurled knob with two sets of two positions—a Flat and Cut position in Phantom-power mode, and a Flat and Cut position in Battery-power mode. Either Cut position activates a 100 Hz high-pass filter. A mic-stand adapter with a large-handled tightening screw is permanently affixed to the bottom of the SASS-P.

Made of molded high-impact plastic, the body of this quasi-binaural microphone is shaped to form a barrier (baffle) between the two PZM elements and angled surfaces (boundaries) beneath them. A small mount suspends each PZM a few millimeters above its angled boundary. The two boundaries block sound from the rear, causing the omni PZMs to behave directionally for mid and high frequencies; in addition, pieces of foam rubber on either side of the baffle help to limit overlap of high frequencies between the two sides, further focusing directionality of the mic's high-frequency pickup.

Low-frequency pickup, however, remains omnidirectional. Thus, at high frequencies the SASS-P acts like a coincident pair, producing mostly intensity cues; at mid frequencies it acts like a near-coincident pair, producing both intensity and timing cues; and at low frequencies it acts like a closely spaced pair, producing primarily timing cues—pretty much how the human head's hearing works. However, the clever design of the SASS-P gets around the playback limitations of binaural capture, resulting in a well-focused, mono-compatible stereo image that translates nicely over both stereo speakers and headphones.

The SASS-P MKII comes with an enormous, high-quality, foam-lined flight case, underscoring the mic's intended professional use. The case has form-cut recesses for the mic and accessories, as well as extra spaces for cables, tapes, or what have you. Accessories include a foam-covered handgrip (which screws into the mic-stand adapter by way of an included brass adapter) and a custom windscreen that fits over the mic like a well-tailored jacket. The system is complete and clearly well thought out.

MBHO MBNM 622 E PZ. If you thought the SASS-P took the prize for most peculiar-looking stereo mic, check out the MBHO MBNM 622 E PZ (see **Fig. 13**), which our **EM** reviewer described as looking "like a cross between a sundial and a futuristic spacecraft." (For the full review of the MBNM 622 E PZ, see the October 2001 issue of **EM**.)

As discussed earlier, the MBNM 622 modifies Jecklin's OSS technique by using boundary-layer rather than standard (pressure-gradient) condenser omni capsules. MBHO calls this "OSS II." Boundary-layer mics require a boundary, so the designers ingeniously bisected the Jecklin disk in the horizontal plane with a second disk made of metal. This metal disk forms the boundary (or part of it) for the caps, which are mounted beneath bullet-shaped metal hoods 2.75 inches from the half Jecklin disk on either side. The bottom side of the disk, or base of the mic, is covered with a thin layer of felt to reduce vibration and slippage. The half Jecklin disk is a plastic semicircle sandwiched between two foam-rubber semicircles for a combined thickness of almost an inch. That puts the two caps approximately 6.5 inches apart.

The metal hoods suspend the boundary-layer caps a few millimeters above the metal boundary on one end; on the

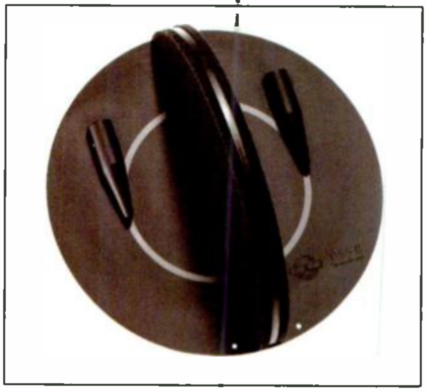

FIG. 13: The MBHO MBNM 622 E PZ's warm, mellow sound, powerful low end, and unique stereo imaging tend to inspire creative use—it's a wonderful "effect" mic with distinct capabilities.

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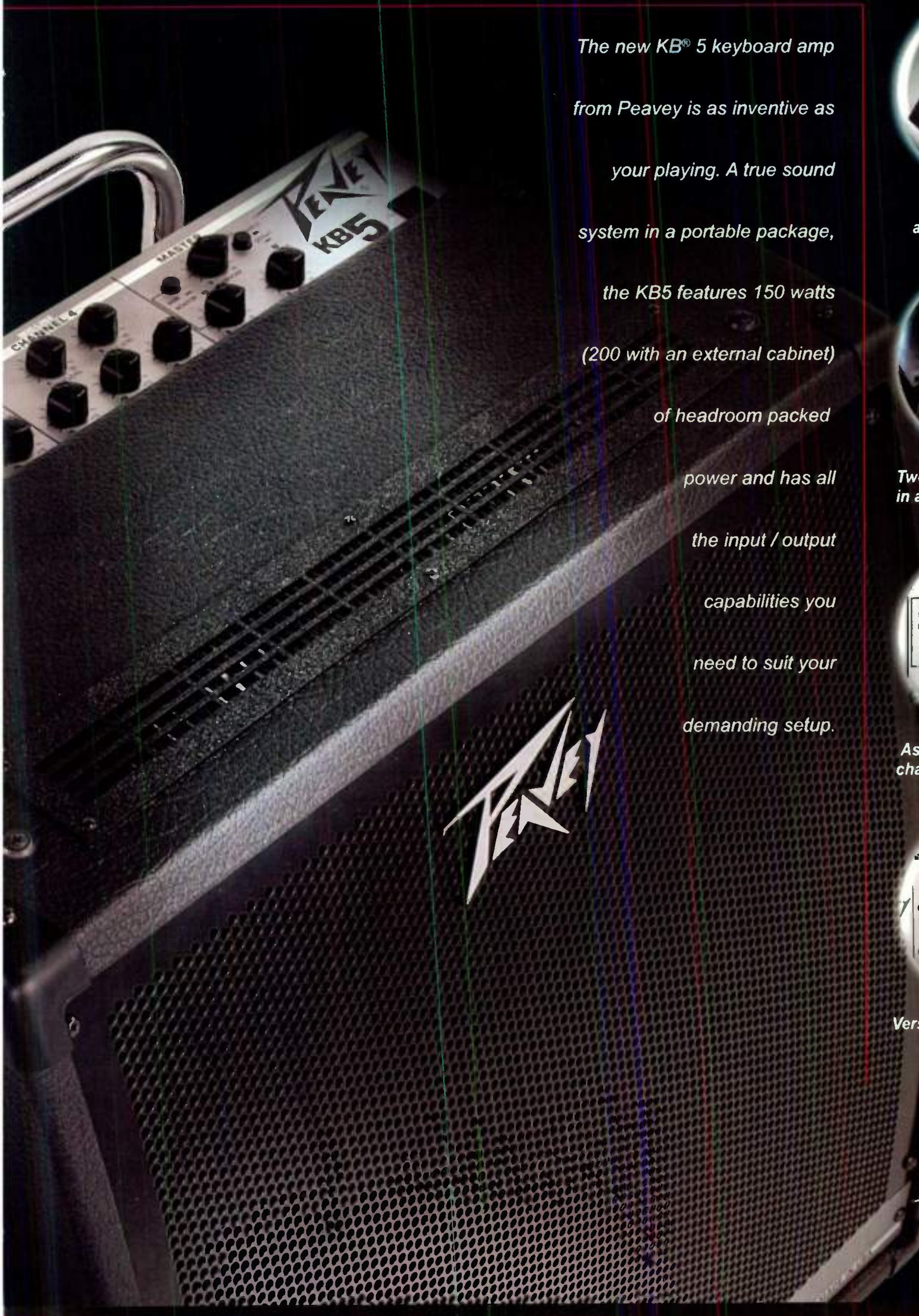

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other end the hoods open out to standard XLR connectors, which feature gold-plated posts. A blue circle painted on the boundary plate passes directly beneath the two PZMs, visually delineating capsule placement with a sort of bull's-eye. Two small holes on one end of the plate allow the MBNM 622 to be mounted vertically on a wall. (I used pushpins, which worked fine.) The mic can also be positioned on the floor or any other large, flat surface (boundary). Typically, the bigger the boundary, the deeper the bass pickup. The sound is also affected by acoustic qualities of the boundary—glass and carpet boundaries, for example, yield quite different results.

The MBNM 622 comes with no accessories, but then, it doesn't really require anything more than a couple of mic cables and phantom power. Clearly, this mic is not designed for handheld use or for outdoor duties (thus no need for a windscreen). A carrying case might be useful, though—the MBNM 622 doesn't readily fit into a small pack or bag, and I'm not so sure I'd attempt carrying it by hand through airport security these days.

Røde NT4. Of the six mics evaluated here, the Røde NT4 (see **Fig. 14**) is the new kid on the block; it came to market only last year. (For a full review of the Røde NT4, see the October 2002 issue of **EM**.)

The NT4 is a distinctive-looking mic. A cast-metal, rocker-shaped assemblage on top serves as a stereo bar, perfectly positioning the two capsules in the classic over-and-under XY configuration. The angle is fixed at 90 degrees. Unlike the other three XY mics in the test bunch, the NT4 employs "true" condenser elements (as opposed to electrets) with "full-size" ½-inch, gold-sputtered diaphragms. The capsules, which can be unscrewed from the mic for easy repair or replacement, derive their cardioid patterns by the traditional mechanical means of porting (note the two rows of narrow ports around the top of each cap).

The NT4's hefty, heavy-gauge metal body has a nice, satin nickel finish. Beneath the capsule assemblage, a rounded, internally threaded ring unscrews to provide access to the electronics inside. The bottom portion of the mic also unscrews and slips off to reveal the battery compartment—the NT4 can be powered by 9V battery as well as by 12V, 24V, or 48V phantom power.

An oval window tucks the NT4's on/off switch out of mishap's way. Just above the switch, a tiny red LED indicates battery condition. It lights briefly when the mic is switched on, indicating a good battery; if the LED stays on, the battery needs replacing.

Røde outfits the NT4 smartly with a tough, molded-plastic carrying case. Inside, cut-foam recesses fit the mic, a foam windscreen, a seriously beefy mic clip, and two shielded cables: an 11.5-inch, 5-pin XLRM to dual XLR, labeled "L" and "R" in big letters on the split end; and a 10-footer terminating in a stereo miniplug.

Sennheiser MKE 44P. The smallest and lightest of the bunch, the Sennheiser MKE 44P (see **Fig. 15**) is both unobtrusive and robust, the kind of mic you can carry around worry-free in a baggy pocket and then deploy at a moment's notice, clandestinely if need be, and hold in your hand for long periods with only minimal fatigue. The grille basket on the 44P is virtually crushproof, and the body is made of a superhard, seemingly indestructible black composite plastic with a lustrous shine. The finish nicely offsets the white silk-screened lettering, L/R indicator logo, and other graphics (which include a pin-out diagram).

A threaded, knurled ring just beneath the basket unscrews and slips off, releasing the basket and capsule assembly, which can then be pulled off—carefully—and detached from the shaft. The bottom, interior portion of the assembly makes electrical contact with the shaft via five pins. Removing the assembly reveals the battery compartment, which takes a standard 1.5V AA. The mic can also be phantom powered, of course. Two red recessed switches on either side of the mic shaft can be moved only by pen or other pointed implement. One is the on/off switch (for battery operation; when phantom is supplied the mic is automatically on). The other is a 3-position switch offering low-cut filters at 150 Hz and 250 Hz, as well as a "flat" setting.

The 44P is nicely balanced for handheld applications; its oval-shaped shaft inclines the mic to lie "horizontally" in the hand, orienting the caps side to side rather than on the vertical plane. Still, left-right stereo orientation is not especially intuitive, due mainly to the round shape of the grille basket, which offers no visual clue. It doesn't help,

FIG. 14: The Røde NT4 is a fully professional XY stereo mic utilizing classic design features. The overall sound is bright and likable, with forward high mids, great transient response, and textbook XY imaging.

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More than the Sum

either, that the on/off switch is located on the side of the mic rather than on top (top being defined as the center axis of the stereo field), or that your thumb tends to obscure the graphics—the 44P's only positive indicator of left-right orientation—when you hold the mic. Of course, none of this is an issue once you get used to it, but I do like being able to confirm mic orientation with a glance.

The 44P employs miniature electret condenser cardioid elements. The caps are configured in an over-and-under XY orientation with a fixed 90-degree angle.

The 44P comes in a sturdy, hard-plastic storage case with form-fit recesses for the mic and included windscreen and cable. The windscreen is noticeably high-quality—that kind of foam with the shiny, plush outer layer. The 3.75-inch shielded cable is also high-quality, featuring color-coded Neutrik connectors with gold-plated pins and, stamped on the cable, the Georg Neumann logo. However, the mic end (5-pin XLRM connector) splits immediately into two separate cables, which I find less convenient to work with than a single-piece cable.

LET'S ROLL (TAPE)

As we get into the evaluations, bear in mind that all the manufacturers represented here make other, high-quality microphones, and some also make high-end stereo mics. Beyerdynamic, for example, produces the acclaimed MC 742 stereo mic ($4,999), which offers both variable-angle XY and M-S capabilities. So you should take this review as reflecting only on the specific models involved, and not on the capabilities of the manufacturers—for the most part, these mics do not represent their makers' best efforts.

The key question underlying this review—what can you get in a stereo mic for a thousand bucks or less?—brings together a fairly diverse lot. Beyond the main categories of XY and quasi-binaural, the mics also fall into other camps based on design and construction. For example, of the four XY mics, three—the Audio-Technica AT825, beyerdynamic MCE 82, and Sennheiser MKE 44P—employ miniature electret condenser elements fixed inside foam-lined grille baskets, whereas the fourth, the Røde NT4, sports ½-inch-diameter, separately detachable, "true" condenser caps. That, as well as other differences—materials, weight—predisposes the electret models to ENG, interview, and other mobile, on-the-scene–type recording duties. The NT4, on the other hand, though portable, is inclined more to studio and other indoor, mic-on-a-stand–type applications.

The other two mics, though quasi-binaural, are also distinct functionally (as well as sonically). The MBHO MBNM 622 E PZ, for example, is clearly not designed for handheld applications (though I suppose you could always mount it on a board and put a handle on the back), whereas the Crown SASS-P MKII is built to accommodate both studio and street (though you might feel a bit conspicuous packing the thing around town).

As for differences between XY and quasi-binaural mics, a big one, naturally, is stereo spread. In general, hard-panned signals from XY mics do not sound hard-panned. That's because XY patterns tend to spread elements around a semicircle stretching between approximately 9 o'clock and 3 o'clock pans. Thus, the speaker image appears more in front of rather than "around" the listener.

To get an equivalent width of soundstage from the quasi-binaural tracks, I found I had to pan them in as close as 10 and 2 o'clock. At the other extreme, when the quasi-binaural tracks were panned hard left and right, the stereo image seemed to wrap all the way around the speakers, pushing elements that had appeared in the room at 10 and 2 o'clock nearly into the "corners" at 8 and 4 o'clock. (That may not qualify as angular fidelity, but I rather liked having the extra width to work with.)

BENCHMARK PRESS

Though this is not a review of the Schoeps CMXY 4V, it was the benchmark mic for the comparisons, so some observations are in order. Not surprisingly, given its pedigree and price, the CMXY was the smoothest, most accurate, and best-sounding mic of the bunch. ("That's my guitar!" exclaimed Metzler upon first hearing the benchmark tracks.) The CMXY truly did serve to keep our perspectives in line.

But something else distinguished the Schoeps mic from the others: its exceptional ability to render depth of field. All the mics were good at creating breadth in the stereo field. Some produced a wider and others a narrower spread, and some were more focused than others; but they all placed elements in their proper positions—relatively so, at least—across the stereo soundstage. The CMXY stood head and shoulders above the rest, though, for its uncanny ability to reproduce a real sense of the *depth* of the soundstage. On the recordings of Klippert's band in the large room, the CMXY's stereo image was so clear and deep I felt I could throw rocks into it and hit the different performers—I knew just how far each was from the microphone. I could even sense height in the image, reflecting the fact that Klippert had stood above the mic and aimed his voice down as he sang. In contrast, the six test mics sounded more two-dimensional; they reproduced the width of the stereo field but not much sense of its depth.

Audio-Technica AT825. The AT825 was a consistently good performer,

FIG. 15: The ultralight and sturdy Sennheiser MKE 44P is well designed for unobtrusive stereo capture. However, its sharply rolled-off low end makes for an overall thin sound.

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typically producing a well-balanced and seemingly preequalized sound. The mic tends to highlight certain defining frequencies in the lows, high mids, and highs while downplaying typically troublesome low mids. This is borne out by the AT825's frequency-response plot, which shows a mostly flat line from 30 Hz to 20 kHz, with gently rising boosts centered around 50 and 80 Hz, between 5 and 6 kHz, and around 12 kHz—all common points of enhancement.

On drums, this resulted in nice tonal separation around the kit, with kick, toms, snare, and cymbals each well represented and occupying its own frequency niche. The AT825 captured a nicely balanced view of Klippert's large band, as well, with full and present bass, forward high mids, and crisp, sometimes edgy highs ("sizzly" was the word Klippert used). In both cases, the only thing conspicuously absent was low-mid warmth. Transient response was fast, if somewhat controlled sounding. Stereo spread and angular fidelity were very good—the 110-degree angle seemed to strike a good balance between the source sound and surrounding ambience.

On classical guitar, the AT825 captured a nice balance of treble and bass, but it favored the highs and lacked in fullness and low-mid warmth. Metzler pointed out that it made the nylon strings on her guitar sound "metallic"—an inexcusable sin for classical guitar.

Spectral accuracy might not be the AT825's claim to fame, but this mic seemed to always sound good—of the three XY electret models it consistently produced the most usable tracks. I came to think of it as a point-and-shoot, hard-to-go-wrong kind of mic; it tends toward a controlled, preequalized, final-mix sort of sound, making it a natural pick for fieldwork, concert taping, and general stereo applications. Though not really a studio-quality mic, the AT825 can be trusted to produce usable results in a variety of studio applications. It's reliable and easy to use, too.

Beyerdynamic MCE 82. Beyerdynamic makes clear right up front that the MCE 82 is not intended as a professional-quality studio microphone; according to the accompanying literature, the mic is "suitable for stereo recording in ENG/EFP [electronic film production] applications, home recording, reporting, film, video and interviewing." Just the same, the 82 proved a decent performer—especially given its low price—typically producing a hotter, more forward sound than the AT825, but with less bass, fuller mids, and a tendency toward some harshness in the high mids. (The mic's frequency-response plot shows a relatively smooth, flat response from 100 Hz to 20 kHz, with the low end rolling off to approximately 5 dB down at 50 Hz. Two gentle boosts can be seen centered around 3 kHz and between 8 and 9 kHz.)

On the drums, the 82 produced an aggressive, somewhat over-the-top sound I rather liked; it helped make my dark and dead "drum room" sound brighter and more alive. (Of course, that same tendency might not be so welcome in a bright, reverberant space.) Dynamic response was impressive, as evidenced by strong attacks. Stereo spread is a bit tighter on this mic as compared with the AT825, and the sound is slightly drier. Frequency-wise, bass and low-mid content were under-represented (making the kick sound boxy), hats and snare sounded overly bright, and a small China cymbal translated very harshly. A touch of board EQ brought things into better balance—a cut at 12 kHz, boosts at 75 and 200 Hz.

The large-band tracks, too, sounded weak on lows and a bit harsh in the high mids, and again the sounds seemed more forward—closer to the listener—than with the other mics. Both Crews and La Velle noted that the "very high end" was subdued and that there wasn't as much capture of "airy" frequencies or room ambience with this mic.

EMWEB CLIPS

The 82 couldn't always handle Metzler's hard strums, particularly the *rasqueado* (a flamenco term for a four-finger-rolling hit on the strings). Metzler, though, was relieved that the 82 didn't impart much "metallic" sound to her guitar's nylon strings. In that sense, the guitar sound was truer—less equalized sounding—than the AT825's. To my ear, though, the overall sound was ever so slightly out of focus.

Though not intended for critical studio work, the MCE 82 still performed admirably all around, proving it can pinch-hit if necessary in the studio. Fieldwork is its bent, however, and here it has the advantage of light weight, intuitive operation, and a nonreflective, nonslip surface. In addition, the MCE 82 produced the least amount of handling noise of the bunch.

Crown SASS-P MKII. Remarkably, the Crown SASS-P MKII's frequency-response plot shows a flat line from just above 50 Hz all the way to 15 kHz or so, after which it rolls off in a perfect arc to 5 dB down at 20 kHz. Given that perfection, I couldn't help but wonder whether Crown had taken a few liberties in "rounding off" the response figures. Then again, nothing sounds "off" with this mic spectrally—it does a great job of representing the whole frequency range in a balanced fashion, tending to turn out complete, if somewhat controlled-sounding, "finished mixes." I didn't find myself reaching for EQ to fix things.

The SASS-P is not called Stereo Ambient Sampling System for nothing—this mic excels at capturing coherent snapshots of total-room sound. That makes it especially effective for recording large, spread-out groups (such as

More than the Sum

Klippert's band) but less so for recording single instruments at closer range (such as the classical guitar). On Klippert's group, the SASS-P's stereo picture, though unrealistically wide, was well balanced and very appealing, conveying a distinct sense of the space and the performers within it. Elements were spread noticeably further left and right than they had been in the room, creating a very wide, distinctly rectangular-shaped image. Klippert described this as "letterboxing"—a reference to the technique used to make motion pictures fit TV screens.

But for all its spacious width and delicious ambience, the SASS-P stereo image still felt somehow flat or two-dimensional, at least in comparison to the benchmark mic's. If I threw rocks into this image, it seemed, they'd tear through to the other side. (I've noticed a similar "flatness" with boundary-layer mics before, which leads me to wonder if this quality might be inherent in the PZM's method of pickup—rather than hit the caps directly, all signals must first reflect off the boundary. Does the flatness of the boundary impart a flatness to the sound?) Just the same, Klippert and I agreed that, of the six mics, the SASS-P was the overall best performer in this application.

As a drum overhead mic, the SASS-P produced a dramatic but again unrealistic sound with an overly wide stereo spread (easy enough to fix with panning, of course). As with the AT825, there was a sense of the individual elements having already been equalized. Transient response was good but not exciting, and overall the sound had a kind of controlled, "boxed in" quality—not rock 'n' roll by any means.

The literature that accompanies the SASS-P explains that putting the mic closer than 3 feet from a source typically results in a "hole in the middle" effect, so I listened carefully while positioning the mic on Metzler's guitar. But even well positioned, this was just not the right transducer for reproducing the warm, rich timbres of the classical guitar. Though it did nicely portray the spaciousness and character

Stereo Mic Specifications

MANUFACTURER	MODEL	ELEMENT	POLAR PATTERN/TYPE	FREQUENCY RESPONSE	DYNAMIC RANGE	SELF-NOISE	SIGNAL-TO-NOISE RATIO
Audio-Technica	AT825	permanently charged, fixed-backplate ("electret") condensers	stereo XY (fixed 110 degrees)	30 Hz–20 kHz (±3 dB)	102 dB	24.0 dB	70.0 dB
beyerdynamic	MCE 82	permanently charged, fixed-backplate ("electret") condensers	stereo XY (fixed 90 degrees)	70 Hz–20 kHz (±2 dB); 5 dB down at 50 Hz	104 dB	24 dB	62.0 dB
Crown	SASS-P MKII	boundary-layer "electret" stereo condensers	quasi-binaural stereo	20 Hz–18 kHz (±2.5 dB in reverberant sound field)	127 dB	20.5 dB	73.5 dB
MBHO	MBNM 622 E PZ	boundary-layer "electret" stereo condensers	quasi-binaural stereo (modified OSS)	10 Hz–26 kHz (±1.5 dB)	130 dB	30.0 dB	64.0 dB
Røde	NT4	small-diaphragm, externally polarized, DC bias capacitors ("true" condensers)	stereo XY (fixed 90 degrees)	20 Hz–20 kHz (+2.5 dB/–8 dB)	>128 dB	16.0 dBA	78.0 dB
Sennheiser	MKE 44P	permanently charged, fixed-backplate ("electret") condensers	stereo XY (fixed 90 degrees)	40 Hz–20 kHz (+3 dB/–12 dB)	101 dB	25 dB	>58.0 dB

of the room, and had no problem handling transient spikes from the flamenco passages, the spectral picture was neither true nor particularly flattering to the instrument. Metzler described the sound as "flat," "unnatural," and "kind of metallic sounding."

MBHO MBNM 622 E PZ. The MBHO MBNM 622 E PZ was a distinctive voice in the group, typically capturing a warmer, mellower sound than the other mics, with a strong emphasis on lows and low mids. The high end was generally subdued (which helped in rounding out some brittleness in Klippert's Kay archtop guitar) and sometimes slightly "brittle" sounding.

The 622's sonic presentation is influenced to a surprising degree by the boundary you attach it to. I experimented with several different boundaries, including a large pizza box, a piece of carpet, various sizes of plywood and hardwood boards, and several different walls and floors. The differing sounds I got indicate that the 622's response varies considerably depending on size and composition of the boundary, as well as on overall room sound.

For the large-room recording of Klippert's band, we started with the 622 positioned on the hardwood floor just in front of Klippert (guitar, vocals) and Noertker (upright bass). This produced warm, rich bass, only too much of it. We got a more balanced sound by mounting the mic on a piece of plywood held at around waist height by a sheet-music stand. This tamed the low end to manageable proportions and helped bring out the highs. The resulting tracks were still dark sounding compared with the others, and not particularly accurate—the marimba sounded like a piano in some passages. Still, I and others found the sound interesting and compelling—the 622's "take" on sounds is different from that of the other mics.

The 622 was especially flattering to lower-register instruments—drums, bass, the trombone, and bari sax. Stereo localization was generally good, but like the SASS-P the 622 distributes sounds very broadly around the stereo field, to the point that you can't quite trust their positions. In addition, high-frequency reflections in the large room sometimes "jumped over" the Jecklin disk, muddling timing cues and confounding localization. That happened during a loud passage played by the marimba, which was positioned off to one side of

MAXIMUM SOUND-PRESSURE LEVEL	LOW-FREQUENCY ROLLOFF	POWER REQUIREMENTS	CABLE	DIMENSIONS	WEIGHT	PRICE
126 dB SPL	150 Hz, 6 dB/octave	5–52V phantom; AA battery	16.5' shielded 5-pin XLRF to dual 3-pin XLRM	8.43" (L) × 2.44" (D)	8.50 oz.	$525
128 dB SPL	100 Hz, 6 dB/octave	8–52V phantom; AA battery	8.8' shielded 5-pin XLRM to dual XLR	8.66" (L) × 2.01" (D)	7.23 oz.	$409
148 dB SPL (for 3% THD @ 1 kHz)	100 Hz, 12 dB/octave	12–48V phantom; (2) 9V batteries	n/a	11.50" (W) × 5.28" (H)	1.06 lb.	$995
130 dB SPL (for 0.5% THD @ 1kHz)	n/a	16–48V phantom	n/a	6.00" (H) × 12.00" (D)	1.13 lb.	$572
143 dB SPL	n/a	48V, 24V, or 12V phantom; 9V battery	11.5' shielded 5-pin XLRM to dual XLR; 10.0' shielded 5-pin XLRM to stereo miniplug	9.13" (L) × 1.26" (D)	1.06 lb.	$899
126 dB SPL	150 Hz; 250 Hz	12–48V phantom; AA battery	3.8" shielded 5-pin XLRM to dual XLR	7.50" (L) × 1.60" (D)	4.41 oz.	$795

the room; the reflections evidently rivaled the direct sound, because the marimba sounded as if it were being played on both sides of the room.

In my studio I wanted to mount the 622 on the ceiling, directly over the drums, to take advantage of the uninterrupted surface. But that would have required drilling another hole or two in the mic base, so instead I tacked the mic to the wall behind the drum kit. My studio is acoustically treated and fairly dead, and the 622's sound "reflected" that fact: the resulting drum sound was very dark and punchy, with full low mids and fat lows. Actually, the sound was unlike anything I've gotten in my studio before—the 622 gave an incredible roundness and fullness to the kick and toms. Just for fun (on a different drum session), I put up the 622 as a supplementary mic to a pair of condensers I often use as overheads. After getting the drum balance between the condenser overheads and other mics (a dynamic each on kick and snare), I sneaked in the MBNM 622 tracks beneath the others. This worked wonders to fatten up the toms and kick drum, and to my surprise didn't introduce any phase weirdness (though maybe I just got lucky).

To record the classical guitar, I tacked the 622 to a 1-inch-thick, highly polished maple board (see **Fig. 16**). The mic was not very flattering in this application, however—the tracks were simply too dark and bassy to be of much use. Metzler, who proved a stickler for tone, put the 622 at the bottom of her list.

Though the 622 never produced a very balanced or natural sound in any of the tests, it usually did bring something new and different to the table. For that reason, I came to think of it more as an "effect" mic than as a documenter of reality. I can see the 622 being very useful as a supplemental, "creative" transducer in a variety of studio applications—it kept on surprising me with its unique voice.

Røde NT4. Though not the favored performer in every case, the Røde NT4 proved the most versatile and consistently professional-sounding stereo mic throughout the range of applications—not surprising given its design, components, and higher cost. It also had the hottest output of the bunch.


BRIAN KNAVE

FIG. 16: After experimenting with different boundaries, I settled on using a polished maple board to mount the MBNM 622 stereo mic for recording the classical guitar.

The NT4's frequency-response plot shows a smooth but somewhat "bumpy flat" (plus or minus 3 dB) response from 20 Hz to 12 kHz, with gentle boosts between 100 and 200 Hz, others at 2 and 3.5 kHz, and the broadest between 5 and 8 kHz, with a slight dip between 4 and 5 kHz. A steeper peak gathers around 12 kHz, after which the high end rolls off smoothly to 8 dB down at 20 kHz. This mirrors the NT4's overall sound—bright, present, and immediately likable, but with a high-mid emphasis that can lead to harshness. In terms of frequency balance, the NT4 sounded similar to the beyerdynamic MCE 82, only warmer, fuller, and better focused.

Excepting the benchmark Schoeps mic, the NT4 was Metzler's and my favorite on her guitar. She described the sound as having a "nice balance" that was "almost there." However, we both noted the mic's excesses in the 5 kHz region, and Metzler thought that overall the bass and treble ranges seemed not well integrated—"too separated" was how she put it.

The NT4 proved very good as a drum overhead, scoring highly in angular fidelity and realism. Transient response was excellent, resulting in punchy toms and definitive stick-to-cymbal sounds. Interestingly, I noticed a slight amount of natural compression on very hard hits. Fortunately, the natural compression sounded favorable—more an enhancement than a liability. Frequency-wise, the NT4 came up a tad short on bass and low mids (though not as short as the beyerdynamic mic), and the high mids were consistently forward and sometimes brash—the small China cymbal was still a bit much. Generally, the balance of sounds benefited from minor cuts at 5 and 12 kHz and boosts at 75, 100, and/or 200 Hz.

The NT4 produced a nice, believable stereo image of Klippert's band. Crews was impressed by the clarity of the low end; Klippert, though, felt that the bass got a bit "lost" under the horns, which were well represented. La Velle remarked that the NT4 had "more body" than the other XY mics. Also, the NT4 evidenced better SPL handling than the other XYs—it didn't flinch or falter, as some of the others did, during a passage in which Klippert's voice got extremely loud. Although Klippert preferred the SASS-P and MBNM 622 on his group, the NT4 was his primary pick of the XY mics.

EMWEB CLIPS

Sennheiser MKE 44P. The Sennheiser MKE 44P is a curious case, and I'm still trying to figure out what its designers had in mind. The main issue is the mic's low-end response, which unfortunately is very weak. Even according to the frequency-response plot, the 44P's low end starts rolling off at around 300 Hz, dropping to 12 dB down at 40 Hz—and that's with neither of the two low-cut filters engaged. My question is, what's the point of having not just one but two low-cut filters on a mic with essentially no lows?

Needless to say, I found no reason to engage either filter on the 44P. Still, I regret to report, all the tracks I recorded with the mic registered precisely the deficiency revealed by the mic's response plot. That is, they all lacked bass and sounded thin—whether compared to the other mics or not. That's the problem when there aren't enough lows: even if the high-end response is good, the overall sound will be thin. High and low exist on a continuum, after all; what we hear first and foremost from a source (or mic) is its "overall" sound. It takes trained ears to mentally break up that overall sound into distinct bands and then analyze them individually. Of course, that's just what musicians and recording engineers do.

Metzler knew her guitar sound intimately, and she rightly pointed out that the 44P was quite true to the sound of the nylon strings in the high-mid range; unlike several of the other mics, it didn't impart a metallic quality to them. In addition, the 44P's transient response was very good, and the mic never overloaded on forceful passages. It was for these reasons, I suspect, that Metzler chose the 44P as her second favorite mic of the test bunch, after the Røde NT4. She just couldn't deal with any amount of metallic sound imparted to her beloved guitar. To my ear, however, the missing lows seriously compromised the guitar's overall sound.

I detected a similar high-mid "band" of accuracy on the drums. And again, the 44P's transient response was very accurate—in my notes I scribbled "dynamic truth." But unfortunately, none of that saved the mic's overall sound. What are drums, after all, without lows? Moreover, in contrast to those truthful high mids, the 44P boosts other, higher frequencies (10 and 15 kHz) to sometimes painful effect—I could hardly listen to the drum tracks from start to finish, even though I had played dark, hand-hammered cymbals using wood-tip sticks.

The 44P's performance in the big room was better, evidently because there was so much more going on frequency-wise that the rolled-off low end wasn't so apparent. Still, the sound was nothing to salivate over, and all it took was a few seconds listening to any of the other tracks to be reminded of what was missing. I really do wonder what the designers of this microphone were thinking. The only applications I can think of that the 44P would be suitable for are interviews, boardroom meetings, and some types of ENG. Then again, it might prove ideal for recording outdoors on windy days—the rolled-off bass response should make it much less susceptible to wind noise.

To its credit, the MKE 44P is a sleek, lightweight, stealthy little mic that nicely fits the hand and keeps a low profile. If Sennheiser were to firm up its bass response and tone down the piercing highs, it could be a contender.

HEARING HEADS

The testing done for this review was not meant to be conclusive; more time spent with each mic would undoubtedly have yielded more insights into its pros and cons. But given the likely range of uses **EM** readers would have for stereo mics, the tests provide fair and helpful analyses.

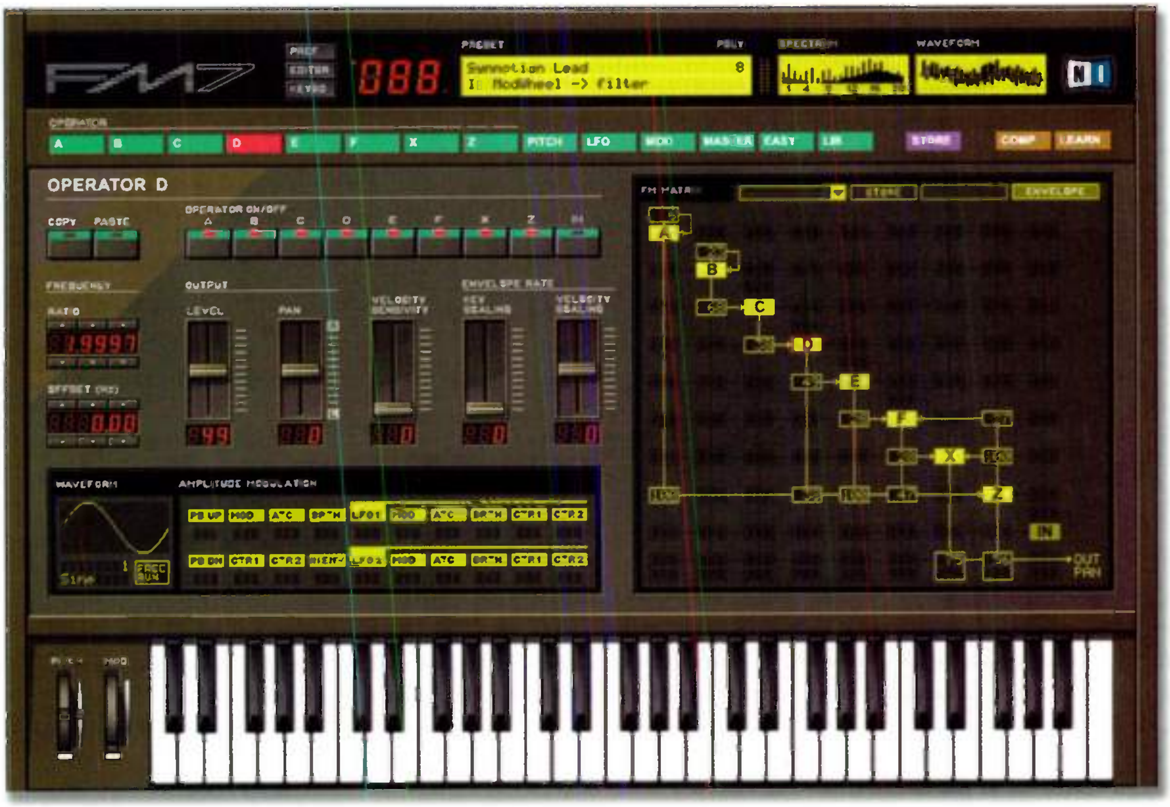
If anything is clear from the results, it's that the pickings are good among affordable stereo mics. Whether your focus is studio recording, fieldwork, concert taping, or some combination of the three, a thousand bucks opens the door to a diverse range of stereo mics, one of which should fit your needs.

I hope the information in this article will help guide prospective stereo-mic buyers through the maze of stereo-miking considerations. I also hope it turns more people on to the simple pleasures and rewards of recording with stereo microphones. There's magic in the act of human hearing, and by mechanically embodying and extending that magic, stereo mics do more than the sum of their parts might suggest.

Senior Associate Editor **Brian Knave** *thanks Scott Boland, Eli Crews, Maresa Danielsen, Wes Dooley, Sean Green, André La Velle, Gary Platt, Brittany Riddell, Sheri Seybold, Karen Wertman, and all the musicians who contributed their time and talents.*

We welcome your feedback. E-mail us at emeditorial@primediabusiness.com.

Smooth Operators



Mastering the art of FM synthesis with FM7.

By Jim Aikin

If you've been paying attention to the trends in synthesizer design for the past few years, you've probably noticed some curious twists and turns. Analog synths, which were declared dead in the mid-'80s, are making a decisive comeback. And yet, strict analog isn't enough to satisfy many of the musicians who are gravitating to analog-style instruments. Many of the new analog-flavored synths—the digital ones, at least—are tricked out with a knob or two for adding more types of tone color with frequency modulation (FM).

That's ironic, because it was FM that killed analog synths the first time around. When Yamaha's DX7 burst onto the scene in 1983, its lively, detailed sound rang the death knell for a whole generation of analog keyboards.

After a few years, though, listeners wearied of the FM sound, which can be a bit squeaky-clean. The fashion wheel turned again and sample playback synths became king. But FM never entirely slid from view. Yamaha continues to build FM synths (its current model is the DX200 groove box), and numerous other manufacturers, from Access to Propellerhead, have added FM to their hardware and software synths' timbral resources.

FM keeps bouncing back because it's such a useful synthesis technique. It's a fast, efficient way to generate an enormous variety of expressive tones. And with a little roughing up from a modern distortion effect, it doesn't have to sound squeaky-clean at all.

Smooth Operators

Native Instruments' FM7 is the ultimate FM synthesizer. On one hand, it's closely based on the design of the DX7 and can load and play preset data from the DX and other vintage Yamaha FM synths. (Dozens of banks of DX7 presets are available for free online. A good place to start your search is at www.synthzone.com/yamaha.htm.) Yet FM7 goes much further than the DX7 thanks to its dual filters, saturation stage, and flexible signal routing. Also, the DX7 was not the easiest synth to program, with its single data-entry slider and skimpy two-line LCD. Programming the FM7 is a breeze by comparison.

Even so, FM programming isn't entirely intuitive. Finding the right waveform for a particular sound is not as easy as picking it out of a menu. In this article, I will start by describing the basics of FM synthesis, using FM7 for our experiments. The concepts in the first part of the article apply, with only minor tweaking, to any synth that features FM synthesis. The latter part of the article will cover advanced concepts that are specific to FM7, such as looping envelopes and modulation routings.

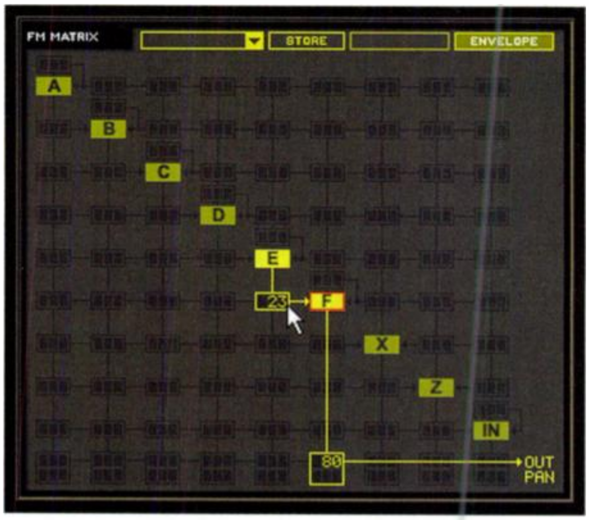


FIG. 2: To add frequency modulation (FM) in the Matrix display, simply click in the box shown and drag the mouse upward.

THE BIG IDEA

The musical applications of digital FM synthesis were first developed in the early '70s by Dr. John Chowning at Stanford University. At that time, digital synthesis was performed on large, slow mainframe computers. One second of audio could take many minutes to compute. Chowning explored the fact that frequency modulation not only gave him a wide variety of tone colors but allowed him to compute the desired waveforms much more quickly than was possible with other types of synthesis, such as additive synthesis.

Even the most hidebound classical musicians are familiar with frequency modulation: it's nothing but our old friend vibrato. In vibrato, the frequency of a tone increases and decreases—the pitch goes up and down—in a repeating, periodic way. You can hear this effect on a synthesizer simply by modulating the frequency of an oscillator using a repeating LFO waveform.

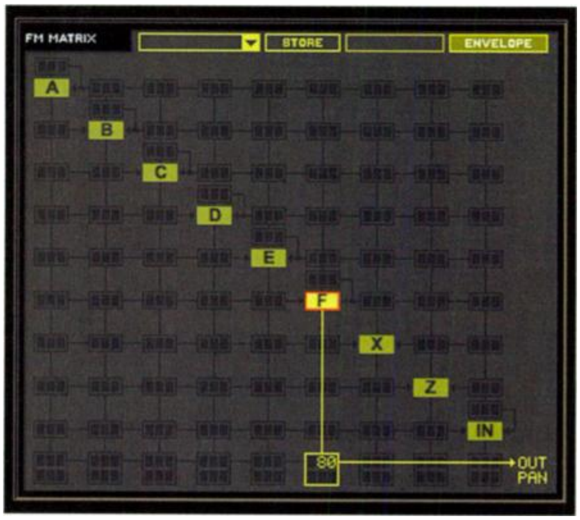


FIG. 1: FM7's Matrix display is used to route audio signals among the operators. Immediately after a patch has been initialized, it looks like this.

Chowning discovered that when the modulating wave is fast enough, our ears can no longer hear the discrete rising and falling cycles of the vibrato. Instead, we perceive the changes in the pitch of the main oscillator (called a *carrier*) as changes in tone color. When the modulating wave (called a *modulator*) is faster than 20 Hz, new overtones are added to the sound of the carrier. The strength of the overtones will depend on the amplitude of the modulator: stronger modulation means more overtones. And the frequencies of the newly generated overtones will depend on how the carrier and modulator are tuned with respect to one another.

Confused yet? Don't be. Launch FM7 and follow the exercises below. Before you know it, you'll be an FM whiz.

FM NATION

Begin by clicking on the library (Lib) button. Before going any further, click on the Save All button in the upper-right corner of the Library display and save your bank of presets under a new name.

Now, choose a preset and click on the Init Edit Buffer button. When you play the keyboard, you'll hear a simple sine wave tone—the raw ingredient of FM synthesis. Although pure sine waves have no overtones, they are good for a few things, such as adding suboctaves and high, ringing overtones. For most purposes, however, a sine wave will sound more interesting when we mess with the tone a little.

Now click on the green F button in the top row. If the right half of the window (as shown on p. 75) doesn't look like **Fig. 1**, click on the Matrix button, located near the upper-right corner. The Matrix display is your window into a vast world of FM programming.

Smooth Operators

Right-click (Windows) or control-click (Mac) on the letter E near the middle of the matrix. It will turn bright yellow.

Left-click (Windows) or click (Mac) in the black field just below E and directly to the left of the F. While holding a key on your MIDI keyboard, and without releasing the mouse button, drag the mouse upward. Your screen should look like **Fig. 2**. As the numerical value in this field (which is called the FM index) increases, the tone will get progressively brighter and more piercing. As you drag the mouse up and down, you will get a wah-wah kind of sound. Just about everything you do in programming FM7 will be a variation on this simple mouse move.

What you are doing in the above example is listening to the oscillator in F as the waveform of E's oscillator is modulating F's frequency. But because the frequency of E's oscillator is greater than 20 Hz, the frequency modulation causes changes in the tone color of F. (In the Init patch, both oscillators are tuned to the same frequency.) As the output of E increases in amplitude, the waveform of oscillator F acquires more and more overtones. The Waveform display at the upper-right corner of the FM7 panel shows a rough picture of what's happening to the sound.

OPERATORS ARE STANDING BY

If FM synthesis were limited to simple wah-wah sounds, it wouldn't be used very often. The power of FM synthesis lies in combining signals in various ways to create complex musical tones.

The signals come from *operators*. In Fig. 1, the letters A through F represent six individual operators. Each operator has two basic parts—an oscillator and an envelope generator. The oscillator creates a raw waveform, and the envelope generator shapes the loudness of the waveform over time. The following exercise will demonstrate how this works.

Start with the Init program and right-click (Windows) or control-click (Mac) on E, which switches operator E on. Raise the FM index in the box directly below E and to the left of F (see Fig. 2) to 40. When you play the keyboard, you'll hear a somewhat nasal tone.

Next, left-click (Windows) or click (Mac) on the E in the matrix, or click on the green E button in the top row (It will turn red). That will bring the controls for operator E into the main part of the window.

Left-click (Windows) or click (Mac) on Envelope in the upper-right corner. (From now on, I will use the word *click* on its own to mean left-click in Windows and a simple mouse-click on the Mac.) That will switch the display from Matrix to Envelope and Keyscaling screen for operator E (see **Fig. 3**). Now, click on the square handle in the upper center of the Envelope display and drag it straight

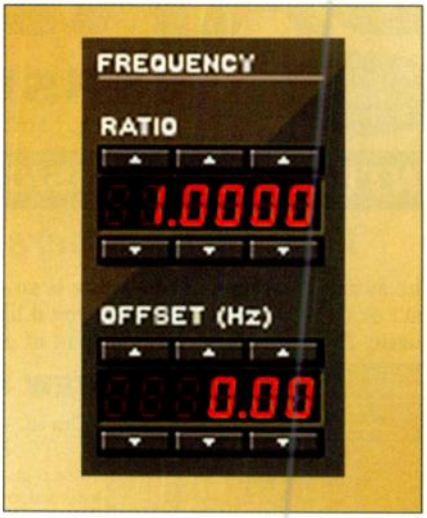


FIG. 4: This shows an operator's frequency controls.

down. That will reduce the envelope's sustain level to zero. When you play the keyboard, you should hear a tone with a pleasant knocking sound as its attack, followed by a sustain made up of the original sine wave. The shape of the attack will be visible in the Envelope display. If the entire tone fades out, or if the nasal tone sustains without changing, it's because you skipped a step above and you are editing the envelope of the wrong operator.

In this example, the signal from operator E starts at a high level, but quickly fades out because its amplitude is being shaped by the envelope generator. As E fades out, the waveform of F (which is what you're hearing) changes. Overtones are being stripped out of the tone, as if the sound of operator F were passing through a lowpass filter. Although FM7 has filters, we're not using them yet. This filterlike effect is accomplished by simply controlling the amplitude of operator E, which is the modulator.

Switch back to the Matrix display and take a look at all of those little black fields in the matrix. Each of them controls a signal routed between two operators. So far, I've been using operator E to modulate operator F. But by switching any operator on and then dragging in the field that connects it to another operator, you can easily create tonal changes.

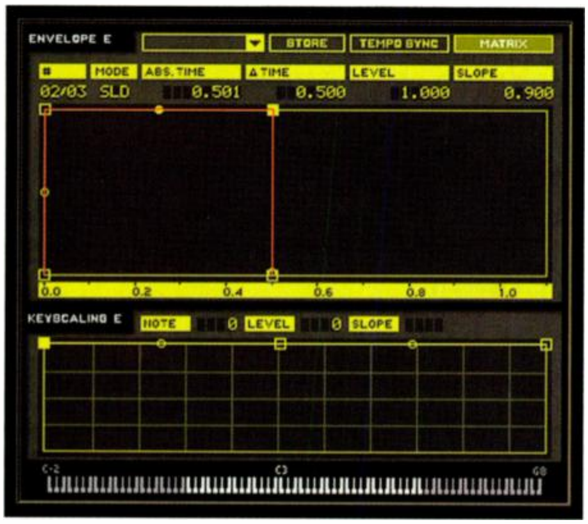


FIG. 3: This is the Envelope and Keyscaling display for an FM7 operator. The envelope is in the upper part of the display.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Smooth Operators

GOLDEN RULES

There are two rules to remember when editing in FM7. First, an operator has to be switched on—illuminated bright yellow—for it to have any effect. Second, the row of fields along the bottom of the matrix connects the operators to the output. If an operator isn't connected to the output—either directly or because it's modulating another operator that's connected to the output—you won't hear it.

In the original DX7, you could choose from a set of 32 predetermined methods of connecting operators to one another and to the output. These were called *algorithms*. In FM7, the algorithms are entirely user-configurable using the Matrix display.

FREQUENCY RATIOS

As mentioned earlier, the nature of the harmonic spectrum coming from the carrier operator in FM depends on the relationship between the frequencies of the carrier and the modulator. The following exercise demonstrates how that works.

Grab the envelope sustain handle for operator E, which you lowered in the previous exercise, and pull it back up to the top. This time, we want the modulator to sustain at its full level.

Switch to the Matrix display and make sure the FM index below E is set to at least 40. A setting of 50 might be even better.

Locate the frequency controls on the left side of the panel (see **Fig. 4**), and make sure the E button in the top row is still red. You're going to be changing the frequency of operator E.

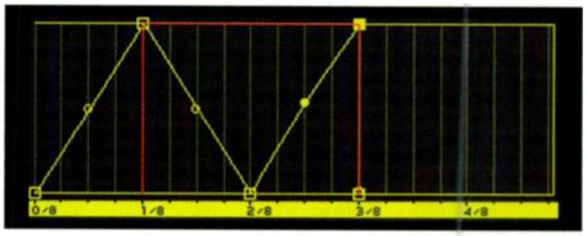

FIG. 6: This is an example of a basic looping envelope.

Click once on the tiny up-arrow button directly below the word "Ratio." You can do that while holding down a key on your keyboard if you like. The numeric display below the arrow will be 2, and the tone will change suddenly. If you're familiar with analog waveforms, you'll probably be able to identify the new tone as being similar to a square wave.

Switch back and forth between 1 and 2 a few times, using the up and down arrows at the left side of the Ratio display. Then try increasing the ratio still further, to 3, 4, 5, and so on. You'll hear progressively thinner but brighter tone colors.

For practical purposes, it's best not to go higher than 10 or 11 with the Ratio control. When you play in the upper range of the keyboard, higher ratios will produce *aliasing*, a clangorous effect that will interfere with your ability to play chords and melodies.

Aliasing has its uses, but FM7 also has tools for taming it. If you want to use a high ratio for a modulator and you need to play upper notes on the keyboard, use the Keyscaling display (below the envelope) to scale down the amount of the high-pitched modulator at the upper end of the keyboard. You can edit Keyscaling the same way you edit Envelope: click on the square handle at the right side of the display and drag it downward.

Now, set the Ratio of operator E back to 1 or 2 and click on the up or down arrow in the middle of the Ratio display. The tone will start to beat.

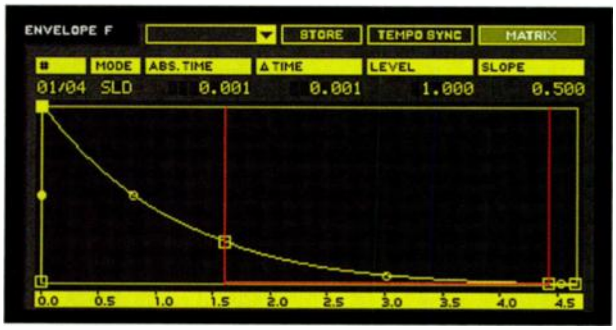

FIG. 5: This is an example of an envelope used to re-create the sound of an electric piano.

One of the operators is now being detuned from the other. The beating effect is similar to what happens in any other synth when you detune one of the oscillators—the tone gets richer.

As you play up the keyboard, you'll hear the detuning get progressively faster and the sound becoming twittery. To avoid that, you can use the Offset parameter for detuning instead of the ratio. The Offset parameter detunes an operator by a constant frequency. Offset values between 1 and 1.5 are good for a rich chorusing effect.

Try turning down the ratio value for operator E to 0.0000. At this point, the modulator will have a fixed frequency of 0 Hz, so it won't affect the tone of the carrier at all. As you turn up the offset to 3.00 or 4.00, however, you'll start to hear vibrato. Increase the offset value gradually while holding down a key, and you'll hear the phenomenon described at the beginning of the article: when the frequency of the modulator reaches 20 Hz, the vibrato will meld into a tone color.

Reset the frequency parameters of operator E to a ratio of 1.00 with no offset, and use the up and down arrows beside the Ratio display to change the frequency of operator F (the carrier). Listen carefully to the variety of tone colors that are available.

Try setting the Ratio parameter of operator F (the carrier) to 0.00 and its offset to 1.00 or thereabouts. When the modulator is set to a low value you will get vibrato, but when the carrier is set to a low value the result sounds like chorusing. This technique is useful for creating rich FM sounds.

THAT DARN DX EP

In this exercise, we'll take some of the techniques we've learned so far, add a

Smooth Operators

few new ones, and create a fair approximation of a classic FM sound—the DX7 version of the Rhodes electric piano. Start with the Init patch, as before. Edit the envelope of operator F so that it looks about like **Fig. 5**. That operator will make the main body of the piano tone.

Switch on operator E, then route it to operator F using the Matrix display and give it an FM index of 33. Tune operator E's frequency ratio to 10.00. This will create the bell-like tone.

Give operator E an envelope similar to that of operator F, but somewhat shorter. The bell-like overtone should die away while the main body of the tone is still audible.

Set operator F's Velocity Sensitivity slider to 50, and the same slider for operator E to 75. That will cause low-velocity notes to be softer and somewhat less bright.

Set operator F's Envelope Rate Velocity Scaling to 30, and the same slider for operator E to 50. At this point, you may want to go back and fiddle with the envelopes for these two operators to get a better approximation of the shape of an electric-piano tone. If your only experience with FM electric pianos has been with sampled ones, you'll hear something new and gratifying: an FM electric piano responds smoothly and evenly to MIDI Velocity data, in a way that not even the most expensive sampled electric piano on the market can duplicate.

Our piano still sounds a little wimpy, so let's add another color to the mix. Switch on operator D. In the matrix, route it to operator E, with a level of 35. Give operator D an extremely short envelope. The envelope should have an instant attack and then fall to zero in 0.02 seconds or less. In order to do this, you'll need to right-click (Windows) or control-click (Mac) on the Time ruler below the envelope and drag to the left in order to zoom in.

Next, tune operator D to some unusual ratio, such as 3.3980. This is the point at which FM programming becomes an art. The idea is to use operator D to imitate the sound of an electric piano's hammers striking the tines. It should make a brief percussive *thwock*. If it makes a tone with a perceptible pitch, the envelope is too long. Now, set operator D's Velocity Sensitivity slider to about 75.

To test whether operator D is doing its job, listen to a few notes with it switched on and then with it switched off. You can switch it on and off in the Matrix display or with the Operator On/Off button. You shouldn't be able to hear operator D as a separate element in the sound, but you should notice when it is missing, because without it the tone should sound a little too clean. (For more on this technique, see the sidebar, "Dissecting FM.")

There are many ways to refine this program. A master FM programmer might even set up a real Rhodes and compare its tone with the FM sound one note at a time to get the envelopes and operator tunings just right.

You can use the general principles in this exercise for creating many types of FM sounds, not just that of electric pianos. The idea is to use different operators to craft different parts of a complex sound. In some FM patches, there may be only one carrier, and the primary modulator might be modulated by a second modulator. In other patches, the six operators might be configured as three carrier/modulator pairs. In still others,

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DISSECTING FM

The key to understanding FM patches created by other people is muting and unmuting the operators one at a time to hear how each contributes to the sound. By studying the Matrix display, you'll be able to see which operators are carriers and which are modulators. The carriers are connected to the Out in the right-hand column, as shown in the factory patch called Six-String (see **Fig. A**). If there's more than one carrier, start by switching off all but one of the carriers, so you can hear its contribution to the sound. Then switch its modulator(s) on and off, one by one.

In Six-String, operators C and F are carriers. Each of them has two modulators in a stack configuration: A modulates B, which in turn modulates C. The same thing happens with D, E, and F. When you mute F you'll hear only the bright portion of the sound, which is coming from C. When you mute C and unmute F, you'll hear that F is providing the dark body of the tone.

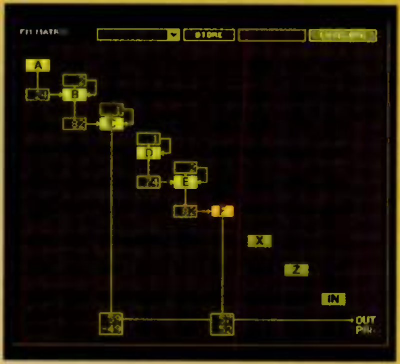

FIG. A: The Matrix display shows the algorithm for the Six-String patch. The parameter fields just above the operators allow each operator to modulate itself.

While listening to operator C, mute and unmute A. You'll hear that it's adding a lot to the brightness of the tone. Its frequency ratio is set to 20.0063, which is quite high. Try lowering this to 19, 18, and so on, and listen to the changes in the tone. Tuning each operator correctly is essential to creating a rich and lifelike FM tone.


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three or four carriers tuned to different frequencies might all be modulated by a single modulator. The possibilities are endless.

PUSHING THE ENVELOPE

FM7 provides an Easy page, which offers traditional synthesis parameters, such as ADSR envelopes for amplitude and timbre. However, if you confine your editing to those parameters, you'll miss out on much of the power FM7 has to offer. Here's a quick tour of the envelope section that highlights a few possibilities that you may not be aware of. For full details on FM7's envelopes, consult the owner's manual.

The DX7's envelopes had four level settings and four rate controls. But unlike a traditional ADSR, these envelopes didn't have to start and end with a level of zero, and the attack peak could be higher or lower than the sustain level. FM7's envelopes preserve that design, with one small exception: carrier envelopes always shut off, so you won't create stuck notes. FM7 envelopes also have more rate/level pairs (up to 31 per envelope), the ability to loop, and the ability to sync to a specific tempo. Let's begin by exploring the animated rhythms that you can create with looping envelopes.

Initialize the edit buffer, as before. Turn on operator E and set it to modulate operator F in the matrix. This time, set the modulation level at 60.

Select operator E for editing. If its Envelope display isn't visible, click on the Envelope button, which is in the upper-right corner.

Click on the Tempo Sync button. That will cause a grid of vertical lines to appear in the Envelope display, and the Time ruler below the display will switch from showing seconds to showing beat values.

Try dragging the square handle in the upper-left corner of the envelope to the right. That will increase the attack time of operator E, giving you a wah sound. Notice that the handle snaps to the grid as you move it. Park this handle on the line above the 1/8 marker.

Next, right-click (Windows) or control-click (Mac) on the red line along the top of the envelope. That will create a new handle. Drag this handle down to the bottom of the display and position it above the 2/8 marker. If the Mode parameter is set to Fix, you may need to click on the setting to switch it to SLD (slide), to create enough room.

Drag the handle at the upper-right corner of the envelope left or right as needed, so that it's positioned above the 3/8 marker. Click on the round handle in the middle of the right-hand envelope segment, and drag it downward so that the segment is linear rather than curved. At this point, your envelope should look like **Fig. 6**.

When you play the keyboard, you'll hear a repeating wah-wah sound. That is because the envelope for operator E is looping. The red lines indicate the start and end of the loop. So far, it's not a very interesting sound, but we're not finished yet.

Edit operator E's envelope so that it looks like **Fig. 7**. If you set it up correctly, you'll hear a catchy rhythm that loops every two beats.

Now activate operator D. Use it as a second modulator for operator F, as in the electric piano voice you created earlier. Give it a different rhythmic envelope, perhaps one that loops every five 16th-notes rather than every four eighth-notes, so that the two envelopes combine to make a longer rhythm. (If they also drift out of sync with one another, you need to download the version 1.1 upgrade of FM7 from the Native Instruments Web site.) Finally, tune operator E's ratio to 1.5 and operator D's ratio to 3.5. That will give the tone a fat suboctave growl.

Again, there are many other tricks that you can try with this patch. For instance, try running operator F through the filter, adding a rhythmic filter envelope, and panning the outputs of F and the filter slightly apart. You may have to experiment for a while to find combinations of envelopes that give you rhythms you like, but there's no shortage of possibilities to explore.

TONE-SHAPING TRICKS

Most synthesizers let you control the filter cutoff frequency from the mod wheel or joystick. FM7 takes the concept much further. The output level of any operator can be controlled by the mod wheel, Pitch Bend, Aftertouch, Breath Control (Control Change 2), or either of FM7's two assigned MIDI Control Change types.

FIG. 8: This setting controls the level of the operator from the mod wheel.

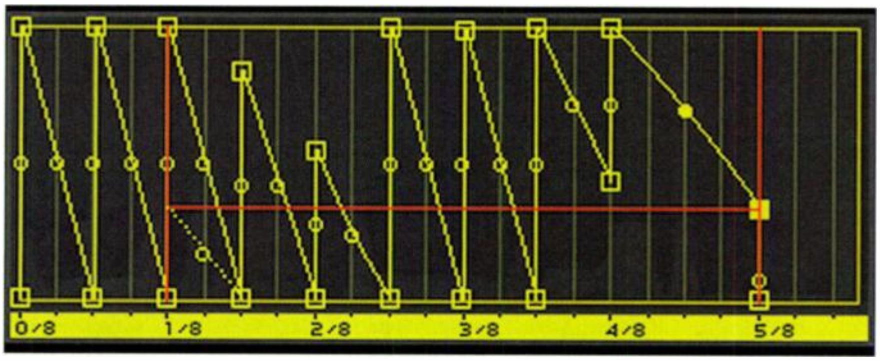

FIG. 7: This envelope will generate a more interesting rhythm.

If an operator that's acting as a modulator is tuned below the carrier, you can use the mod wheel to bring in a suboctave. Because the modulator has its own envelope, the mod wheel can be assigned to add an attack transient to certain notes in a phrase. And because the carrier's level can also be controlled, you can easily set up a patch in which the mod wheel or another controller crossfades between two completely different tones.

To try out this concept, start with the rhythmic patch you created in the previous exercise. Select operator E for editing. In the Amplitude Modulation strip along the lower left (see **Fig. 8**), click in the data field below Mod and drag the mouse up. A setting of 80 or 85 will work well.

The mod wheel is already set up to create vibrato in the Init patch. For this example, we don't want that, so click on the green Mod button in the upper row. In the lower center of the large signal-routing matrix, you'll see a setting of 20 that routes the mod wheel (in the top row) to pitch (in the right-hand column). Turn this setting down to zero.

Now play the keyboard. Your patch should sound much more subdued than before, because FM7 has automatically lowered the level of operator E to give the mod wheel some room to maneuver. As you're holding a note, push the mod wheel up. The sound will become more aggressive as more of operator E's output is sent to the carrier.

The power of FM synthesis lies in combining signals in various ways to create complex musical tones.

FMADNESS

In this article I've only scratched the surface of FM7 programming. Other areas you can explore include the choice of waveforms for each operator (using the Waveform selector shown in **Fig. 8**), the filter and saturator, the built-in delay/reverb effect, the audio input, and operator feedback, not to mention the many ways the six operators can be configured in different algorithms. If you're willing to take the time to learn FM7's tricks, you'll find yourself the master of a rich and endlessly creative sound palette.

Jim Aikin *writes about music technology for a variety of publications. His book* Software Synthesizers *is scheduled for publication in mid-2003.*

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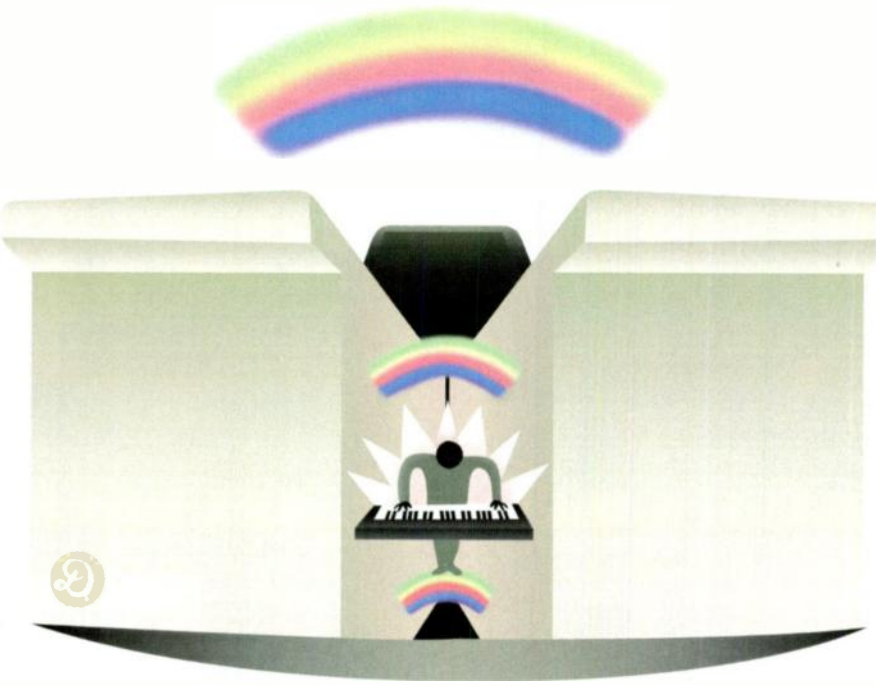
DESKTOP MUSICIAN

Playing in the Cracks

Explore the world of alternate tunings with software synths.

By Jim Aikin

Computers offer musicians such an endless smorgasbord of resources that it's easy to overlook some fascinating possibilities. One topic that doesn't get talked about much is using a computer to explore alternate tuning systems. In this column I'll discuss some of the software tools that you can use to make music in alternate tunings. I'll also explain a few of the techniques that I've developed, and I'll point you to Web sites where you can learn more.



JACK DESROCHER

TUNING UP

First, it's important to understand the phrase *alternate tunings.* Most musicians today take our 12-note-per-octave, equal-tempered (evenly divided) scale for granted. But there's nothing natural or inevitable about such a system; it's a very effective compromise tuning system, or temperament, that became popular in Europe during the 18th century. Any system other than this temperament qualifies as an alternate tuning. In other words, some or all of the notes in an alternate tuning will fall "in the cracks" between the keys on a conventional keyboard.

When people hear that description, they sometimes nod sagely and say, "Oh, you mean quarter-tones!" Actually, quarter-tone tuning, in which each equal-tempered half step is split into two equal parts, is one of the least interesting alternate tunings. Its frequency spectrum isn't inherently stepped, or quantized, at all. Rather, it's a continuous rainbow of pitches. A pitch that falls between two of the keys on the equal-tempered keyboard can be tuned to any frequency value at all, not just to an equal-tempered quarter-tone.

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You can play notes that are "in the cracks" using a Pitch Bend wheel. If you're a guitarist, you can get to them by bending the strings. But those techniques can be inconvenient. What if you want to play a chord in which each note requires a different amount of Pitch Bend? You can't do it.

What makes such chords musically powerful is that their intervals have distinct colors. They can be sweet, sour, bright, or harsh, in ways that conventional 12-note equal-tempered intervals and chords can't match. Once you've heard a few of these intervals in action, you'll never perceive standard chords in the same way again. In fact, a word of warning may be in order: experimenting with alternate tunings can spoil you. You may start noticing how ugly a major triad on a piano or synthesizer can sound.

Going into a full discussion of tuning theory would take up far more space than we have available here. (Allaudin Mathieu's book *Harmonic Experience* provides a good overview, as does David B. Doty's *Just Intonation Primer*, available from the Just Intonation Network, www.justintonation.net.) Briefly, we can talk about two types of alternate tunings: *equal-tempered scales* and *just intonation*.

In an equal-tempered alternate tuning, the octave is divided into more (or fewer) than 12 steps, but the steps are still equally spaced. Tunings with 19 and 31 notes per octave are especially interesting, but it's difficult to play music in those tunings on a standard MIDI keyboard because one octave of notes will be stretched across an octave and a half or more of keys. Using MIDI sequencing is a practical option, however.

Just-intonation tunings can be played from the keyboard, because the number of scale steps per octave is up to you. What distinguishes these tunings is that the intervals are based on pure mathematical ratios. For instance, the ratio of a perfect fifth is exactly 3:2, and a major third is 5:4. So if a given note is vibrating at 100 Hz, the major triad that uses that note as a root will include a third vibrating at exactly 125 Hz and a fifth vibrating at exactly 150 Hz.

By contrast, 12-note-per-octave equal temperament is based on the 12th root of 2, which is an irrational number (it's about 1.059463). So a "perfect fifth" on a piano or a typical MIDI synth isn't perfect at all. If the fundamental is at 100 Hz, the fifth is at 149.83 Hz—just slightly flat compared with a 3:2 fifth. The major third in this temperament is noticeably sharp; what should be 125 Hz is actually 125.99 Hz.

The result: all intervals in our conventional tuning have built-in "beats" (interference patterns) between the frequencies. I've recorded a few MP3 files so you can hear the differences between equal temperament and just intonation. To learn more, a good place to start is the Just Intonation Network.

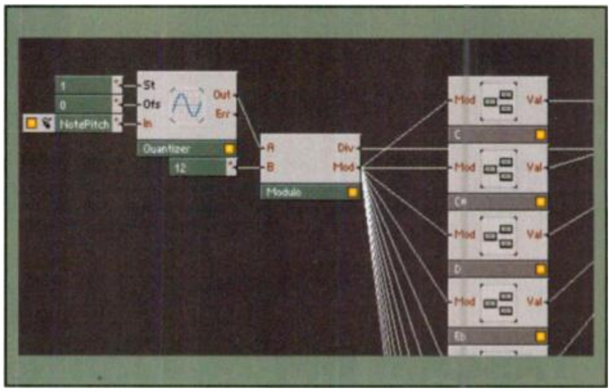

FIG. 2: To find the pitch class of a MIDI note in Reaktor, the MIDI note number (output by the NotePitch object) is divided by 12 in the Modulo object. The rest of the division is sent on to the author's stack of custom switching objects.

HARDWARE TUNE-UPS

Over the years, numerous MIDI synthesizers that supported user-programmable alternate tunings have come and gone. Yamaha's DX7II/TX802 line had especially useful tuning tables, because they allowed each key in the full MIDI range to be tuned to any pitch. In other words, they weren't limited to 12 repeating notes in every octave. The most tunable synths that I know of in the current market are the E-mu Proteus 2000 series, which include 12 full-range tuning tables.

The tuning tables in hardware synthesizers suffer from some limitations that make exploring tunings a bit awkward. The tables are almost always calibrated in fractions of an equal-tempered half step (usually 64, 100, or 128 increments per half step) rather than using raw frequencies. To get a pure interval, you have to apply a mathematical formula (more on that later) or use your ears. Even then, intervals that appear to be the same size based on the settings in the LCD may be slightly different because of the way the synth calculates its pitches.

Some of the newer synthesizers can change their tuning in response to

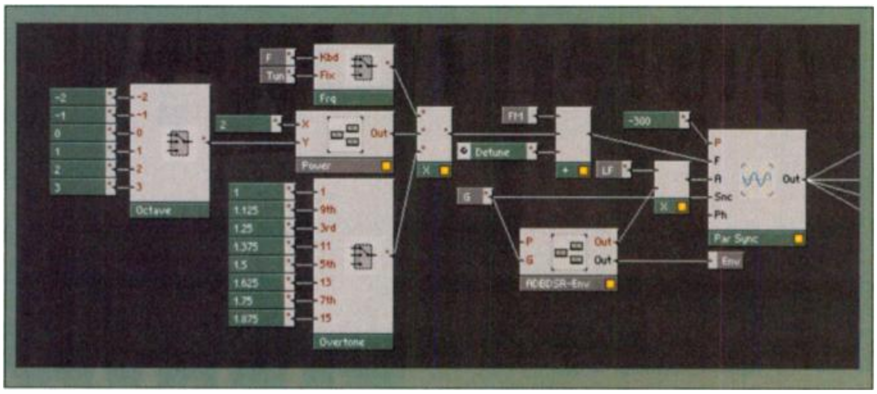

FIG. 1: Above is a portion of an FM operator in the author's just-intonation synth in Native Instruments Reaktor. The oscillator is at the far right; note the constant value of –300 attached to the P input.


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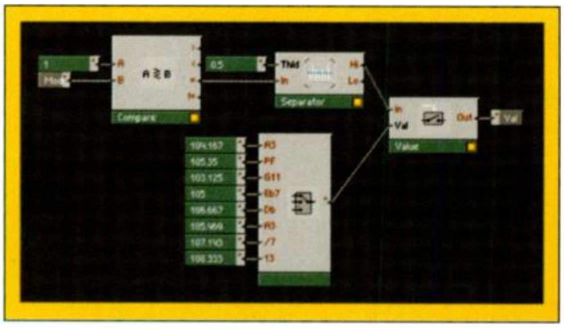

FIG. 3: Above is the macro for tuning a single key (here, C♯) in the author's Reaktor synth. Tuning values are held in the constants attached to the panel-switch object in the center of the macro.

MIDI microtuning messages. Most allow their tuning tables to be edited on the fly as part of their own System Exclusive implementation, so you can embed SysEx data in a sequence if you need more than one tuning in a given piece.

Even when a tuning table is active, however, all of the instruments I know of continue to set the depth of their response to MIDI Pitch Bend data in equal-tempered half steps. Setting up a Pitch Bend processor that "knows" about the current tuning is not especially difficult. You can do it yourself with a software-based modular synth, but the details are beyond the scope of this column.

SOFTWARE TO THE RESCUE

There are several software solutions with which you can explore alternate tunings. Cycling '74 Max/MSP, Native Instruments Reaktor, and Csound offer a great deal of flexibility, but they all require you to do some programming. Justonic Tuning's Pitch Palette software (www.justonic.com) offers a more user-friendly approach that will appeal to musicians whose interest is more casual.

Pitch Palette takes advantage of the fact that some synths allow their tuning tables to be edited on the fly by incoming MIDI data. You can create a just-intonation tuning in the software simply by typing the ratios in an on-screen keyboard and waiting a few milliseconds while the tuning is transmitted to the synth. You can then immediately start to play your MIDI keyboard. Pitch Palette even lets you retune the synth "on the fly" by dedicating a couple of octaves of your MIDI keyboard to tuning control. Sadly, most of the hardware synths that the software supports are no longer in production.

Justonic's Pitch Palette Home Studio Pack ($79) includes the Roland Virtual Sound Canvas (VSC-3) software, which eliminates the need for a compatible hardware synth. The Virtual Sound Canvas is not ASIO-compatible, however, so the performance latency won't allow you to enjoy playing music in any of the Pitch Palette's dozens of preset scales. Also, the VSC-3's tuning resolution, like that of many hardware synths, is not fine enough to remove all of the beating from supposedly pure intervals.

Csound is a free music-programming language with which you can do anything from alternate tunings to beat-mangling or designing your dream reverb. (See "Csound Comes of Age" in the July 2002 **EM**.) Unlike commercial synths, which prefer to translate MIDI note numbers directly to equal-tempered half steps, the "native language" of Csound oscillators is hertz, so creating any frequency that you need is a piece of cake. The way I prefer to create just-intonation scales in Csound involves a few simple steps:

1. In your orchestra file, define a global variable for the base frequency of your piece (the frequency on which all of the other notes will be based).

2. Instead of using one parameter field (p-field) in the score to specify the note's pitch, use three fields: one for the octave and one each for the numerator and denominator of the ratio you want to hear. (Ratios are the same as fractions).

3. In the instrument, use a little multiplication and division to arrive at the correct frequency as shown in this example:

```
ioct pow 2, p5
inum = p6
iden = p7
ifrq = gibasefrq * ioct * inum / iden
```

The values *inum* and *iden* hold, respectively, the numerator and denominator of the pitch ratio. The p-field data (p5, p6, and p7) for each note is taken from the Csound score, and the global variable *gibasefrq* is defined at the start of the orchestra file. The library function *pow* raises the first parameter (2) to the power of the second parameter (p5, the octave number).

I've tried other methods, such as storing precomputed ratios in a Csound table, but I always end up wanting to hear intervals that aren't in the table. This method lets me be impulsive (or as impulsive as you can ever be in Csound, which is not very).

REAKTOR MELTDOWN

Like other commercial synthesizers, Reaktor starts with the assumption that you want to play equal-tempered half steps. Its oscillators have a P (pitch) input that conveniently accepts MIDI note numbers. My first attempt at a tunable Reaktor synth involved adding or subtracting small decimal values to or from the MIDI note number before sending it to the oscillator.

Getting the intervals perfectly in tune by ear proved to be difficult. The following formula for computing the number of equal-tempered cents in a ratio was supplied to me by David Doty of the Just Intonation Network:

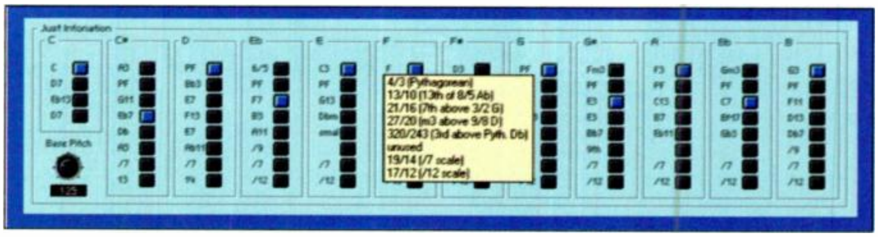

FIG. 4: The author's bank of pitch-selector switches as they show up in the Reaktor panel. He has entered text in the ToolTip pop-up for each pitch: here it's giving information on the choices for the F key.

log(ratio) × (1200/log(2))

The value of 1200/log(2) is about 3986.3. Armed with a decent pocket calculator, you can quickly figure out that log(3/2) × 3986.3 is about 701.95 cents. In other words, a perfect fifth is about 1.95 cents wider than an equal-tempered "perfect" fifth.

Rather than sweat over a calculator, you may find it easier to use the Reaktor oscillators that have F (frequency) inputs. Those inputs can accept either an FM modulator signal or a fixed data value in hertz. The trick is that you also have to attach a constant with a very low value (–300 works well) to the P input, as shown in **Fig. 1**. If you don't do that, the oscillator's F input won't be calibrated correctly.

When designing an FM instrument for just intonation, I decided to skip the normal fine-tune control for the modulator oscillators in favor of a panel switch that lets the modulator be tuned instantly to any of the basic overtones up through the 13th. The frequency sent to the oscillator is arrived at by multiplying the outputs of the Frq, Octave, and Overtone panel switches as seen in Fig. 1. The F input to the Frq switch is used when the keyboard is active; it receives a value from my bank of tuning switches. The Tun input receives a basic tuning offset value for use when the modulator is in fixed-frequency mode (ignoring the keyboard, in other words).

To play a Reaktor synth tuned to just intonation from a standard MIDI keyboard, you have to detect which *pitch class* (C, C♯, D, and so forth) is being played, look up the correct tuning ratio for that class, and then multiply it by both the base frequency and an octave factor (some power of 2). The first stage in the process, which uses Reaktor's Modulo object to divide by 12 and output the remainder, is shown in **Fig. 2**. The quantizer module seen in Fig. 2 solves a potential problem: Reaktor's NotePitch object doesn't always send out integers, but the rest of my algorithm needs to receive and process integers.

That part of the process sends its output to 12 switch mechanisms. Each switch mechanism then inspects the pitch-class number and decides whether to ignore the incoming event or process it and pass it on to the output. Whichever piece of data arrives at the output is passed on to the oscillator. The setup for the C♯ pitch-class macro is shown in **Fig. 3**, and the panel switches used in my tunable synth are shown in **Fig. 4**.

Up to eight constants are attached to each panel switch. When the user presses a switch in the bank shown in Fig. 4, the value of one of the constants is sent to the Value object in Fig. 3 and stored there. The Modulo object in Fig. 2 sends out pitch-class messages indiscriminately to all 12 of my pitch-class macros. When the Compare object in

SPREAD OUT!

Figuring out where various interval ratios in just intonation are located in a scale is not always easy: Let's see, is 7/5 higher or lower than 11/8? Microsoft Excel provides a quick solution. (You should be able to do much the same thing with any other spreadsheet program.) Set up a column of numerators and a column of denominators. In the third column, called ratio, enter a formula of the form B2/A2, dividing the item in a cell in the first column by the item in the corresponding cell in the second column. Drag this formula cell all the way down the third column; in Excel, this automatically enters B3/A3, B4/A4, and so on in the relevant cells. Finally, sort the data by the ratio column, as shown in **Fig. A**. Problem solved.

	A	B	C	D
1	numer	denom	ratio	name
2	1	1	1	C
3	16	15	1.06667	C#
4	13	12	1.08333	C#
5	9	8	1.125	D
6	7	6	1.16667	Eb
7	6	5	1.2	Eb
8	5	4	1.25	E
9	4	3	1.33333	F
10	11	8	1.375	F#
11	45	32	1.40625	F#
12	3	2	1.5	G
13	8	5	1.6	Ab
14	13	8	1.625	A
15	5	3	1.66667	A
16	27	16	1.6875	A
17	7	4	1.75	Bb
18	16	9	1.77778	Bb
19	9	5	1.8	Bb
20	15	8	1.875	B

FIG. A: Here are some useful and good-sounding tuning ratios, computed in Microsoft Excel.

Fig. 3 detects that an incoming message has the value of 1 (the C♯ pitch class), it sends a 1 from its = output to the Separator; if the incoming message has any other value, the Compare sends a 0 from this output. The Separator passes the incoming message to its Hi output if the value is greater than the value at the Thld (threshold) input, in this case 0.5. That message then triggers the Value object to send its stored data value to the Val output. The rest of the algorithm (not shown) multiplies this data by the octave value and any other needed offsets before sending it on to the oscillator in Fig. 1.

If you're not a Reaktor programmer, that may not make much sense. The essential steps to remember, which can be used just as easily in Max/MSP as in Reaktor, are as follows:

- The pitch class of the current MIDI note is found by using modulo division on the note number.
- The ratio value to be used for each pitch class has to be stored somewhere—either in a table or in a simple variable (such as Reaktor's Value object). You can create and store the value using whatever mechanism suits your needs, up to and including separate knobs or sliders for the numerator and denominator of each ratio to be computed.
- The pitch-class number triggers the reading of the corresponding stored value.
- After retrieving the correct stored value, the algorithm has to restore the octave transposition, which was stripped off when the pitch class was identified, and multiply by the base pitch of the tuning.

Reaktor owners can download my tunable version of one of the factory synths from the **EM** Web site. To hear the difference between equal temperament and a user-programmed tuning, just press the Equal Temp button.

WHY BOTHER?

The restless, edgy quality of Western music in the past 250 years is due not only to the turbulent culture in which composers have lived but to the tuning we have been using. With just intonation, music can be soothing and centered, yet with the aid of a computer, you still have the freedom to modulate to distant keys, because the tuning can be changed on the fly.

If you're working in a more aggressive style, try running some perfect intervals through an overdrive effect. They're unbelievably solid and rich. Alternate tunings aren't for everyone, but they can be the "secret sauce" that gives your music a flavor all its own.

Jim Aikin *writes about music technology for a variety of publications, which doesn't leave him nearly enough time to play with his still-evolving Reaktor tuning synth.*

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Look Who's Talking

Teach your synthesizer to talk.

By Brian Smithers

Call it a singing robot or a talking synthesizer—there's something irresistible about crossing human speech with otherworldly timbres. A *vocoder* (see **Fig. 1**) does just that, mapping tonal characteristics of one sound (typically speech) onto another sound (typically a synthesizer). From Wendy Carlos's synthesized Beethoven choir in *A Clockwork Orange* to recent hits by BT and Madonna, the vocoder has stood the test of time.

Although it may sometimes seem otherwise, nothing says that we can't use a vocoder for more than mapping speech characteristics onto a synth tone. Using a drum pattern to shape the output of a string pad is just one of many interesting and effective variations. Once you understand how a vocoder works its magic, your imagination will doubtless conjure many more uses for it.

After a bit of history, I'll take a look at the inner workings of a vocoder and see what makes it tick. Then I'll discuss what makes one vocoder sound different from another. Be sure to check the **EM** Web site (www.emusician.com) for audio examples of vocoding in action, as well as for a do-it-yourself guide to vocoding.

BACK TO THE FUTURE

The vocoder and its fraternal twin the *voder* were hatched in the late 1930s by a Bell Labs engineer named Homer Dudley. The words *vocoder* and *voder* are derived from "voice coder," and the devices were designed to help reduce the bandwidth required for speech transmission. Interestingly, although we might think of vocoding only in musical terms, many of us also use it daily in nonmusical communications. In fact, Dudley's work is the basis for cellular

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BRIAN SMITHERS

FIG. 1: This figure shows part of the vocoder collection at the Audio Playground Synthesizer Museum, the world's largest such museum (www.keyboardmuseum.org), assembled for research on this article. On top sits the classic Korg VC-10, dating from 1978. At the top of the rack is the MAM VF-11 (current), and below that is an MAM line mixer. The red unit in the middle is the Electrix Warp Factory (1999), and below that is another Korg, the DVP-1 (1986).

and Internet telephony. Vocoding was also adapted by the Department of Defense to encrypt voice transmissions during World War II.

Dudley realized that the human mouth, throat, and nasal cavity together constitute a complex time-varying acoustic filter that shapes the timbre of the basic tone produced by the vocal folds. To demonstrate this remarkably unromantic observation, try the following test. First, check to be sure you're alone (trust me on this). Now repeat slowly after me: "Waouwayweewohwooo." Notice that the tonal variations are produced by changes in the position of your lips and tongue. Your vocal folds can change the *pitch* of your newfound mantra, but the *timbre* is all in your face.

Based upon this straightforward truth, Dudley constructed the vocoder to analyze the varying timbre of the speech input and apply those variations to a synthesized tone instead of the tone created by the vocal folds. The voder, by contrast, generated speechlike output from oscillators and a filter bank controlled by an operator at a specially designed console. The voder was a hit at the 1939 World's Fair, but the vocoder turned out to have the staying power.

UNDER THE HOOD

Let's get away from the "talking-machine" paradigm and use the technical name *modulator* for the speech input, because its timbral variations are used to modulate the sound of the synthesizer. The synthesizer tone is called the *carrier*, because it carries the tonal imprint of the modulator to the final output.

The modulator signal is split into its component frequencies by a bank of bandpass filters much like a graphic equalizer (see **Fig. 2**). The output of each band is analyzed by an envelope follower, a device that creates a control voltage (CV) corresponding to the signal level present at its input. In this way, as the relative strength of various frequency bands vary in the modulator, a set of CVs are created whose variations track the modulator's changes. The modulator has effectively been reduced to control data.

The carrier wave is simultaneously passed through a bank of bandpass filters, ordinarily set to the same frequency bands as those used on the modulator. Instead of being sent to envelope followers, however, the carrier's component bands are sent to voltage-controlled amplifiers (VCAs). These VCAs are controlled by the CVs from corresponding frequency bands of the modulator and map the amplitude fluctuations measured at each modulator band to its corresponding carrier band. The various bands of the carrier are recombined at the output. Thus, the timbre of the carrier tracks that of the modulator while the carrier provides the pitch information.

Vocoding, then, is an example of *subtractive* synthesis. No additional frequency components are added to the carrier wave by the process—they are merely altered according to the analysis of the modulator. That is an effective way to synthesize speechlike sounds, because your voice also uses subtractive techniques, as you can demonstrate for yourself. Once again, look over your shoulder to be sure you're alone. Now stick out your tongue until it touches your chin, open your mouth as wide as you can, and sing a tone. Not pretty, is it? That is the brightest timbre your voice can produce—the raw unfiltered waveform of your vocal folds. (Okay, you can stop singing now!) All other sounds that your voice makes are created by filtering this raw tone with your oral and nasal cavities.

PRACTICAL VOCODING

I like to think of vocoding as a kind of timbral sculpture. A sculptor never adds

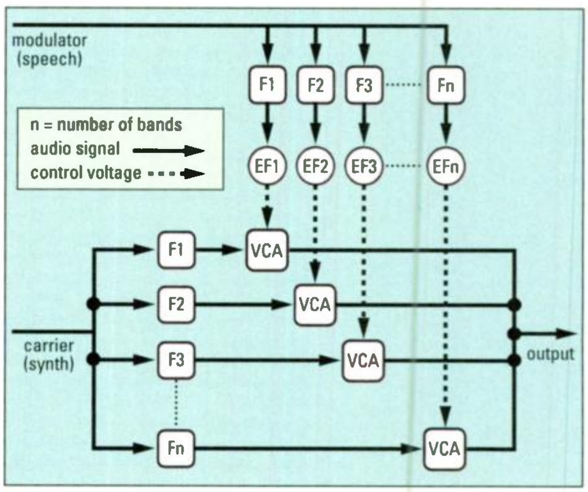

FIG. 2: The modulator is divided by a bank of bandpass filters (F), and then each band is analyzed by an envelope follower (EF). The carrier passes through a similar bank of filters. The control voltage (CV) of each EF controls the output level of the VCA on its corresponding carrier band. The modulated carrier bands are then recombined at the output.

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anything to a block of marble; rather, he or she only chips away at it to reveal the "form within." That requires starting with a block that's the right size and shape to accommodate the sculptor's vision. Because we can't add any harmonics in vocoding, we need to start with a carrier that has plenty of harmonics to "chip away."

Favorite carrier timbres include pulse (square) waves, sawtooth waves, string samples, brass samples, white noise—the same sort of harmonically rich tones on which filter sweeps are effective (see **Fig. 3**). A sine wave is the worst possible carrier for vocoding, because it has no harmonics to filter. No matter how many frequency bands are tracking the modulator, only one has any energy to be affective, thereby reducing the vocoder to a single-band envelope follower.

By the same logic, a modulator should be either harmonically or rhythmically active or there's little point in vocoding it. A static modulator timbre, for example, takes the envelope followers out of the equation and reduces the vocoder to an equalizer. In addition to speech or song, effective modulators include drums, arpeggiated synths, or even a sound like a trumpet with a wah-wah mute. I know an engineer who used his young nephew's violin to modulate a string patch, thus creating a more natural-sounding attack despite the fact that the violin's pitches were unrelated to the song (to put it politely).

JOIN THE BAND(S)

The number of frequency bands has a profound effect on the character of a vocoder. In general, the more bands of analysis and modulation that are used, the more accurately the modulator is represented at the output. For clear speech vocoding, then, having more bands is usually better. Working with a 2-band vocoder, to take the opposite extreme, is like sculpting with a pick axe. Only gross variations in timbre would be represented. Most vocoders offer ten or more bands, with some software vocoders offering hundreds of bands.

Of course, how these bands are distributed is important, as well. There's nothing that says the bands must be evenly spaced; in fact, concentrating the bands in the midrange frequencies where speech is most interesting can help to achieve clarity.

Basic vocoder design as I've described so far has one major flaw in its ability to reproduce intelligible speech. Although it's quite good at reproducing vowel sounds and sustained consonants such as *m* or *n*, it doesn't do a good job with percussive or sibilant consonants such as *t* or *s*. Those are called *unvoiced* sounds, and their waveforms are not periodic, making most of our favorite carriers poor choices to represent them.

One fix for that is to mix some of the modulator input with the modulated carrier so unvoiced sounds are heard in their original form. Alternatively, blending some white noise with the carrier wave before modulation adds some frequency components that help make unvoiced sounds clearer. A more elegant solution is to use a detector circuit capable of distinguishing unvoiced sounds by their frequency characteristics. When an unvoiced sound is detected in the modulator, the vocoder can momentarily switch from its primary carrier wave to a secondary carrier that is better suited to representing percussive or sibilant sounds.

Clarity isn't everything, though. Sometimes your goal will be to make the most ear-catching sound, rather than the most articulate. For that reason, some vocoders feature adjustable resonance on the carrier's filter bank, allowing the user to narrow the bands for a more biting sound.

Gender-bending effects can be produced by remapping the CV signals to the VCAs of noncorresponding frequency bands. Shift the CVs to higher frequency bands, for example, and the timbre of the output takes on a higher character, even becoming chipmunklike in sound when taken to an extreme.

FIG. 3: The Orange Vocoder from Prosoniq features both sampled and analog-style carrier waveforms. With two oscillators, EQ, and reverb, it offers lots of tools for shaping sounds.

Other variables found in some vocoders include attack and release times for the VCAs, independent output level for each frequency band, chorus, built-in carrier waves, and more. Because vocoders are often built in to effects devices or synthesizers, the list of sound-processing variables is virtually endless.

VARIATIONS ON A THEME

Any number of alternative techniques exist that imitate the vocoder's ability to create the illusion of speech. The simplest device is the *talk box*, which is nothing more than a speaker with a tube attached. The tube carries the sound of the carrier from the speaker to the performer's mouth, where the carrier is shaped by that time-varying acoustic filter we discussed earlier. The modulated sound is then picked up by a nearby microphone. The talk box is a favorite with guitarists and has been used famously by Peter Frampton, Joe Walsh, and many others.

Phase vocoding is a digital technique that uses the Fast Fourier Transform (FFT) to analyze the modulator, resulting in a detailed description of its sonic architecture. Once the modulator has been reduced to a set of instructions, it can be reconstructed, either in its original form or in a modified form. Phase vocoding is therefore

a *resynthesis* technique, and it is a particularly smooth way to separate the modulator's pitch and time components for time stretching and pitch shifting. Phase vocoding is found in such products as Csound, U & I's Metasynth, Tom Erbe's freeware SoundHack, and the CDP Composers Desktop Project, to name but a few. The Kyma System from Symbolic Sound also features a huge array of phase-vocoding capabilities.

Digidesign offers a pair of TDM tools called Bruno and Reso that present interesting variations on the vocoder sound. Bruno uses a method of time slicing to extract the timbre of the input signal. These time slices are then crossfaded back together at pitches determined by MIDI input. Reso adds harmonic overtones to the input signal by using a resonance generator, and pitch is also controllable by MIDI input. Notice that Bruno and Reso act directly on the input signal, so there isn't really a modulator/carrier relationship. For that reason, it's useful to think of them as modifying the pitch information of an input signal, as opposed to a traditional vocoder, which modifies the timbral characteristics of the carrier signal. (You say "tomato" . . .) While they can make vocoder-like talking effects, they are even more interesting to use for adding pitches to drum parts and other nonpitched sounds.

VOC-ODE TO JOY

Aside from the sheer giggle factor of sounding like the Cylons from *Battlestar Galactica*, the best thing about vocoders is that they are a textbook example of the power of modulating one signal with another. If you understand the principles of vocoding, using an LFO to create vibrato on a synthesizer and Velocity switching on a sampler all seem like child's play.

For a do-it-yourself adventure in vocoding, go to www.emusician.com. There are also audio examples to give you an idea of ways you might use a vocoder in your own music. Just remember that anything goes and that creative vocoding involves more than just using speech as a modulator.

Be warned, though: it's almost impossible not to get silly when you're playing with a vocoder. While researching this article I kept looking over my shoulder to see when I would be carted away in a straightjacket for making banjo sounds, reciting limericks, barking into the microphone, and other strange behaviors, all strictly in the name of science. Oh, and if you can manage to use a vocoder and not *sing* the part you're playing, you have more discipline than I do!

Brian Smithers *is Course Director of Audio Workstations at Full Sail Real World Education in Winter Park, Florida. Thanks to Joseph Rivers of the Audio Playground Synthesizer Museum (www.keyboardmuseum.org) for his generous assistance and access to the museum's collection of classic and contemporary vocoders.*

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REVIEWS

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Live just got better.

By Len Sasso

Ableton clearly listened to its customers when designing the Live 2.0 upgrade. In addition to numerous user-interface enhancements, the company has added three time-warping options and the ability to turn time-warping off; multichannel, tape-style recording; and a tempo track. There's also beefed-up automation and MIDI control, some new effects, and a rewritten manual. The downloadable upgrade ($69) requires a Live 1.0 serial number; you can request a boxed version for an extra $20. The box contains the rewritten manual (which comes in HTML format with the download) and the installation CD, which contains roughly 250 MB of new samples.

Live is a standalone audio sequencer especially designed for performance but also well suited for recording, arranging, and remixing. It runs under all recent versions of Windows and both Mac OS 9 and OS X. It is ReWire aware and can function as either master or slave (but not both and not as a slave in OS X), and it will host VST effects plug-ins. For those new to Live, you'll find a review of version 1.1 in the June 2002

FIG. 1: Live's Session view is set up like a typical mixing desk, with channel strips along the bottom. Audio clips are triggered from the Clip Pool in the center. Clips can be triggered using MIDI, the computer keyboard, or the mouse.

issue of **EM** and a master class on version 1.5 in the December 2002 issue. For this review, I'll provide a brief overview of how Live works and then get straight to the new features, of which there are plenty.

Although Live runs in a single window, it has two separate-but-linked views corresponding to its two modes of operation. The Session view is laid out like a standard mixing desk, with channel-strip controls along the bottom (see **Fig. 1**). Unlike a mixing desk, there is a Clip Pool in the center for recording and triggering audio clips. In a performance you would trigger individual audio clips or whole rows (called Scenes) using MIDI, the computer keyboard, or the mouse. The Arrange view is laid out like a typical audio sequencer, with audio tracks running horizontally (see **Fig. 2**). You can capture a Session-view performance in the Arrange view, drag clips there and arrange them manually, or record live performances (a new feature in 2.0).

The Session and Arrange views are connected in two ways: they share audio channels, and they are always both active for playback, even though only one is visible. Initially all audio channels

FIG. 3: Live's Clip view is where you choose the time-warping method as well as set various clip parameters. Here, a three-measure section of the audio clip has been selected for looped playback using the Beats method of time-warping.

play Arrange-view tracks, but as soon as a clip is triggered in the Session view, it takes control of its audio channel. In short, that allows you to set up an arrangement of a song, then substitute clips from the Session view in real time. That is the key difference between Live and other audio-sequencing software. Live's unique setup offers many other performance, composition, and recording options as well.

WARP FACTOR

Perhaps the biggest change in version 2.0 is the inclusion of three new time-warp algorithms. Time-warping is the method by which Live keeps audio clips of different tempos in sync. In previous versions, Live simply sliced a clip into equal-sized pieces and sequenced the slices at the master tempo, time-stretching or -compressing the individual slices to fill the allotted time. That method, with occasional manual adjustments to Live's inferred tempo and beat count, works very well for highly rhythmic material such as drum loops and dance tracks. For melodic or ambient material, it is less satisfactory, and for sound effects, which you usually don't want to be warped at all, it is a major nuisance.

Live now has four time-warp modes as well as the option to turn the feature off. Beats is the original mode, just described. Tones is a granular method best suited to melodic material such as vocals, leads, and bass. Live uses pitch analysis to determine the optimal grain size relative to an average grain size that you specify. Texture gives you absolute control over the grain size as well as an amount of randomization around that size. It is best suited for pads, ambient sounds, and complex instrumental textures. Finally, Re-Pitch uses classic resampling and is similar to changing the speed of a tape or record. It is the method to use for DJ-style, variable-speed turntable effects and for time-stretching to match a desired pitch shift. The MP3 example Warped compares the Beats method with the others.

EMWEB CLIPS

As mentioned previously, turning time-warping off entirely is useful for

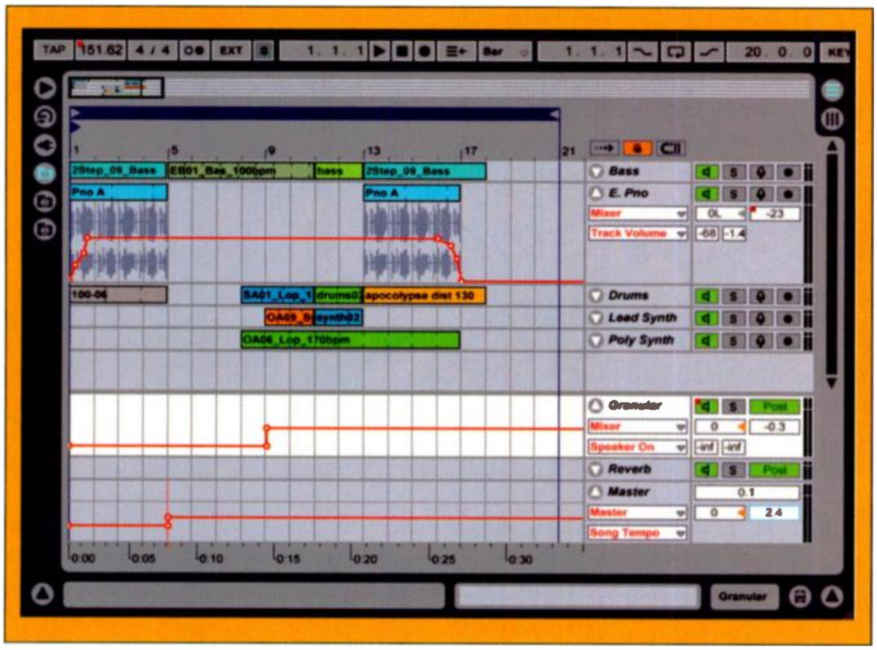

FIG. 2: Live's Arrange view is laid out like a typical audio sequencer, with horizontal tracks holding audio clips. The Arrange and Session views share audio channels, and both are always active during playback.

material such as sound effects, horn stabs, orchestral hits, and individual percussion sounds, all of which should be played as is. If you load such hits into the Clip Pool with time-warping turned off and assign MIDI notes to trigger them, Live becomes a basic sample player. Turning time-warping off is also useful for long audio files, to which you would like to match Live's tempo (instead of the other way around).

The time-warping method can be set individually for each clip in a song, whether the clip resides in the Clip Pool or in the Arrange view. Furthermore, you can use the same clip at different places with different settings. A preference setting lets you choose whether warping is activated when a clip is added, and that setting also determines whether warping is used when clips are auditioned in Live's Browser. It's a little inconvenient to have to open the Preferences window to change that option, and Ableton intends to add it as a panel feature in a future release. In any case, it's great to be able to browse with time-warping turned off.

Two other new features—tap tempo and a metronome—are very helpful when setting up a clip's time-warp markers. Tap tempo allows you to use the computer keyboard or any MIDI controller to tap in Live's tempo to match the tempo of an unwarped clip. (If you try to tap in the tempo of a time-warped clip, you'll be chasing your tail as Live tries to sync the clip to your tapping.) In a nice touch, tapping can also be used as a count-in to start Live playback and recording. The metronome provides a quick way to hear if you've set up your loops and warp markers correctly, although you can use a simple drum clip for the same purpose.

FOR THE RECORD

Audio recording has always been a key feature of Live, but until now, it has been limited to the Session view. Version 2.0 allows tape-style recording in the Arrange view and also includes separate input-monitoring buttons. (Previously, all or only record-enabled channels were monitored for input.) Tape-style recording is multitrack, and remote commands (MIDI and computer keyboard) can be assigned to toggle recording of individual tracks. Unfortunately, you can't toggle recording and monitoring with the same remote command, which makes manually punching in and out a bit difficult. Recording in the Session view is, of course, still possible, and in either view you can simultaneously set the tempo and initiate recording by tapping a four-beat count-in. You can also use the metronome in either view.

Minimum System Requirements

Live 2.0
MAC: G3/233; 256 MB RAM;
Mac OS 9.1 or OS X 10.1.5
PC: Pentium II/400; 128 MB RAM;
Windows 98/2000/XP

The difference between tape-style (Arrange-view) and Session-view recording is that in tape-style recording, clips are time-stamped—the clip is placed in the arrangement at the time it was recorded. Although you can toggle recording on and off in the Session view, each recorded clip is simply placed in the Clip Pool without any reference to its time position in the overall recording. Tape-style recording also supports punch-in and looped recording. In the case of looped recording, all takes of a given track are recorded in a single audio file, within which you can single out the desired take in Live's Clip view (see **Fig. 3**).

Live also provides automation recording, in which all clip-triggering activity in the Session view and all mixing activity (including effects parameter changes) in either view are recorded in the Arrange view. Among other things, that allows you to convert a Session-view recording of audio clips to an Arrange-view recording after the fact by simply triggering the recorded clips at the times you want them to play in the arrangement. In short, almost any way that suits your performance and recording preferences is accommodated by some combination of Live's recording schemes.

One other recording note is that Live accepts input from three sources: your audio interface (meaning the outside world), other ReWire slave applications running simultaneously, and Live's own output (meaning any of the outputs to your audio interface and any of Live's send buses).

Multitrack recording is especially useful with ReWire applications (such as

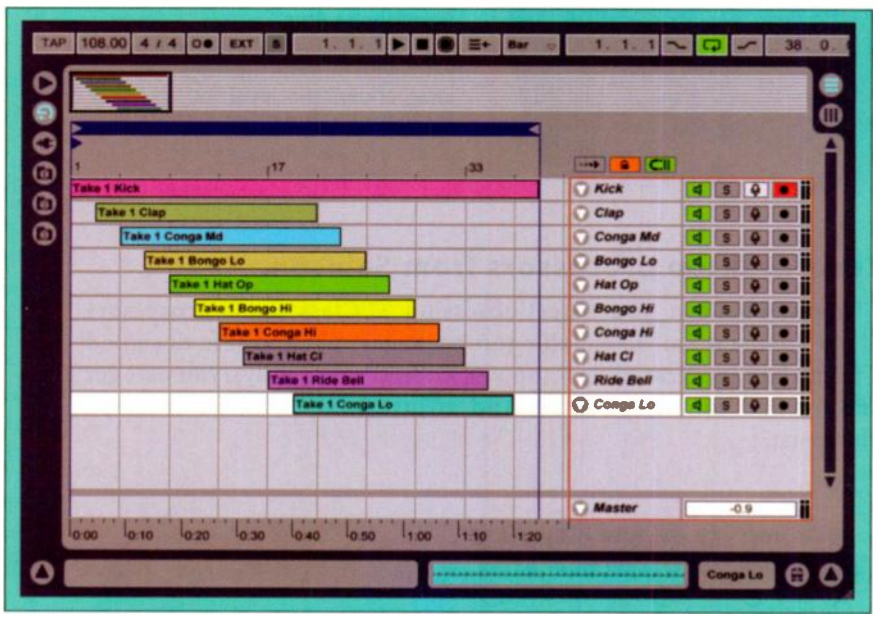

FIG. 4: Multitrack, tape-style recording in the Arrange view allows you to assemble complex arrangements from several input sources in one pass. Here, separate ReWire inputs from the ten pads of Reason's ReDrum drum sequencer were recorded in a single pass.

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Propellerhead Reason) that have multiple outputs, because it allows you to take a multipart loop and turn it into an extended multitrack recording. The MP3 example TenDrum was made from a one-measure ReDrum loop in Reason by routing each of ReDrum's ten drum pads to separate tracks (see **Fig. 4**). Recording Live's own output is useful for splitting off different parts and send effects in a multitrack recording. When you simply want to bounce an arrangement, Live's Render function is faster and easier.

EMWEB CLIPS

ODDS AND ENDS

Enhanced time-warping and tape-style recording are the main buzz in the 2.0 upgrade, but there are also numerous small improvements and additions to make life easier. User-interface improvements include resizable Arrange-view tracks, a compact Session view in which the Clip Pool is suppressed to make more channels visible,

PRODUCT SUMMARY

Ableton
Live 2.0 (Mac/Win)
loop sequencer
$399
update from version 1.0
$69 (download)
$89 (CD and manual)

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

RATING PRODUCTS FROM 1 TO 5

PROS: New time-warping options. Tape-style multitrack recording. Seamless ReWire integration. Easy file management in dedicated Browser.

CONS: Manual has no index. Must open Preferences window to change Browser Warp mode. Can't toggle recording and monitoring simultaneously.

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and a full-screen mode. MIDI and keyboard remotes have been added to change Scenes and to launch the currently selected Scene. Audio clips can now be renamed and deleted directly in the Browser, and all clips recorded in a Live set are placed in an automatically generated set folder.

Live now offers a choice between true solo switches, which mute all other tracks, and Pre-Fader Listen (PFL) switches, which route selected tracks to an alternate output (headphones, for example). Previously, soloing was accomplished by routing the PFL output to the master output. The new method is an improvement, but it's hard to imagine why the same switch is used for both modes, making it necessary to choose between them with a menu option.

Automation has been improved in several ways. You can now lock automation, which prevents it from changing when a clip is moved. Automation selection has been simplified by the addition of parameter menus, and as mentioned, individual tracks can be resized to better view and edit automation by hand. A new, automatable crossfade parameter has been added, and tracks can be individually assigned to either side of the crossfade. Finally, there's tempo automation, which is especially handy when applications that don't support tempo changes (such as Reason) are slaved to Live.

In the effects department, the most notable addition is support for presets for Live's built-in effects. (Previously, only VST effects could have presets.) The two new effects, a gate effect and Redux (for bit-crushing and downsampling), are handy but unexciting. New bandpass and notch modes are announced for the Auto Filter but have not been implemented as of this writing.

Live's new manual is a great improvement over the original. It includes a 40-page getting-started tour and a very helpful section on time-warping. Unfortunately, it lacks an index, and for some reason, the key-commands summary was left out of the printed version. A PDF version would be more convenient than the HTML version provided with the download, but that serves the purpose. The 250 MB of new samples and loops are also a welcome addition to Live's already excellent factory sound library. Most of those are in the PowerFX collection of piano, organ, bass, drum, and orchestral clips, and they are quite nice.

Live has gained a well-deserved reputation as a unique tool for real-time audio sequencing and recording. It is equally at home in performance and desktop composition. The upgrade (I recommend the boxed version) is incredibly cheap for what has been added, and the full package is certainly fairly priced.

Len Sasso *can be contacted through his Web site at www.swiftkick.com.*

DAVE SMITH

EVOLVER

A synth-building legend combines analog and digital technology into something new.

By Geary Yelton

It's practically impossible to review the Evolver, a monophonic analog-and-digital hybrid synthesizer from Dave Smith Instruments, without first focusing on Dave Smith himself. His contributions to the field of electronic musical instruments—including the Sequential Prophet-5, wave sequencing, and a simple idea that grew to become the MIDI specification—have already given him a degree of immortality. Consequently, when he built and began selling a new instrument of his own design and under his own name, **EM** had to take notice.

Weary of software theft and willing to buck the trend toward software-based instruments—a trend that Smith helped initiate with Seer Systems Reality—he has produced what he calls "the ultimate dongle": a hardware-based instrument. The Evolver combines digital and analog oscillators, digital and analog filters, an analog-style sequencer, external-audio-processing capabilities, and a matrix-style user interface in a tabletop synthesizer and effects box that's capable of producing sounds like no other.

ORIGIN OF THE SPECIES

The Evolver is housed in an unassuming black steel box with a slightly sloped top covered by a blue vinyl control panel. At the top of the panel is a row of eight infinite-rotation rotary encoders with a button labeled Main to their left. With the button in one position, the eight knobs select Programs and Banks, set the master transposition, change Volume, and control the sequencer clock. In the other position, they control other global parameters such as Fine Tune, MIDI Channel, data filtering, and so on.

Below the knobs is a printed matrix that lists 128 parameters (see **Fig. 1**). Half the parameters are selected by pressing one of eight buttons to the left of the matrix and then turning a knob. You select the other half by first pressing and holding the Shift button, which locks when you double-click it. Two additional buttons are devoted to the sequencer, and the Reset button turns all notes off and resets all MIDI controllers. All knobs and all but two buttons each have an associated LED that lights to indicate whether it's active.

The Evolver's user interface harnesses the time-honored tradition of parameter names printed in rows below reassignable knobs, serving the same noble purpose they have in earlier instruments: economy. Indeed, the Evolver's economy extends to size as well as cost. If any other type of user interface had been used, the front panel would have needed to be much larger. As I learned my way around the Evolver, I discovered other cost-cutting measures. For example, although the knobs feel reasonably solid, the buttons are tiny and flimsy, more suited to a clock radio than a piece of studio equipment.

For heavy-duty programming, then, it would make sense to rely on a software-based graphic editor such as Emagic SoundDiver (see **Fig. 2**). Fortunately, you can download a SoundDiver adaptation from the manufacturer's Web site (www.davesmithinstruments.com). If you happen to own a Peavey PC 1600x control surface, you can download presets for controlling the Evolver. You'll also find scripts for naming Evolver Programs in Steinberg Cubase. In addition, MOTU Unisyn 2.0 offers an Evolver editor, and you can download Matthew Davidson's editor application for Mac OS 9 (http://apocalypse.org/~matthew/resources/evolver/).

The value of the selected parameter is shown in a large, three-digit LED not unlike those found on Sequential synthesizers of yesteryear. The disadvantage of such a display, of course, is that it can't show real words, only numbers and cryptic abbreviations. It also means that Programs can't have names on the control panel—only numbers.

Because there's no dedicated Volume knob, you can't instantly change the Evolver's output level unless the Main button is in the correct position; in some circumstances, that could be a problem. I'd also appreciate being able to select Programs directly from the front panel without having to turn

FIG. 1: Synthesizers don't come much smaller than the Evolver, the first product from Dave Smith Instruments. At the heart of the user interface is a matrix of parameters printed on the control panel.

a knob in order to step through them.

The Evolver's rear panel is best described as "no-frills" (see **Fig. 3**). It has two ¼-inch audio inputs, two ¼-inch audio outputs, three MIDI ports, and a power jack. I was disappointed by the lack of an output for stereo headphones, and neither output jack serves that purpose. (You could always use an adapter, but it wouldn't be stereo.) Also conspicuous in its absence is a power switch. I already own too many devices without power switches. Was it really cost prohibitive to add a single switch or a Volume knob with an Off position? I'd have gladly paid a few dollars more for their inclusion. The Evolver draws its power from a 13V wall wart that's accompanied by enough adapters to operate nearly anywhere in the world.

The 64-page manual, which is provided in paper form, is well organized and informative. Because it has no index, though, I was pleased to find a searchable PDF version on the manufacturer's Web site. The manual begins with a very brief tutorial section and follows with frequently asked questions, overviews of the user interface and sequencer, and detailed explanations of every programming parameter. Near the end are lists of modulation destinations and sources, as well as information on the Evolver's MIDI implementation.

FIG. 3: The Evolver's rear panel embraces the concept of minimalism: two ins, two outs, three MIDI ports, and a power connection.

THEORY OF EVOLUTION

The Evolver's synthesis architecture is unique, but it should be familiar enough for any experienced synthesist to grasp quickly. First and foremost is the presence of two traditional analog oscillators and two digital wavetable oscillators. Each oscillator pair is assigned to separate stereo signal paths—individual oscillators hardwired either left or right—and those paths carry the signals through all subsequent processing stages, extending all the way to the audio outputs. The Evolver is monophonic, yes, but it's a true stereo instrument. In combination with two such different oscillator designs, its stereo nature gives the sound a wonderful depth and dimension that most mono synths are sorely lacking.

The Evolver's two analog oscillators are pretty standard, offering a choice of sawtooth, pulse, and triangle waveforms as well as a combination of sawtooth and triangle. The pulse wave provides full-range pulse-width modulation—just as you would expect in any serious subtractive synthesizer—and you can narrow its width down to nothing. One analog oscillator can be hard-synced to the other for thick, biting timbres when one is modulated by an envelope generator.

Smith modeled the Evolver's two digital oscillators after those of the Sequential Prophet-VS, a classic instrument that he designed in 1986. In fact, they contain the same 95 sweepable 12-bit wavetables as the VS oscillators (minus noise, which is supplied by the noise generator). Love it or hate it, they exhibit the same audible aliasing in the upper frequencies. You can also load 32 of your own wavetables into the digital oscillators (as you could on the VS), but you'll have to wait until software is available for that purpose. The digital oscillators offer frequency (FM) and ring modulation, and either oscillator can modulate the other, even simultaneously. The resulting timbres can quickly get out of hand.

FIG. 2: A computer-based editor offers the best means to get a grip on programming the Evolver. An adaptation for Emagic SoundDiver is shown here.

Each oscillator provides a separate Glide parameter, so you can independently assign portamento to any of the four oscillators. When you specify different Glide rates for each, the resulting animated sound can make you quickly forget that you're playing a monophonic synthesizer.

The signal path has two dedicated stereo filters—an analog, lowpass, voltage-controlled filter (VCF) and a digital highpass filter. You can toggle the VCF's cutoff slope from 2-pole to 4-pole, effectively altering its character from fat to fatter, and turn up the resonance to drive it into self-oscillation in 4-pole mode. A parameter called Split emphasizes the filter's stereo characteristics by allowing you to raise the left side's cutoff frequency while lowering the right side's. I'd rather have independent control of two lowpass filter frequencies, but short of that, Split's implementation is an imaginative solution. Normally, the stereo 4-pole highpass filter is placed after the lowpass filter and voltage-controlled amplifier (VCA), but when you're processing external signals, you can place it before the VCF for greater flexibility.

The Evolver provides three ADSR envelope generators (EGs). One is hardwired to control the filters, another to control amplitude, and the third may be routed to any modulation destination. However, you can assign any of the three EGs to any modulation destination.

Four identical LFOs are also assignable to any destination you please. Each LFO generates five waveforms and syncs to the sequencer or MIDI, if desired. At its lowest frequency, the LFO cycles only twice a minute.

One of the Evolver's programming features is tunable Feedback. Each channel has a tuned delay line; most of the time, it simply adds an interesting distortion by emphasizing the Feedback frequency. At high Feedback levels, the Grunge parameter makes the sound considerably more aggressive.

The Evolver is monophonic, yes, but it's a true stereo instrument.

You can also play melodically by modulating the Feedback frequency with the sequencer. When you route the signal back to the filter inputs, the result is essentially Karplus-Strong plucked-string synthesis.

To aid in your quest for radical timbres, the Evolver offers additional methods for distorting audio. You can place the stereo digital distortion processor either before the analog filter, where it affects only the external audio inputs, or after the VCA (but before the delay processor). An Output Hack parameter imparts an especially noisy form of distortion. You can also mangle an external signal with Input Hack and then soften its effect with analog filtering.

Like any sophisticated synthesizer, the Evolver relies heavily on its modulation capabilities. You can route 24 sources to any of 55 destinations, but the way that control signals are routed, each source can have no more than one destination. However, that limitation doesn't appear to be a serious problem because it's not entirely true. You also have four user-definable modulation routings in addition to four predefined external modulation sources: mod wheel, Aftertouch, breath control, and MIDI foot controller.

PROCESSING PLANT

The ability to process external audio is part of the Evolver's raison d'être. You've never had so many ways to distort, twist, and mangle any sound. To that end, its stereo inputs can handle a wide range of signals, from line levels to high-impedance signals from guitars and keyboards. As an aid to setting the input gain, the front-panel LEDs serve as peak-level meters in combination with the main display, which shows the input level in 3 dB increments. Thanks to its many forms of distortion, the Evolver excels as a distortion processor.

A digital envelope follower responds to signals present at the left input; you can assign it as a source for any modulation destination. Likewise, the Peak Hold parameter can use the changing level of the left input as a modulation source. Typically, these sources are especially useful for modulating filter and envelope parameters, but you can achieve some unorthodox effects by applying them to modulate oscillator frequency or delay time, for example.

The Evolver's only traditional time-modulation effects processing is a 3-tap stereo delay line. Regrettably, the delay is monaural, which precludes programming cool stereo ping-pong effects. Nonetheless, the Evolver's factory Programs make extensive use of the delay, squeezing every ounce of functionality from its three taps. There's no reverb, but you'd never know it to hear some of the Programs. I wish that a built-in reverb processor had been included; I'm sure, however, that that would have driven the cost up considerably. You can use the Evolver solely as a freestanding effects processor, but it probably won't take the place of a more general-purpose multi-effects box.

SEQUENCE OF EVENTS

Nothing beats an analog-style sequencer when it comes to programming sounds that evolve. Accordingly, the Evolver's sequencer is an essential part of its design. Although the Evolver has only 8 knobs, they function as if they were controlling a sequencer with four rows of 16 knobs. In essence, the sequencer stores four sequences that run in parallel. Each has a maximum length of 16 steps, and you can set the length of each independently; for example, two 16-step sequences can play alongside an 8-step sequence and a 5-step sequence. You can't, however, chain one sequence after another. In addition, the sequencer does not transmit MIDI data.

Each sequence is a modulation source that you can assign to control oscillator pitch, filter frequency, or any other modulation destination. Most of the time, at least one destination will be pitch. Of particular interest to microtonal musicians is that pitch is adjustable in quarter-tone increments. By assigning each row to modulate a different oscillator's pitch, the sequencer can even play four-note chords. Try doing that with any other monophonic synthesizer! One of the most important sequencer applications is modifying the digital oscillators' wavetable for wave sequencing.

Programming the sequencer is easy enough. By pressing the Sequencer button in combination with the Main button's two positions, you can turn the knobs to specify the value of each sequencer step, either as it's stopped or running. It also responds to MIDI notes for entering pitch, much like step-time entry in a computer-based sequencer.

As with the LFOs, you can slave the sequencer clock to MIDI. Except for applying the clock's Swing parameter, all steps are of equal duration, but you can simulate rhythmic variety by defining selected steps as rests. I used to own an analog sequencer that let me assign one row of knobs to determine the duration of each step, and I'd love to be able to do that on the Evolver.

Evolver Specifications

Sound Engine	analog and digital subtractive synthesis
Polyphony	monophonic
Programs	(384) rewritable
Oscillators	(2) analog with 4 waveforms, hard sync; (2) digital with 128 wavetables, FM and ring modulation; (1) noise generator
Filters	(2) voltage-controlled, analog resonant lowpass, switchable 12 or 24 dB per octave; (2) digital highpass, 24 dB per octave
Amplifiers	(2) VCAs
Envelope Generators	(3) ADSR
LFOs	(4) multiwaveform with MIDI sync
Effects	3-tap delay with MIDI sync (1 sec. maximum); stereo digital distortion
Sequencer	4-parameter × 16-step analog-style with MIDI sync
A/D/A Conversion	48 kHz, 24-bit
Analog Audio I/O	(2) unbalanced ¼" TS inputs; (2) unbalanced ¼" TS outputs
MIDI I/O	In, Out, Thru
Front-Panel Controls	(8) infinite-rotation rotary encoders; (13) buttons
Display	3-digit LED
Power Supply	13V wall wart, 100–240V AC
Dimensions	10.75" (W) × 1.50" (H) × 6.00" (D)
Weight	3 lb.

PRODUCT SUMMARY

Dave Smith Instruments
Evolver
synthesizer module
$499

FEATURES	3.5
EASE OF USE	3.0
QUALITY OF SOUNDS	3.5
VALUE	4.5

RATING PRODUCTS FROM 1 TO 5

PROS: Innovative architecture. Stereo I/O. Real analog filters. External audio processing. Cool distortion effects. Flexible hands-on sequencer. Lots of unique timbres.

CONS: No headphone output, power switch, or dedicated Volume knob. Three-digit display. Flimsy buttons. No reverb.

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Web www.davesmithinstruments.com

SOUNDS OUTRAGEOUS

The Evolver's emphasis is on making sounds that change over time—evolving, if you will. The sequencer's ability to modulate any mod destination helps a lot in that department; a sound can change quickly or slowly, subtly or radically, with as many as 16 distinct phases. Add to that four LFOs, four fixed modulation routings, and four user-assignable modulation routings, and you're well on your way to breathing life into your music.

You might be confused if you first connect a MIDI keyboard before you begin to explore the Evolver; that's because the first Bank of 128 Programs contains sequence-driven sounds, and the first 69 don't respond to MIDI notes. To hear those, all you need to do is press the Start button. By combining analog and digital timbres, the first Bank does a fine job of showing off what the Evolver can do. You're sure to be amazed at the variety of tones that spill from your speakers. Most of those Programs, however, do more to show off the Evolver's capabilities than to serve any useful musical function. I hope that many factory Programs will be replaced as talented programmers get their hands on the Evolver.

The second Bank's Programs are keyboard oriented, and most of them are programmed to respond to various real-time control parameters such as Aftertouch and the mod wheel. Bank 3 is a mixed bag containing drones, sequence tones, and keyboard timbres. Some of the drones are perfect if you're scoring a sci-fi film and need to impart an alien atmosphere. In addition, 20 Programs in Bank 3 are for processing external audio inputs, and half of them are specifically intended for electric guitar. The guitar-oriented Programs are especially fine, probably owing to Smith's previous work on the Roger Linn Design AdrenaLinn, but I wish there were more of them.

SURVIVAL OF THE FITTEST

You say you want an evolution? Dave Smith has done his best to supply you with the next step in audio synthesis at a reasonable price, and he's succeeded admirably well. Costing not much more than some soft synths, the Evolver packs a lot of sound-design and sound-manipulating power into a desktop unit that's smaller than most software boxes. And like any good synthesizer, the Evolver is a boxful of fun that will keep you tweaking and making music for hours at a time.

Other than a few rather expensive modular synthesizers, I haven't seen another instrument that combines analog and digital oscillators since Roland introduced L/A synthesis in 1987. The Evolver is one of those few electronic instruments that sound totally original. I can guarantee that if you buy one and learn to program it, you'll be making sounds that you'd never make without one.

EM *associate editor* **Geary Yelton** *lives in Charlotte, North Carolina. He has been playing, programming, and writing about synthesizers for 30 years.*

TC-HELICON

VOICEONE

Give your vocals a lift with the latest modeling technology.

By Mark Nelson

In a perfect world, every vocalist would sing in tune and place each phrase neatly in the pocket. Capable harmony singers would hang out on every street corner, and no one would ever catch a cold. Matching the perfect singer to the track would be child's play, and the producer would never, ever decide to change the key after the singer has left for the Bahamas.

It's unlikely you live in a perfect world. Ever since the earliest days of recording, studio engineers have sweated to repitch warbling crooners, turn smooth-cheeked altar boys into would-be rock gods, tease a harmony track from thin air, or mask the nose job that a diva had between the original session and the overdubs.

Combining voice modeling (VM), intelligent pitch correction, and spot-on pitch-shifting, the VoiceOne from TC-Helicon puts the most-essential vocal manipulation tools into a single rackspace. Unlike the VoicePrismPlus, it offers no dynamics processing, EQ, or effects. Nor do you get four harmony voices, although with a little effort the VoiceOne can create stunningly realistic harmony parts that far surpass anything I've heard previously.

LOOK AROUND

With an attractive metallic-blue faceplate, the VoiceOne shares the look of several of TC Electronic's popular effects boxes. Controls are logically arranged around the display. Utility functions nestle around the data wheel, and buttons for editing various VM and pitch parameters are grouped in the center (see **Fig. 1**). Two rotary pots offer manual adjustment of input and output gain over a 24 dB range relative to internally set levels. Dual ladder meters reveal input and output levels at a glance, and, as an unintended consequence, give visual confirmation of the processing delay.

Just as all controls and displays are on the front panel, all connections are on the rear (see **Fig. 2**). The most obvious difference between the VoiceOne and its predecessors is the lack of a microphone input. Analog ins and outs are on balanced XLR connectors only. Given that the unit accommodates a wide range of levels, I'm surprised that TC-Helicon didn't opt for Neutrik combo connectors, which offer ¼-inch and XLR inputs. Oddly, only one of the two analog inputs is active at a time, and the two outputs are split between the dry (unprocessed) and wet signals. To compensate for conversion and processing delay, the dry signal is delayed to match the wet signal.

Digital I/O is on RCA jacks, but the unit can switch from 24-bit S/PDIF to AES/EBU. All processing is at 24 bits, and you can dither the output down to 20, 16, or 8 bits. The VoiceOne supports 44.1 kHz and 48 kHz sampling rates with internal or external clock. Completing the rear view are the usual trio of MIDI jacks, a power-cord receptacle, and a ¼-inch TS jack for a footswitch.

NAVIGATION AND EDITING

The first thing you notice is the large display. Hats off to the design team, which managed to squeeze a lot of information into a limited space. In addition to the level meters, it provides information on the current preset or edit page, sampling rate, base key for pitch-shifting, MIDI status, pitch recognition status, and other functions relating to pitch correction (which I'll discuss later).

The display's right side is home to a row of six bar graphs showing the relative depth of each VM parameter. Corresponding buttons on the faceplate toggle each effect on and off. Double-clicking on a button conjures up the appropriate edit screen. You make edits using up and down arrow keys and the data wheel—a sometimes tedious process, considering the number of choices for some parameters.

Although the VoiceOne's user interface is consistent with those of other TC products, I found a few oddities. The only way to exit an edit screen is to choose another parameter to edit or

PRODUCT SUMMARY

TC-Helicon
VoiceOne
vocal processor
$1,299

FEATURES	4.0
EASE OF USE	3.5
AUDIO QUALITY	4.0
VALUE	4.0

RATING PRODUCTS FROM 1 TO 5

PROS: Realistic pitch-shifting and pitch correction. Extensive control of vocal formants, inflection, and vibrato. Multiple pitch-shift and correction options.

CONS: No ¼-inch analog I/O. No direct input of numerical parameters. Poor documentation.

Manufacturer
TC-Helicon/TC Electronic (distributor)
tel. (805) 373-1828
e-mail info@tc-electronic.com
Web www.tc-helicon.com

FIG. 1: The VoiceOne combines vocal modeling, pitch correction, and pitch-shifting functions in a single-rackspace unit. The front panel provides abundant hands-on control and visual feedback.

press the Recall button. But be careful: pressing Recall twice erases your edits by recalling the previously saved preset. More confusing is that the VM depth meters are active even for parameters that are disabled. More than once, I had to confirm that an effect was on or off. Fortunately, there's a status light on each button.

Overall, editing on the VoiceOne isn't as easy as it should be, and the documentation is tough to understand. If TC-Helicon's writers can't provide an intelligible manual, they should at least proofread the one they have. And they shouldn't make users waste their time downloading names and descriptions of all the parameters from TC-Helicon's Web site (see the sidebar "Online Resources").

A total of 99 factory presets showcase the VoiceOne's capabilities and serve as a starting point for tweaking. Edits can be stored to any of 50 user locations—an adequate number, but I wish I could overwrite the factory presets. Just how many times do you need to dial up something called Can't Sing, except, perhaps, to scare the pants off an arrogant client?

MODEL BEHAVIOR

Although the VoicePrismPlus introduced voice modeling, the second-generation algorithms in the VoiceOne are a significant improvement, with greatly expanded control and realism. Just as digital modeling re-creates the sound of classic guitar amplifiers, VM simulates the characteristics that make a human voice unique: the length of the vocal tract, the shape of the nose, breathiness, vibrato, inflection, and even the rasp of overextended vocal chords. Because each singer is different, it isn't simply a matter of superimposing a model of a famous singer and, voilà, instant Bing. But used subtly, VM is a powerful tool.

Here's a rundown of the VM parameters and what they do. Inflection adds *scooping* (a singer's characteristic of starting above or below a note), portamento, and doubling effects. A variety of humanizing parameters control the inflection's onset, amount, randomness, pitch variation, and timing.

Thanks to an algorithm called Flextime, the processed vocal slows down or speeds up independently of the source. At times, the processed signal hits the note first! Obviously, that requires the dry signal to be delayed significantly; a little deft track sliding in my audio sequencer puts things back in the pocket. For greater realism, Time Randomization decouples the timing of the two voices. For doubling or harmony effects, the results are uncanny.

Vibrato brings pitch and amplitude

VoiceOne Specifications

Analog Audio Inputs	(2) balanced XLR
Analog Audio Outputs	(2) balanced XLR
Digital Audio I/O	(1) RCA in, (1) RCA out; 24-bit S/PDIF, AES/EBU
A/D/A Conversion	44.1, 48 kHz; 24-bit, 128× oversampling
MIDI	In, Out, Thru
Control Jack	¼" TS footpedal
Presets	(100) factory (1 blank); (50) user
Display	23-character, 280-icon STN-LCD
Dynamic Range	100 dB, 20 Hz–20 kHz
Total Harmonic Distortion	<–94 dB (0.002%) 1 kHz, +20 dBu output
Frequency Response	20 Hz–20 kHz, +0/–0.5 dB @ 48 kHz
Power	100–240 VAC; <15W
Dimensions	1U × 8.2" (D)
Weight	4.1 lb.

AVOIDING RODENTS

As anyone who's ever played around with the speed control on a tape recorder or turntable knows, changing a voice's pitch alters its formant characteristics radically. Nearly a half century ago, by multitracking his voice and speeding up the tape, producer David Seville created the Chipmunks, earning himself a string of hit records and immortality of sorts.

To avoid sounding like a chipmunk or giant when you do pitch-shifting, you must re-create the formant characteristics of the original voice—a task that requires a fair amount of DSP power and causes a substantial processing delay. To overcome the problem, the VoiceOne features two basic pitch-shifting modes. Normally, using pitch-shifting along with any of the VM parameters yields the most realistic effects. You can easily correct for formant alterations created by pitch shifts or use VM to simulate the sound of a second vocalist.

However, in a live situation where even a tiny delay won't cut it, choose PureShift mode. That allocates all the DSP processing power to pitch-shifting and correction and completely bypasses VM processing. The result is greatly reduced latency at the expense of a slight degradation in quality. You retain some control over formants by means of a single formant editor. Positive numbers yield progressively more masculine or mature sounds, and negative values sound, well, younger. I discovered I could mutate my yawp into a credible imitation of a little old lady singing the national anthem without too much effort.

PureShift offers one additional formant control: the oddly implemented Hybrid Shifter Ratio. According to the manual, it controls the formants during shifting. At a setting of 100 percent, formants are shifted along with the pitch, and 0 percent is full formant correction. To simplify things, an Auto setting is also available.

modulation to the voice. Styles named Ballad, Broadway, Classic Rock, Nervous Tremolo, and Opera Tenor—50 in all, including my favorite, Sheep—make it easy to find just the right vibe. I noticed a clear improvement over the first-generation VM effect: for the most part, the vibratos were rich and quite realistic. Controlling vibrato depth with my MIDI keyboard's modulation wheel was a hit-or-miss proposition. Although I liked the ability to choose the onset of vibrato, I often overshot the mark and added way too much.

Spectral styles refer to various EQ curves inherent within an individual voice. In conjunction with the resonance parameters, they create everything from subtle to drastic transformations. With 27 styles to choose from, the best way to hear how they work is to loop a vocal and scroll through them. If you need to sweeten a nasal vocal or add depth to a wimpy harmony part, Spectral styles are a good place to start. I'd be tempted to buy a VoiceOne simply for the improvement this parameter made on my voice.

Resonance styles change the voice's harmonic content in more dramatic ways. Some, such as Fat Tongue and Sumo, are fairly descriptive. Others, including the nine Transmute styles, have to be auditioned. If realism is your goal, a little of the Spectral style and Resonance style goes a long way, but there's some wild and crazy fun to be had here.

Two additional parameters, Breath and Growl, are self-explanatory. These parameters are best used in moderation, and a little goes a long way. Of the two, I think Breath might be the more realistic. A parameter called Breath HarmX simulates the harmonic shifts in a natural voice as the vocal folds open wider.

Taken together, the VM parameters are impressive. No matter whether your goal is to double a lead, create phantom harmony vocalists, or subtly alter the character of an existing vocal, you will find what you need.

PERFECT PITCH

Pitch correction and pitch-shifting are two sides of the same coin. Correction is used to fix individual notes in a slightly out-of-tune vocal; pitch-shifting generally implies either creating artificial harmonies or transposing an entire track. The VoiceOne handles each task remarkably well, though in slightly different ways.

As someone who has spent a lot of time recording vocalists who make up in enthusiasm what they lack in technique, I am a firm convert to automated pitch correction. Consequently, I was surprised to discover that my first attempts with the VoiceOne yielded less-than-stellar results. A quick trip back to the user manual revealed the reason: you must select not only a key and scale (as is common to both hardware and software pitch-correction processors), but also the window in which the effect will operate. In other words, if the voice wavers at 150 cents above the target but the window is set to 100 cents, there will be no correction.

Once I got the hang of it, though, I was sold. By setting the correction window suitably wide and manipulating the

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ONLINE RESOURCES

The 26-page manual and skimpy quick-start guide barely scratch the surface of what's inside the VoiceOne. To get the most from the unit, you'll need an Internet connection. TC-Helicon's Web site features ample resources, including white papers, application guides, FAQs, an online tutorial, and even audio files from the mind-boggling demo CD, complete with producer's notes. I'd also suggest taking a run through the online pitch-correction workshop.

To learn how the TC-Helicon products differ from other pitch-shifters, check out the white paper titled "Pitch Shifting and Voice Transformation Techniques," by Patrick Bastien. Likewise, the company's Web site is the only place you'll find a complete listing and description of all the VM parameters.

correction amount, I could fix large errors in pitch while letting the smaller deviations around the note remain, and the results sounded very natural. As a test, I corrected a vocal on the fly, recording the natural and processed vocals on adjacent tracks. Aside from the processed track being noticeably more in tune, the two sounded virtually identical. I heard none of the warbling and chirping that I have heard from other pitch-correction products.

For pitch reference, you select a key, then enter a scale or choose from one of 50 factory scales. Everything from standard diatonic, minor, and blues scales to various jazz, pentatonic, and non-Western scales are offered. The latter are somewhat problematic; the so-called Hawaiian scale is essentially a minor pentatonic variation, despite the overwhelmingly major orientation of postcontact Hawaiian music. Likewise, the Arab scale does not correspond to any of the *maqam* I could find in my library. My guess is that the designers simply totaled up all the permutations of five and seven notes available on the keyboard and doled out colorful names. It isn't a huge failing, but I'd love to see some real microtonal choices offered.

For easy reference, notes of the selected scale are shown on a tiny keyboard in the display. Hollow circles on the keyboard show the source vocal's pitch, while a pair of horizontal meters displays deviation from true (±200 cents) and the amount of correction applied. If you select MIDI in the Scale menu, correction occurs only when you hold down a note or group of notes.

SHIFT AND HARMONY

The VoiceOne handles pitch-shifting tasks as well as any other processor I've heard, thanks to some very cool algorithms to discourage chipmunks and

FIG. 2: All connections are on the VoiceOne's rear panel, which provides analog inputs and outputs on XLR jacks, digital I/O on RCA jacks, and a footpedal input for external control.

other common artifacts (see the sidebar "Avoiding Rodents"). To shift pitch as much as two octaves, select Chromatic mode and spin the data wheel. With accelerated data entry, it takes five revolutions of the wheel to twist your way from –2,400 up to 2,400 cents. Although you can't input numerical values directly, you can shift by scale degrees relative to the scale selected in the Correct Scale menus. However, this requires you to move back and forth between the Correct and Pitch edit windows.

As with pitch correction, you can control pitch-shifting with MIDI data, albeit on a separate channel. And because pitch-shifting overrides correction, you can accomplish both tasks simultaneously, correcting a vocal on the fly and force-shifting selected notes as you wish. That's right, you can replay a vocal line's melody on your keyboard. I can't wait to hear how some creative producer manages to abuse that amazing feature.

Although its primary mission is not harmony processing, the VoiceOne excels at the task. Simply choose a scale degree and select from one of three major and three minor scale-transposition maps, or create your own. Adjusting VM portamento parameters smooths the transitions between harmony notes. Additional adjustments to formant and Flextime parameters make harmonies sound eerily realistic. If I hadn't recorded the track myself, I would have sworn a second singer had entered the room.

HALLELUJAH CHORUS

Whereas the VoicePrism and VoicePrism-Plus straddle the live performance and project studio markets, the VoiceOne is first and foremost a studio tool. Live performers might be put off by the processing delay. But latency can be diminished at a small loss in vocal quality.

The VoiceOne's voice modeling is a big improvement over the technology's previous incarnation, but it still has a long way to go. It's easy to overdo the processing, particularly with the Breath and Growl parameters. But with care, you can create stunningly realistic effects. The pitch correction and pitch-shifting functions are far ahead of anything else I've heard.

The VoiceOne is a great toolbox. If you work with vocals, you'll find a million reasons why you need one.

Acoustic musician **Mark Nelson** *produces the Aloha Music Camp, a weeklong immersion into Hawaiian music and culture held in August on the Big Island of Hawaii (www.alohamusiccamp.com).*

SONIC FOUNDRY

ACID PRO 4.0A (WIN)

The original loop sequencer gets serious.

By Dennis Miller

Sonic Foundry's Acid Pro is widely recognized as the premier loop-assembly tool available for the PC today. Although its interface is not as well suited for live performance as, say, Ableton Live's, Acid's enormous power and real-time processing capabilities make it an ideal tool for music production.

The new version of Acid adds a significant number of features, including support for more audio protocols, such as ASIO; 5.1 surround mixing; and the ability to use VST Instruments, under the control of MIDI tracks. With those and several other additions and enhancements, Acid moves ever closer to being an all-in-one media-production environment.

We reviewed Acid version 3.0 in the May 2002 issue of **EM**, so I'll focus primarily on the new features in version 4.0a and mention existing features only in passing.

GET SOFT

Among the major new features of Acid Pro 4.0a is the ability to use VST Instruments in your projects (see **Fig. 1**). Acid's Soft Synth Chooser will search your hard drive to find all the VST Instrument plug-ins on your system. If you have Instruments scattered among several folders, you can use the VSTi Preferences dialog to identify as many as three locations for Acid to access. You'll also use that dialog to update Acid any time you install a new synth or sampler.

After several attempts, I couldn't get Acid to see all my VST Instruments. Then I tried a tip I found on Sonic Foundry's Web site: I deleted Acid's VST key in the Windows Registry. Everything worked fine on my next attempt.

You can automate Instrument parameters by recording MIDI data directly onto an Acid track (more on MIDI later). In addition, you can add DirectX effects to a synth's output using the same methods you'd use for your audio tracks: adding a Track effect directly to the track, using a DirectX effect with an Assignable effect, or assigning the track to either a sub or the Master bus and applying the effect there—Acid provides lots of flexibility. Building a chain of effects is easy, and once you have a sequence of effects that you like, you can save it for use in other projects.

FIG. 1: Acid Pro 4.0a supports VST Instrument plug-ins and can trigger them from its new MIDI tracks. Note that each track type—video, MIDI, and audio—has its own icon, shown at the far left of each track.

Speaking of effects, you can now automate the parameters of many effects, including Track EQ, Resonance Filter, and Flange/Wah/Phase. Click on the Track FX icon and select an effect from the Audio Plug-in dialog. Then click on the FX Automation icon, and you'll see the parameters of the plug-in you've chosen. Enable automation for one or more parameters, and when you return to the main Track view, you'll find color-coded envelopes for all the parameters you've enabled. In addition to linear envelope segments, Acid Pro 4.0a supports several new fade types, including fast and slow fades.

Minimum System Requirements

Acid Pro 4.0a

Pentium II/300; 64 MB RAM;
Windows 98 SE/2000/ME/XP

MIDI MATTERS

Another major new feature in Acid Pro 4.0a is MIDI recording and editing. You can record in either real time or step time, but there are some limitations. For starters, assuming you want to hear what you're recording, you must route your MIDI controller's input to the soft synth you're playing, by using the MIDI Thru drop-down menu in the Record screen. You can't simply enable a MIDI track for recording and assign that track's output to the synth as you would in a dedicated MIDI sequencer. You also don't double-click on an Event (Acid's name for what some other programs call a clip) as you would in a sequencer to access the data once it's recorded. Instead, you have to click on the track's Track Properties icon, then tab to the Piano-Roll Editor. On the other hand, the data you record is stored in a separate file, which makes using it in another program easy.

Editing features are functional but also somewhat limited. You can only edit controller data in the Event List, and the quantization feature is fairly basic. Besides Cut, Copy, and Paste, you can edit individual Events in the Piano Roll or create new Events, but that's about it. However, if you need more complex MIDI data in your Acid projects, you can easily import existing Standard MIDI Files. Moreover, Acid's painting and drawing tools are especially useful for entering MIDI notes.

Because Acid supports Yamaha's OPT (Open Plug-in Technology) format, you can also use third-party MIDI effects

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plug-ins, such as those made by GenieSys (www.genevoice.com). Unfortunately, OPT is not the format used by Cakewalk in Sonar and Steinberg in Cubase SX.

Like many other audio programs, Acid Pro 4.0a doesn't let you record the output of a soft instrument as you are playing it. Instead, you must record the MIDI Events that will control the soft synth or sampler, then render the track to a new audio file. The rendering process is much faster than real time, and once it is complete, you can paint or draw Events that play the file directly into the new audio track that will have been created.

I GET SURROUND

Acid Pro's new Surround Panner is a very efficient way to create a surround mix. Choose 5.1 Surround as the Master Bus mode of a new project, and you'll see a small surround-pan icon appear on each new audio track. You can move the track's position in the mix by dragging the icon or by accessing the Surround Panner window (see **Fig. 2**), which provides a larger surface on which you can create more precise moves.

In the Surround Panner window, you'll find an LFE tab that is used to send the track to the Low Frequency Effects channel. You can assign a lowpass filter (required by some surround authoring systems) to that channel using the Track's Audio Properties. You'll also find a slider to adjust the volume of the center channel, as well as a Smoothness control, which appears if you have keyframes on your track. Keyframes are used to automate a surround track's pan position and are created directly under the audio display in the main track window. Like breakpoints in envelopes, they determine the pan location at a specific point. Acid interpolates the correct values between each pair of keyframes using the same type of fade curves that it offers for stereo panning and volume control. According to the manufacturer, the Smoothness control affects how the sound moves between points on the pan curve. Because I didn't have a multichannel audio card in my review rig (a Pentium 4/2.4 GHz running Windows 98 SE), I couldn't test that feature.

The Surround features are very intuitive and easy to use, though it would be nice if the pan icon moved to reflect the current location of your sound as it plays back. Sonic Foundry has just announced the new 5.1 Surround Plug-In Pack for Acid, an AC3-encoding and DVD-burning utility, which will further enhance Acid's surround capabilities.

GET THE PICTURE

Acid Pro 4.0a adds several new features that will appeal to anyone working with animation or video. For starters, you can now preview video using pixels that are either square (used for computer-monitor viewing) or nonsquare (used for TV viewing). You can also encode a video and accompanying audio track into MPEG-1 format for use on a VCD or into MPEG-2 format for use on a DVD. The default encoding presets are fairly basic, but the optional Professional MainConcept MPEG Plug-In ($99) gives you a wide range of options. And, of course, you can preview video on an external monitor if you have the appropriate hardware.

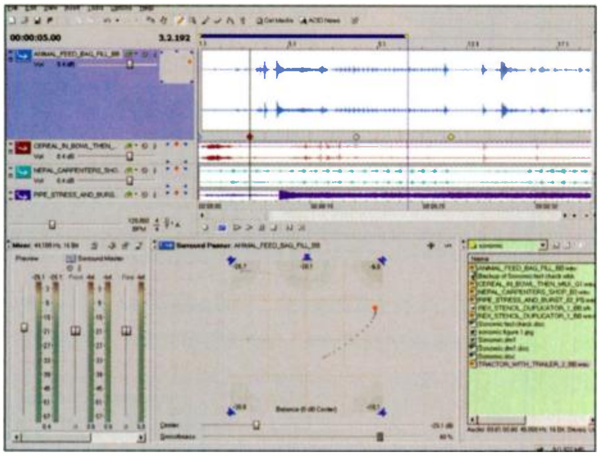


FIG. 2: The Surround Panner, shown at bottom center, gives precise control over the movement of audio tracks. You can automate a track's surround-pan position by using keyframes, which appear as small diamonds under the waveform in a track (see track 1 above).

Acid Pro 4.0a offers a number of other enhancements, such as new pan types (equal power and balance), optimization of the Beatmapper's time-stretching and pitch-shifting algorithms, Windows Media file support, and more customization options for default track properties. In the looping arena, you'll find support for alternate time signatures, from 1/1 to 99/32, as well as new options in the Chopper, which make assembling grooves even easier.

Adding major new features such as MIDI recording and editing and surround sound to a mature program such as Acid can result in a bit of a kludge. But Sonic Foundry has done an admirable job of integrating those new resources in a mostly efficient way. If you're already an Acid user, you'll really appreciate the major new tools that version 4.0a gives you. If you haven't yet joined the Acid community, now is a great time to sign up.

Dennis Miller *is an associate editor of* **EM.**

PRODUCT SUMMARY

Sonic Foundry
Acid Pro 4.0a (Win)
loop sequencer
$499
direct from Sonic Foundry $399
direct download $349
upgrade from 3.0 $149

FEATURES	4.0
EASE OF USE	4.5
DOCUMENTATION	4.0
VALUE	4.5

PROS: MIDI recording and editing. VST Instrument support. Excellent surround-panning capabilities.

CONS: Only one video track. Limited MIDI editing.

Manufacturer
Sonic Foundry, Inc.
tel. (800) 577-6642
e-mail customerservice@sonicfoundry.com
Web www.sonicfoundry.com

MESA/BOOGIE

RECTIFIER RECORDING PREAMP

Authentic live guitar tones without miking speakers.

By Michael Cooper

Mesa/Boogie's new Rectifier Recording Preamp is an all-tube (six 12AX7A dual-triode) DI recording solution for electric guitar, designed to emulate the sound of the company's Dual Rectifier Solo Head blowing through a 4×12 Rectifier cabinet. Although the "Recto" is first and foremost a direct-recording device, the 2U rackmountable unit also provides outputs to feed your guitar amp.

HEY, MR. TIMBRE MAN

Most of the Recto's action takes place on its front panel, where you'll find an unbalanced, transformerless instrument input and control knobs for two discrete channels (see **Fig. 1**). You can sculpt a clean tone suitable for chording on channel 1, for example, and a more aggressive timbre for soloing on channel 2, which offers hotter gain. Then you can switch back and forth between the two channels and sounds by manually flipping a front-panel switch or by stomping on the supplied footswitch's Channel button. A green or red LED lights to alert you that channel 1 or channel 2 is selected, respectively.

Each channel is served by a three-way Mode switch that gives you a starting point for fashioning your guitar tone. Channel 1 modes are labeled Clean, Fat, and Brit, while channel 2 modes are called Raw, Vintage, and Modern. The two channels each provide a row of gain- and tone-control knobs that further sculpt your timbre. Those knobs span the left and center portions of the faceplate and are labeled Gain, Treble, Mid, Bass, Presence, and Master. The Master control sets the relative volume of each channel, allowing you to set solo levels hotter than chording levels, for example. That scheme also lets you crank any channel's Gain knob to get more tube saturation without committing to higher-volume output.

You can further modify channel 1 tones by kicking in the Bright switch or –3 dB pad (which have no effect on channel 2). Activating the Bright switch boosts the highs above the band affected by the Presence control; it's mostly useful for recording applications, as it provides a little sparkle and air that you probably wouldn't notice when playing through a cabinet in a nightclub. The –3 dB switch (which pads the grid of the input tube) works best in situations where a cleaner tone is desired.

The Rectifier provides separate stereo pairs of balanced outputs (dubbed Record Out and Live Out) on its rear panel for direct-recording and externally amplified applications, respectively. You control their levels with two ganged-element potentiometer knobs (that is, one knob adjusts left and right levels equally for a given output) on the front panel. A third knob, labeled Live-Solo, provides an additional boost to the active channel at the Live Outs when you select the Solo function on the Recto's footswitch.

The Rectifier's sturdy, compact footswitch features LEDs that show the status of its Solo and Channel buttons—the only controls the footswitch offers. The footswitch connects to a front-panel jack on the Recto by way of a supplied cable that measures roughly five meters long and is fitted on both ends with ¼-inch TRS plugs.

Also on the Rectifier's front panel are a large, red power-status lamp and power and standby switches. Powering up in standby mode will extend the life of the Recto's tubes.

REAR OF THE RECTIFIER

As mentioned earlier, the Rectifier provides separate left and right Record and Live outputs on its rear panel (see **Fig. 2**). The outputs are all on balanced

PRODUCT SUMMARY

Mesa/Boogie
Rectifier Recording Preamp
guitar preamplifier
$999

FEATURES	4.0
EASE OF USE	4.0
AUDIO QUALITY	4.0
VALUE	3.0

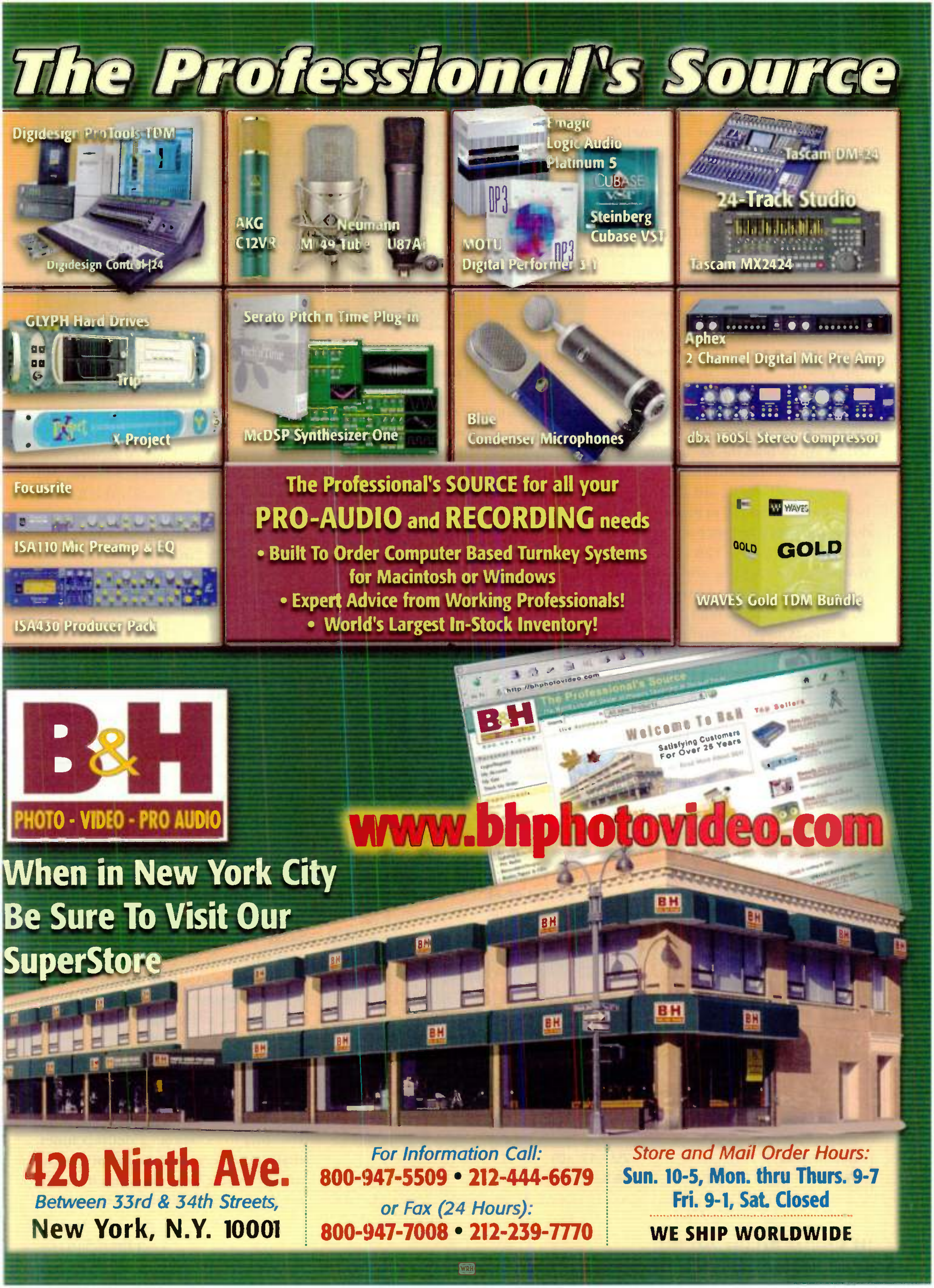
RATING PRODUCTS FROM 1 TO 5

PROS: Authentic, smooth tube timbres. Wide-ranging gain and tone controls. Two discrete channels. Separate pairs of outputs allow simultaneously recording direct and playing live through an amp and cabinet. Rugged construction. Supplied footswitch.

CONS: Knob positions are very difficult to see. Frequency response can be optimized for direct recording or externally amplified applications, but not for both simultaneously. Loading the unbuffered Live Outs causes a significant drop in signal level at the Record Outs. Spotty documentation. Relatively pricey.

Manufacturer
Mesa/Boogie
tel. (707) 778-6565
Web www.mesaboogie.com

FIG. 1: The Mesa/Boogie Rectifier Recording Preamp delivers creamy electric-guitar timbres for recording direct and for live performance.



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TRS jacks but will accommodate unbalanced lines as well. The frequency responses of lead-guitar (channel 2) tones from those two pairs of jacks are simultaneously affected by a front-panel switch that toggles between Live Bright and Warm Record settings. (The Live Bright/Warm Record switch has no effect on output signals derived from channel 1, however.) As you would expect, the Live Bright setting produces a brighter timbre most suited for playing through a guitar amp and cabinet, whereas the Warm Record setting produces a less edgy tone that's best used for routing directly to a recording console.

The Live Bright/Warm Record switch setting affects both the Record and Live outputs simultaneously, so the sound at one pair of outputs will always be inherently different. If you want to record direct and with a mic on your cabinet at the same time, you can choose the Warm Record setting and boost the treble band at your guitar amp to brighten your cab's sound.

An effects loop is located on the Rectifier's rear panel. The effects loop comprises a single send and separate left and right returns on electronically balanced TRS jacks and a wet/dry FX Mix rotary control. The loop is placed in the audio path just before the Record and Live outputs. For the highest fidelity, though, you'll want to forgo the effects loop and patch outboard gear in to your signal chain somewhere downstream from the Recto's Record or Live outputs.

The Rectifier's rear panel provides

FIG. 2: In addition to dedicated Record and Live outputs, the Rectifier's rear panel provides three footswitch jacks and an effects loop with a balance control.

two additional ¼-inch jacks; one is an input for an external device to switch the unit's two channels, and the other activates the Solo function. The owner's manual is vague about how these jacks work except to say that any MIDI switcher that uses "tip-to-ground, latching type logic" (such as the Digital Music Corporation Ground Control) will do the job.

Yet another ¼-inch jack (labeled Modern) provides an external trigger for Mesa/Boogie's Rectifier Stereo 2:100 power amp. The Rectifier's manual states that this jack accepts an "unshielded cable and any tip-to-ground latch type switching logic." When you patch this jack to the 2:100, the Recto's footswitch Solo button becomes a toggle for switching between channel 2's Modern and Vintage modes in live applications. With this setup, the channel 2 Master knob controls the level for Modern mode, and the Solo knob controls the level for Vintage mode. Because the footswitch's Channel button still toggles between channels 1 and 2, you gain footswitch access to three different modes (Channel 1, Modern, and Vintage) when you patch the Recto's Modern trigger jack to a 2:100 amp.

Also on the Rectifier's rear panel are a redundant instrument input, a ground-lift switch, and an IEC receptacle that accepts the supplied detachable AC cord.

TOTALLY TUBULAR

I first tested the Rectifier Recording Preamp with my '62 Fender Strat, patching the Recto's Record Outs to my Yamaha 02RV2's A/D converters. The Recto's wide-ranging controls make it easy to set optimal levels and attain a variety of guitar tones. The Recto can deliver everything from jazz to blues to hard-rock timbres. The Record Out control offers enough gain to easily attain 0 dBfs levels at the 02RV2's insert-return jacks. The Gain and Record Out controls can attenuate down to minus infinity (no sound). Unfortunately, it is very difficult to ascertain the position of any of the Recto's chrome knobs at an angle or in low light, as the hash marks on the knobs are not colored.

In a session with Portland, Oregon–based jazz guitarist Andy Guzie, the Rectifier's clean (channel 1) tones lacked the pristine sparkle and depth we were shooting for. Channel 1 delivers tones that sound very similar to what a high-quality live rig puts out when its tubes are driven lightly. If you're looking for a

Rectifier Recording Preamp Specifications

Inputs	(2) unbalanced ¼" TS, high impedance; (2) balanced ¼" TRS (effects returns)
Outputs	(2) balanced ¼" TRS (Record Outs); (2) balanced ¼" TRS (Live Outs); (1) balanced ¼" TRS (effects send)
Switch Inputs	(1) ¼" TRS; (3) ¼" TS
Power Supply	internal; IEC connector
Dimensions	2U × 12" (D)
Weight	16 lb.

good emulation of a clean guitar tone produced by an amp cab—that is, a sound with a little bit of grit and slightly muted highs—you'll probably love the Recto's channel 1 tones.

The Rectifier's channel 2 sounds great. Crank its gain to the max and the Recto rocks hard, dishing out plenty of smooth, warm sustain. With the Presence control set to about the 10 o'clock position (in Modern mode, with all other tone controls set to about the 1 o'clock position), the Recto lent my vintage Strat a warm, "brown" tone (borrowing from Eddie Van Halen's lexicon) that made me want to play for hours. And as every guitarist knows, a smooth, sustained tone can psychologically make your guitar strings feel silkier and help you play better. The Rectifier's smooth saturation characteristics are arguably the unit's greatest attributes.

Get more tube saturation without committing to higher-volume output.

POWER DOWN

Because the Rectifier's owner's manual was written from a musician's perspective, recording engineers will find that it lacks important documentation. Electronic specifications are almost totally absent.

The unit is less versatile than some competing products that cost less and also offer effects and compression. That said, the Recto's value lies in its uncanny ability to reproduce the authentic timbres of a tube-based guitar amp in a direct-recording environment. If that's the sound you want, the Rectifier Recording Preamp delivers in spades.

EM contributing editor **Michael Cooper** *is the owner of Michael Cooper Recording, located in beautiful Sisters, Oregon.*

BRAUNER

PHANTOM C

This hand-built boutique condenser mic is all about character.

By Myles Boisen

Swimming against the tide of assembly-line condenser microphones, German designer Dirk Brauner has been patiently introducing his handmade tube mics to this country for the past couple of years. The Phantom C is Brauner's newest offering, as well as his first solid-state condenser. Tuned specifically for use in studio-vocal and voice-over applications, this cardioid mic uses discrete FET (field-effect transistor) circuitry similar to that found in the Brauner/SPL Atmos 5.1 miking system, combined with a hand-built, large-diaphragm capsule.

The Brauner Phantom C has a full-bodied yet crisp and detailed sound that is especially flattering to male voices.

LET'S C

Clearly a precision unit, the Phantom C has a chunky, old-fashioned heft to it. Both the mic body and the permanently affixed shockmount boast a satin nickel finish, in contrast to the black finish on the attached swivelmount. The Brauner logo and "Phantom C" designation are handsomely engraved on the on-axis address side of the housing.

In case you missed the references, "Phantom" refers to the fact that this is a phantom-powered mic and the "C" denotes a cardioid-only polar pattern. Brauner's U.S. distributor, Transamerica Audio Group, indicated to me that a variable-pattern version, the Phantom V, is in the works.

The Phantom C boasts excellent technical specifications (see the table "Phantom C Specifications"), including low self-noise and high SPL handling. The unit's specs place it securely in the top ranks of high-output, low-noise mics for the digital age.

As with most premium mics, accessories are part of the appeal. In this department, the Phantom C doesn't disappoint. Its 9-inch-square carrying case is a cut above the competition, with a classy, brushed-aluminum look, rounded corners, a metal latch, and sturdy hinges. Interior features of the case include a pouch for documents, cables, and so on as well as extra-dense foam rubber cut precisely to grip and protect the valuable contents.

The cleverly designed and very sturdy shockmount is another classy touch. It consists of two concentric, C-shaped cages: the inner one snaps to fit—very snugly—around the mic body, and the larger, outer cage suspends the inner cage by six rubber O-rings. This leaves the front of the mic unobstructed, aiding close placement of a pop filter.

An all-metal swivelmount is attached to the rear strut of the shockmount, and the strut also rotates to allow side-to-side angling of the Phantom C. Despite the fact that the mic, after being snapped into place, cannot be moved within the shockmount, once the assembly is attached to a boom stand the mic can readily be angled in any direction, which aids positioning in tight spots.

C ME, HEAR ME

During sessions at the Guerrilla Recording studio, I scrutinized a pair of Phantom C mics on an assortment of vocalists. As you would expect from hand-built mics, there were some differences in frequency response between the two demo units (more on this later), which were numbered 87 and 88.

In general, I found that the Phantom C's presence boost is prominent and centered at a higher frequency than is the case with many other vocal condensers I've encountered. For a non-tube mic, it also exhibits significant warmth in the low mids around 300 Hz. Together, these factors serve to make the mic sound less sibilant, more pleasant, and just plain "bigger" on a variety of singers than many other vocal condensers.

However, on vocalists that don't need a presence lift—"airy"-sounding female singers, for example—the Phantom C's tailored response curve can sound too breathy or overly crisp in the lip-smack range (around 8 kHz). For the same reason, it can also sound slightly raspy on gravelly or overexerting male singers.

On a session with songwriter Ed Reiter, a throaty, country-inflected singer (I think of him as a grittier version of Chris Isaak), the Phantom C's presence boost worked like a charm to bring out enunciation and detail without sounding overly breathy. From the moment Reiter stepped up to the Phantom C, I knew its big, robust, signature sound would suit his voice perfectly.

As Reiter relaxed into his vocal duties, he hit the Phantom C with some very high SPLs. The mic never overloaded. Moreover, it maintained its commanding sound even with above-average

Drum Series I

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CD-ROM $149
Gigasampler, EXS, Kontakt, Akai, Kurzweil, Soundfont

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"Sonic Implants should be commended for providing a high-quality sample collection of these exotic instruments"
-KURZWEIL LAUNCHPAD

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Gigasampler, EXS, Kontakt, Kurzweil, Soundfont

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"The concept behind this CD-ROM is great: include the best-loved and most used Mellotron sounds with a variety of sounds from a classic analog string box and sell it for a nice low price."
-ELECTRONIC MUSICIAN

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"Much attention went into programming the various formats. Each takes advantage of the effects and envelope capabilities of the hardware or software sampler for its format, which is where the character lies."
-ELECTRONIC MUSICIAN

CD-ROM $149
Gigasampler, EXS, Kontact, Akai, Kurzweil, Soundfont

African Percussion

From the spiritual beat of Africa comes a sample library unlike any other. Sonic Implants African Percussion loads up your sonic pallete with no less than **34 unique African drums, shakers, bells and mallet instruments**. An incredible collection recorded at Blue Jay Studios through an SSL console. Partial list includes Bass, Boba, Djembe, Gome, Kaganu, Kepgisu, Kidi, Koboto, Dpanlogo, Sogo, Totodzi, Talking and Surdu drums; Afoxe, Axatse, Caxixi, Egg, Maracas and Shekere shakers; Gankogui, Kenken, Toke bells; Nigerian Logdrums and much more. Stereo samples, natural ambience, heavily multi sampled with up to 4 velocity samples per instrument.

CD-ROM $249
Giga, EXS, Kontakt, Soundfont

Strings MINI

Still a very full featured library, Symphonic Strings MINI weighs in at a hefty 7 CD-ROMs. A reduced rendition of the full blown collection, budget composers can now get access to the **best string library on the market for only $449**. Upgrades to the full version available. Visit website for demos and details.

CD-ROM $449
Giga, EXS, Kontakt, Soundfont

Strat & Fuzz

Widely regarded as **the best sampled Strat available**. Features Seymour Duncan pickups through a Fender Vibrolux amp, multi-velocity mutes, tones and chords so dynamic they pop right off your fingers and onto your guitar tracks. Creamy Fuzz Guitar features a custom built Strat through a Mesa Boogie, with FullTone SoulBender and FullDrive2 pedals to create a very soulful tone.

"This is the best sampled Strat sound I've ever heard... it doesn't just sound real, it plays real ...there is a difference!"
-BRAD LYONS Sweetwater Product Specialist

CD-ROM $79
Giga, EXS, Kontakt, Kurzweil, Soundfont

Symphonic String Collection

More than a year in the making, and involving the talents of an outstanding team of musicians, engineers, and Sonic Implants' own sound designers, the Symphonic String Collection is truly **a state of the art sample library**. Recorded by 2001 Emmy award engineer Antonio Oliart and RIAA award engineer John Bono, and featuring members of the Boston Ballet Orchestra and Boston Pops, the library has been designed for the utmost in **expressiveness and realism**. Rich multi dynamic instruments with a natural ambience intact make it a joy to create with. The Full collection comes on 20 CD-ROMs (or 3 DVDs), and includes **16 different articulations**, up to 4 velocity levels, down bow and up bows, release samples, loops on all sustained instruments, plus a huge variety of string effects. CD-ROM $995 Giga, EXS

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Phantom C Specifications

Element	pressure gradient transducer
Diaphragm	1", 6μ, 24-karat-gold sputtered Mylar
Polar Pattern	cardioid (fixed)
Frequency Response	20 Hz–22 kHz (±3 dB)
Dynamic Range	134 dB
Maximum SPL	142 dB (@ 0.5% THD)
Sensitivity	28 mV/Pa
Signal-to-Noise Ratio	86 dB
Self-Noise	8 dBA
Power	48V phantom
Dimensions	6.38" (H) × 2.00" (D)
Weight	1.36 lb.

amounts of limiting from a Langevin Dual Vocal Combo preamp. To combat a noticeable stridency in Reiter's full-voiced expression, I tried switching to the other Phantom C on hand. That mic (number 87), which I had come to hear as the "softer" of the two, served to tame some high-end raspiness, and also conveyed a smoother tone in the upper midrange. As a result, Reiter's passionate vocal track made it to the final mix with only minimal EQ.

C WHAT I MEAN

Overall, the Phantom C's round and crisp character seemed most well suited to male voices. My studio partner Bart Thurber, who used the two Phantom Cs on a few male rock vocalists, praised the mic in this application, describing it as "smooth, with a lot of character."

My experience using the Phantom C on female singers was more hit-or-miss. At best, the mic complemented lower female ranges, balancing ample tone with its trademark incisiveness. A classically trained female background singer on Reiter's project sounded good on both Brauner mics, taking on a mellow '70s timbre that blended perfectly with the male lead, thanks to the Phantom C's abundant warmth. The Phantom C also worked its magic on violinist Carla Kihlstedt's debut CD, *Two Foot Yard* (Tzadik Records, 2003), gracing a dense track on which Kihlstedt's whispered vocal had to compete with a potent palette of drums, accordion, and strings.

In some cases, though, the Phantom C's forward high end removed it from the running, usually prompting me to reach for a much mellower transducer. For example, singer Vanessa Lowe's first reaction after singing a few lines into the Phantom C was straightforward: "I can't stand the sibilance," she said. Actually, in that instance I hadn't found the Brauner's high end to be unbearable—but it was obviously not the best mic for Lowe's breathy style. (According to Brauner, since my tests the Phantom C has been further improved and retuned and now has an overall bigger response, resulting in an apparent reduction in brightness.)

I was always pleased with the Phantom C's detailed sound.

C IT THROUGH

Beyond its intended function for vocals and voice-overs, the Phantom C proved it can also earn its keep doing other duties around the studio. For example, I got superb results using the pair on a Hammond organ part, amplified through a Motion Sound KBR-M rotating speaker. Due to differences in the sound of the two mics, some EQ was required to provide a balanced stereo image (the two channels were panned hard left and right in the final mix). But the organ track had presence to spare, and I was impressed by how the subtleties of the part carried through to the final full-band mix.

During tests on acoustic bass, Japanese shakuhachi flute, bass drum, saxophone, and other sources, I was always pleased with the Phantom C's detailed sound, as well as its unusual ability to maintain presence, warmth, and focus at distances of 1 foot or more. These qualities are a testament to Brauner's expertise as a mic builder and should also make this microphone a fine choice for acoustic guitar, percussion, and piano.

C WORTHY

I gained additional insight into the Phantom C's sonic signature from standard loudspeaker tests. First, I compared the two mics by placing them side by side 20 inches in front of a full-range monitor in an acoustically dead room. I used Blue Kiwi and Monster cables to connect the test mics to a pair of Grace 101 preamps, and auditioned the sound through two channels of a Soundcraft Spirit board and Tannoy PBM-8 monitors. I used a 1 kHz tone to match the levels of all mic pairs. Once the pairs were matched, I used a variety of mixes to evaluate the frequency response of the mics.

Between the two Phantom Cs, it was immediately apparent that one of the two mics (number 88) had more bite in the upper mids and thus tended to highlight instruments such as trumpet, electric guitar, and snare drum in a mix. This same mic also had a fuller low-end response, and generally sounded hotter on a range of music sources.

Next, I swapped the XLR connectors on both mics to confirm that the sonic differences had originated at the microphones. Indeed, they had. However, the output levels of the two mics were very closely matched, as was self-noise.

Returning to test tones, I found that the two mics, when matched at 1 kHz, rarely matched exactly in other frequency ranges. I noted differences of 2 dB or greater in the high frequencies—

10 kHz and above—as well as at 200 Hz. Of course, these mics are not intended for use as matched pairs in stereo recording, and I am not evaluating them as such. But in light of the significant differences I heard, selecting one representative mic for comparison testing became an issue. After some deliberation, I chose the hotter-sounding of the two mics (number 88) for continued testing against a selection of mics from the Guerrilla Recording vault.

Up against an AKG 414 B/ULS, the Phantom C had much clearer upper mids and highs but seemed a bit sibilant. Low-end response was light compared with the 414, especially in the range of electric bass guitar. However, I should add that that particular 414, a favorite of mine for recording electric bass, has an atypically dark sound. Self-noise was comparable, and the Brauner's output was about 10 dB higher than the 414's.

The next comparison—with a Neumann TLM 103—showed the Brauner

PRODUCT SUMMARY

Brauner
Phantom C
large-diaphragm condenser microphone
$1,750

AUDIO QUALITY	4.0
VALUE	4.0

RATING PRODUCTS FROM 1 TO 5

PROS: Very flattering for many male vocals, especially rockers, rappers, and anyone in need of a little extra cutting power. Precision, handmade European quality. Low noise and high output. Well-made carrying case. Quality shockmount included. Also useful for nonvocal applications.

CONS: High-end presence boost can be excessive on breathy or raspy singers. Noticeable variation in frequency response between two test units.

Manufacturer
Brauner/Transamerica Audio Group (distributor)
tel. (702) 365-5155
e-mail sales@transaudiogroup.com
Web www.braunerusa.com

to be competitive in terms of certain specs: output and noise levels were closely matched. The low-end contours of the two mics were similar, though I found the Neumann to have a more solid punch in the bass and kick drum range. Again, the Phantom C sounded a bit sizzly in comparison, and it emphasized some unpleasant buzziness in a muted trombone passage. (Although I have often recommended the TLM 103 as an excellent all-around studio mic, it is too bright for my taste on some vocalists.)

In an attempt to find a match for the Brauner's distinctive and "toppy" tonality, I tried a few other mics before discovering an unexpectedly close mate: the Manley Reference Cardioid mic. This zippy $3,000 tube condenser can, in my experience, propel even the most mush-mouthed vocalist to the foreground of a mix. In comparison, the Brauner evidenced truer mids and lows and sounded quite warm and full. But it also came across as a bit harsh and metallic sounding compared with the Manley's smooth upper midrange. Engaging a 3 dB low-shelving cut on the Brauner made for a very close overall frequency match. The Manley had an astonishing 10 dB–higher output level, but the Phantom C was about 3 dB quieter in terms of self-noise.

COME C ABOUT ME

The Phantom C is a boutique microphone with a price tag that reflects its handmade heritage. Its cost may keep it off the shopping lists of budget-conscious engineers resigned to choose from a growing number of under-$1,000 European mics. But among those with an eye (and ear) for quality, this solid-state Brauner has a lot of things going for it. The shockmount, classy case, and flawless build quality of the Phantom C are significant selling points, not to mention Brauner's reputation as an uncompromising, quality-first designer.

The Phantom C is all about character; it is not a "flat" mic in any sense of the word. For that reason, it can't be expected to be a cure-all for every vocal-recording challenge. Even the world's priciest mics aren't perfect for every voice out there, after all. But the Phantom C's robust midrange and up-front attitude impressed me, even if I didn't always embrace its high-end sizzle (which Brauner has reportedly toned down a bit). I'd be happy to have a Phantom C in my mic closet to pull out for male voices, rockers, rappers, and any other singer in need of a little extra cutting power. And over time, I'm sure I'd find other instrumental uses for this fine microphone.

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

Quick Picks

ULTIMATE SOUND BANK

Plugsound, vol. 5: World of Synthesizers (Mac/Win)

By Jim Aikin

As a die-hard synthesizer programmer, I wasn't planning to be impressed by Plugsound, vol. 5: World of Synthesizers—but then I started checking out the sounds. Just about every preset I tried was an ear grabber. It may not be the most programmable software synth on my hard drive, but it's musically useful, and that's what counts.

World of Synthesizers ($99) is part of Ultimate Sound Bank's Plugsound series of VST plug-in Instruments. The plug-ins all use the same playback engine, but each product in the series has its own set of samples (more than 623 MB, in this case).

Top Guns

Ultimate Sound Bank created World of Synthesizers by sampling patches from fabled instruments, including the Minimoog, PPG Wave, Roland D-50 and JD-800, Yamaha DX7 and AN1x, Korg Wavestation, Ensoniq VFX, Clavia Nord Lead, and Waldorf Q. Sadly, the documentation doesn't reveal which sounds came from which sources.

The library's 512 patches are arranged in categories: Composite, Voices-Flutes, Tines-Bells, Synth Bass, Pads, Filter Sweeps, Analog Brass, Synth Leads, Keys, Texture-FX, and Short-Rezo. The Synth Bass set is by far the largest, with more than 100 sounds subdivided into several categories: Acid, Analog, Disco-House, Electro-Pop, Garage, RnB-Funk, Ragga-Reggae, Sub, Techno, Trance, Underground, and Vintage. Many of the bass and lead patches are set to mono mode so you can play legato lines with glide—a necessary feature that I appreciated.

Ultimate Sound Bank's Plugsound, vol. 5: World of Synthesizers offers myriad sampled classic-synth sounds in a self-contained plug-in format.

The sounds have just enough animation, with subtle high overtones that you can filter out if you need to. The multisamples in each preset match very well, and none of the samples are stretched so far upward that the animation gets twittery. A bit of chorusing or reverb is often sampled in, but it never gets overbearing. The looping is sometimes noticeable but always extremely smooth.

Stylistically, there's something here for everyone, although the accent is on mainstream pop and new age. Cranked-up leads and hardcore nastiness are in short supply, possibly because the Plugsound series has no built-in effects except for a reverb that was added with version 1.8. (You can download a free upgrade at www.usbsounds.com.) Also, there are no drums.

Getting Down with It

To put World of Synthesizers through its paces, I launched Steinberg Cubase VST and loaded a drum groove into Spectrasonics Stylus (which uses the same underlying synth engine, although the user interface is different). Plugsound is not multitimbral, so I loaded five instances and shortly had a synth-heavy hip-hop riff going with Plugsound playing bass, some bell-like chords, a cutting lead, and a couple of sparse background lines. On my 1.5 GHz machine, I was still using less than 20 percent of the CPU, even after I added a reverb plug-in to the lead. That's one of the advantages of Plugsound: because complex sounds are played back from samples, they conserve CPU cycles.

The downside is that you can't do much to customize the sounds. The programming controls in Plugsound are bare-bones, though still quite useful. You get ADSR envelopes for filter and amplitude; a choice of four filter modes; knobs on the filter for cutoff frequency, resonance, key tracking, and envelope depth; and an LFO that can modulate panning, pitch, amplitude, or filter cutoff. Other than rate and depth knobs, the LFO has no controls—no waveform choices, for example, and certainly nothing as sophisticated as clock sync. (According to the distributor, clock sync is coming in a future update.)

The mod wheel doesn't even use the main LFO. It adds vibrato from an invisible LFO, which is set much too fast for my taste. The setting is identical on all World of Synthesizers patches. You can't change the filter cutoff's Velocity response, either, but I didn't feel that was a problem, because the response felt good.

In addition to the main voice filter, Plugsound includes something called a UVI filter, which has its own mode or cutoff and resonance sliders. On my system, the filter proved to be something of a CPU hog, but fortunately it isn't needed for most sound-shaping tasks.

Boot Up, Plug In, Jam On

I might prefer some other synth for situations where I need pinpoint real-time control over the sound, but for meat-and-potatoes jobs—from comps and atmospheres to throbbing bass and spicy lead licks—Plugsound, vol. 5: World of Synthesizers is going to get a workout. I can lay down a track while the inspiration is fresh: all I have to do is grab a preset and go.

Overall EM Rating (1 through 5): 4
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SONIC FOUNDRY

CD Architect 5.0 (Win)

By Dennis Miller

It's back! After being pulled from the market more than two years ago, Sonic Foundry's CD Architect ($299, retail; $209.97, direct download; $239.96, boxed) has returned in a new and improved version. Formerly available as a plug-in for Sound Forge, CD Architect 5.0 now runs standalone and has significantly more features than the original. With support for 32-bit, 192 kHz audio, nearly two dozen included audio effects, and sophisticated CD-layout and -burning tools, CD Architect comes very close to being an all-in-one audio-mastering and -production platform.

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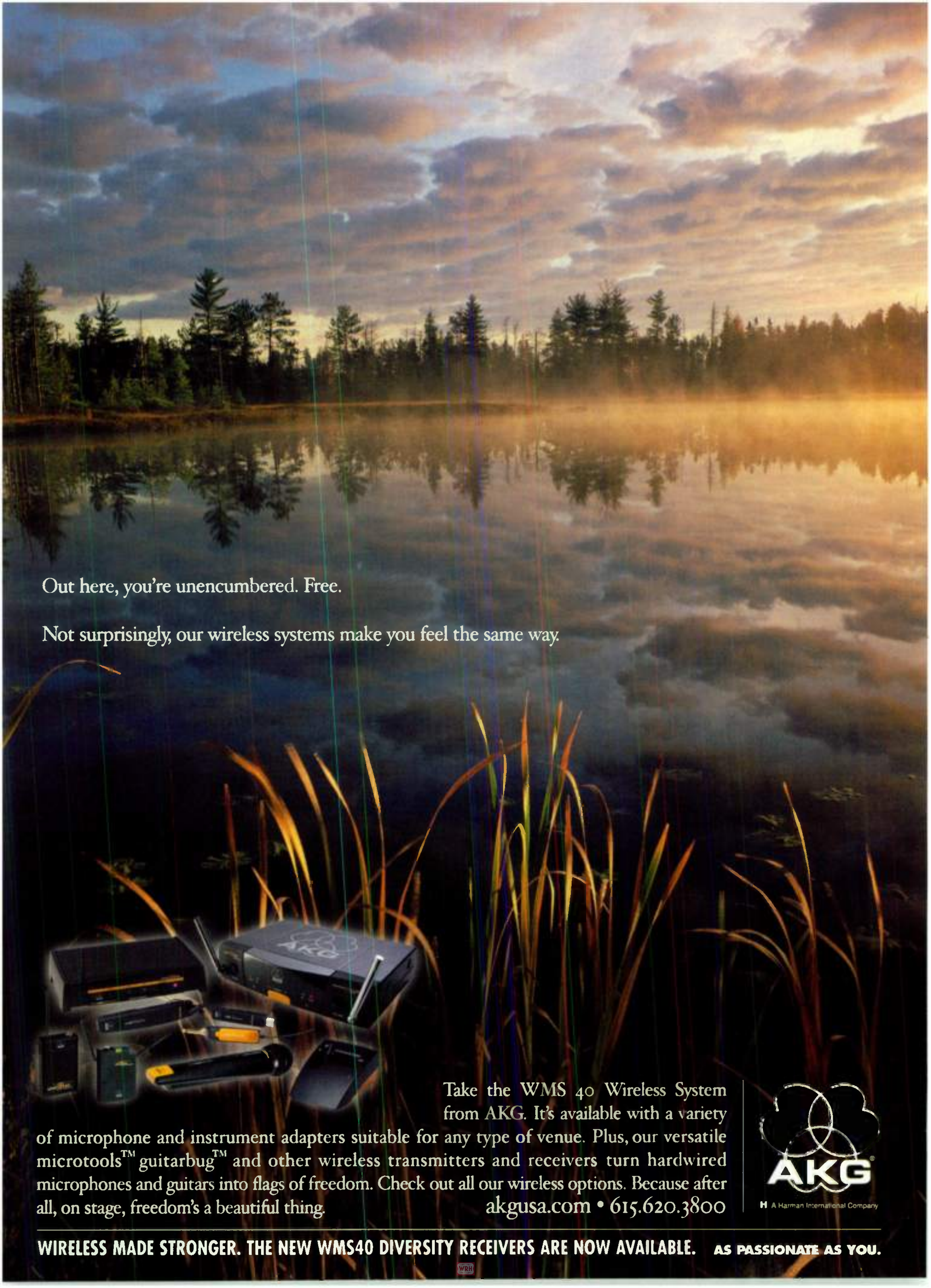

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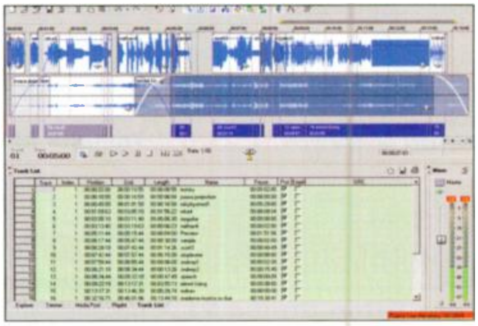
combines features from Acid, such as real-time pitch-shifting and time-stretching, with two stereo tracks of audio-editing and -mastering features from throughout Sonic Foundry's product line. Acid or Vegas users will feel at home with CD Architect's interface, which is clearly modeled after those programs.

Split It Up

As in the earlier version, CD Architect's interface is split among several main windows. The Timeline is a graphic display where you'll arrange the Events that will become the tracks on your CD. Events are complete audio files or selected portions. You can overlap multiple files, even if they have different sampling and bit rates, to form your tracks. At the bottom of the Timeline is the CD Layout Bar, where you'll see small, blue rectangular boxes that show the name and duration of the tracks in your project. By dragging those along the Timeline, you can easily rearrange the material on your CD. You can also type a new Start point into a track box's Properties dialog to change its position—very handy for making fine adjustments.

The Playlist is a text-based screen that resembles the Event Editor found in many sequencers. In the Playlist, you'll see a list of all the Events in your project along with their start and end times, durations, and various other parameters. There's a column to enable normalizing individual Events; as elsewhere, you can modify a parameter by simply typing a new value into the appropriate field. You can't, however, drag in the box to change values, which would be useful.

The third major window is the Tracklist, which displays many of the same parameters as the Playlist, but its properties reflect information about entire tracks. Among other options, you can drag single or multiple tracks to new positions, set the Pause time between tracks, and toggle Protection and Emphasis. Right-clicking on a track's name lets you preview the track or



Sonic Foundry's CD Architect 5.0 is a standalone application that offers considerable resources for mastering and authoring CDs.

jump directly to its position in the Timeline, or both. If you choose Zoom, the entire Timeline window will zoom to fit the length of the track. That type of integration makes configuring a CD project quick and efficient.

More than Core

In addition to its core mastering and burning features, CD Architect offers several editing options and authoring utilities to ensure that you get the job done well. For example, it includes 20 real-time DirectX effects that you can apply on an Event-by-Event basis. Other features include real-time time stretch—just hold down the Control key and drag an Event to the desired new length—and multiple fade types that you can use with the Automatic-crossfade function. You can also print or export your tracklist in a variety of formats, though you can't design CD covers from within CD Architect—at least not yet! And of course, nearly all current CD drives are supported, including USB and FireWire models.

Sonic Foundry has recently released a plug-in for Acid that allows it to burn multichannel audio onto a DVD. It would be great to see that feature added to CD Architect. In addition, adding a few more stereo tracks to the interface would be useful and might even allow users to work entirely within the program—as it stands, you'll probably want to keep your main audio editor on hand.

Much has changed in the CD-burning world since CD Architect was last on the scene. Today, multitrack audio editors such as Steinberg WaveLab and Magix Samplitude, not to mention Sonic Foundry's own Acid software, offer sophisticated CD-burning capabilities. Moreover, consumer programs such as Roxio Easy CD Creator and Nero 5.5 offer ever more high-end functions. Yet with its professional CD-authoring and -burning features, included DirectX effects, and efficient (and customizable) work flow, CD Architect 5.0 should assume a comfortable position in its niche. If you're not happy with your current CD-burning tools, give CD Architect a try.

Overall EM Rating (1 through 5): 4

Sonic Foundry; tel. (800) 577-6642; e-mail customerservice@sonicfoundry.com; Web www.sonicfoundry.com

EASTWEST

Aerosmith's Joey Kramer Drum Loops and Samples

By Dan Phillips

The seminal band Aerosmith broke new ground by combining funk grooves with hard-hitting rock, and the band's distinctive sound was due in large part to the kit work of drummer Joey Kramer. In this loop collection from EastWest, Kramer teams up with Aerosmith producer Marti Frederiksen to deliver an exciting assortment of rhythms à la carte.

The *Joey Kramer Drum Loops and Samples* library is available in four versions: as two audio CDs ($89.95), two CD-ROMs in Akai format ($179.95), and as a ten-CD set containing multitrack Pro Tools and Acidized WAV versions ($299). The Pro Tools and WAV versions each consist of eight CDs of multitrack material along with the two audio discs. The Pro Tools version also works with any Mac program capable of reading Sound Designer II files; I reviewed the library using Mark of the Unicorn's Digital Performer.

Groovy, Baby

The different library versions all include 34 "grooves." Each groove is presented in several variations, including simple-busy, hats-ride-cowbell, fills, and so on. A groove might have as few as 2 variations or as many as 30.

Grand Canyon, for instance, includes 24 variations with 8th-note and 16th-note hi-hat patterns, closed and half-open hi-hats, cowbells as "ride" (simple or busy with and without tasty hi-hat accents), sparse and busy kicks and snares, and intro and outro fills.

The audio CDs provide stereo versions of all the loops (about an hour's worth of material) and also include a small collection of single hits. The CDs are essentially the same except that the first disc (labeled Dry) has only the natural room sound, and the second (labeled Wet) adds reverb. For my taste, the Dry loops have plenty of natural ambience; the Wet loops seem a bit boomy. Still, it's always nice to have options.

Multitracks

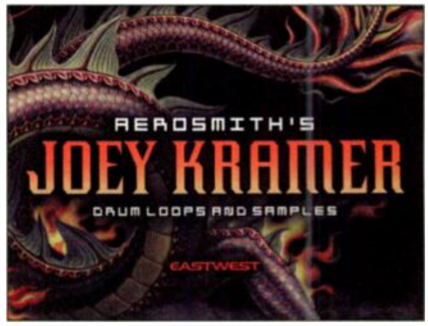
The multitrack versions offer a substantial 2.9 GB of data. All of the stereo mixes are included, of course, but the real gold is in the multitrack masters, which provide separate tracks of mono kick, snare, and hi-hats, along with stereo toms and stereo room ambience. Two of the grooves, Grand Canyon and Jelly Box, include a mono cowbell track.

With the multitrack grooves, you can import the loops into your favorite DAW and mix the rhythm tracks to suit your taste. Simply adjust the levels, EQ, compression, and other effects on a track-by-track basis. On the other hand, these recordings are so good that you may not want to mess with them very much—adjectives like *thick, full,* and *slamming* come immediately to mind.

It's also worth noting that you shouldn't expect the sort of track separation you'd get with MIDI drum samples. These are live recordings of an acoustic kit getting a serious workout; Kramer clearly shows the skins no mercy, and there's plenty of microphone leakage to prove it.

High Style

As good as these recordings sound, of course, it's Kramer's expertise behind the kit that really makes them shine. Although he can certainly get fancy when he wants to—with facile hi-hat work and subtle snare ghosting—some of my favorite grooves are deceptively simple.



Aerosmith's Joey Kramer Drum Loops and Samples **from EastWest offers 34 distinctive grooves with dozens of variations for a wide range of styles.**

Gorilla Salad, Garden Gate, and Saddle Bag, for example, are based on the most basic rock beats: snares on 2 and 4, eighth-note hi-hats or toms, and simple kick patterns. However, Kramer's renditions are anything but bland; they're a full-out assault, driving relentlessly forward without pause or hesitation.

My other favorites include Crown and Feathers, a 12/8 pattern with ghost snare notes and a beautiful feel, and the almost gentle (for Kramer, at least) cowbell grooves of Grand Canyon.

My only serious gripe is in the area of documentation. Although a few tracks are identified as fills, it would be handy to have more complete descriptions of the variations in the written documents and in the file names. File selection would be easier if you could see which grooves had intros, outros, cowbells, toms on eighth notes, and so on. Moreover, the file names show only the track information (hi-hats, snare, kick, and so on) and not the name of the groove itself, which makes things a little tricky if you try to use several grooves in a single project.

However, these minor complaints don't even come close to stopping me from recommending this collection. The material is nothing short of fantastic, combining first-class recordings with phenomenal playing. If you're looking for hard-hitting rock loops, put this one on your shopping list.

Overall EM Rating (1 through 5): 4.5
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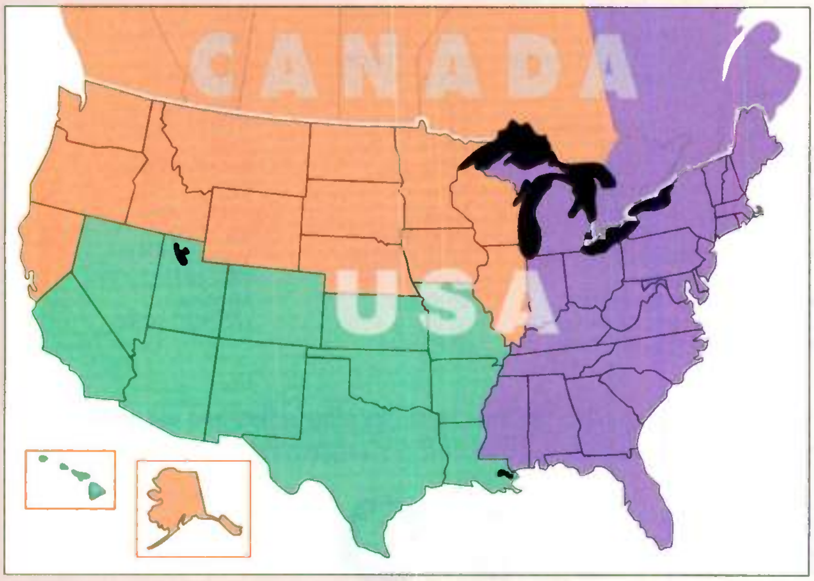

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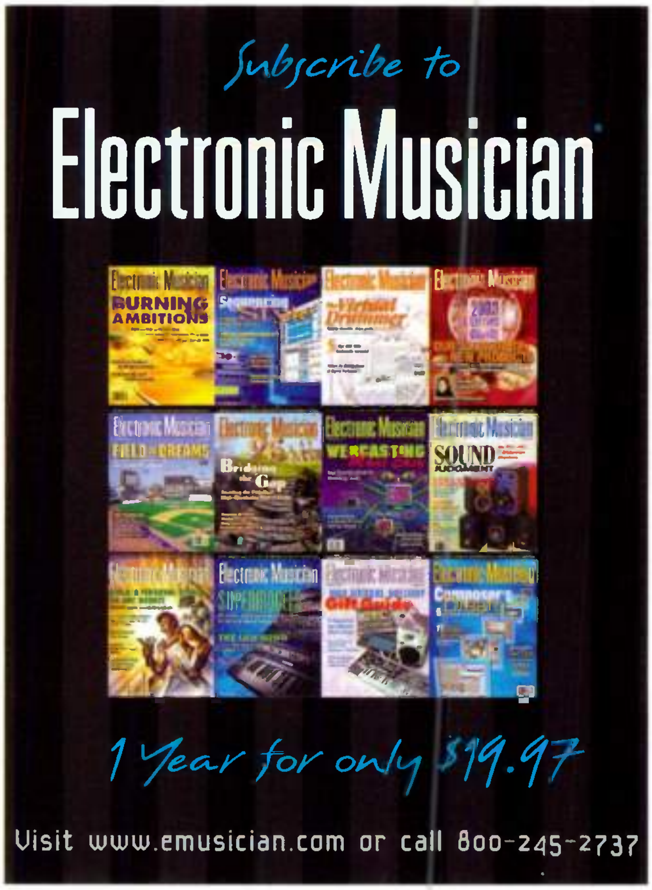
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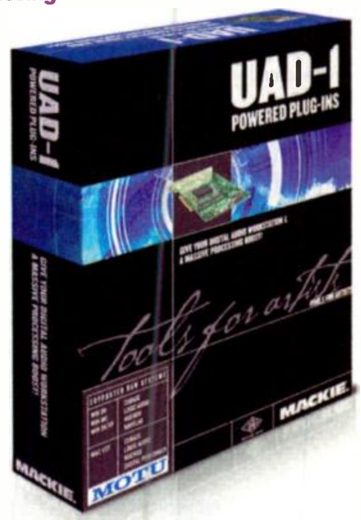
UAD-1 Version 3 is here, with multi-card support and many other enhancements. Run dozens of sophisticated effects plug-ins inside Digital Performer without bringing your Mac to its knees. What's the secret? UAD-1 is a custom DSP-equipped PCI card. It's like adding an extra $20,000 worth of effects gear to the dozens of native plug-ins included with DP. UAD-1 ships with a growing list of powered plug-ins, including Nigel, a complete palette of guitar tones combined with every effect a guitarist could ever need. Authentic vintage sounds include the Pultec Program EQ, a stunningly realistic recreation, and the1176LN Limiting Amplifier and Teletronix LA-2A Leveling Amplifier, two more analog classics reborn inside Digital Performer. Apply liberally with host CPU cycles to burn.



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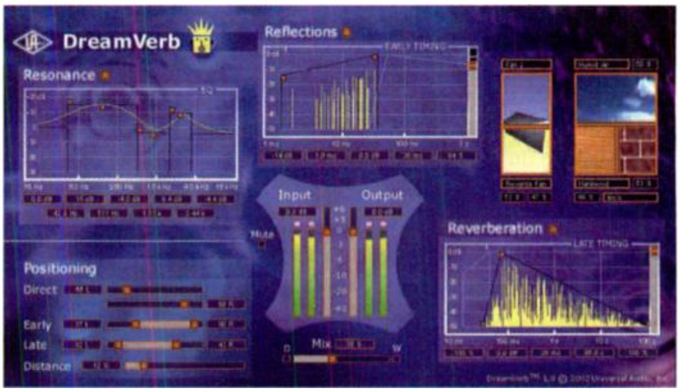

DreamVerb's unparalleled flexibility, power and intuitive interface allows you create any acoustic space inside Digital Performer using a huge list of different materials and room shapes. Blend or "morph" different room shapes and surface materials, adjusting the effect of the room materials on the sound space, and even varying the density of the air to simulate different ambient situations. With flexible 5-band active EQ and unique level-ramping for the early and late reflections, you can create sounds ranging from ultra-realistic dynamic room simulations to lush reverb effects. Universal Audio's proprietary smoothing algorithm lets you adjust parameters in real-time with no "zipper-noise" or other undesirable artifacts.

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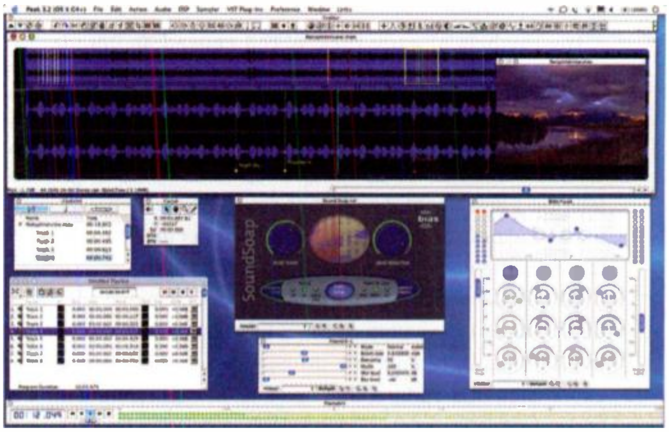

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FINAL MIX

By Larry the O

Does Sex Belong in Music?

Absolutely. No question about it. Case closed. Good night, ladies and gentlemen. See you next month.

(Okay, picture this: I've *just* finished clicking the Send button on this month's "Final Mix," and in less time than it takes a plug-in to process a four-letter word, an e-mail from the "First Take" guy pops up in my In box, pointing out that while the publisher will be tickled pink to have the extra ad space to sell if my column ends after five short sentences, the "Letters" column, which he, as editor, must deal with, will be a living hell.)

Well, I suppose there's always more to say about sex, isn't there?

The rhythms of early rock 'n' roll caused an uproar because, it was alleged, it inspired prurient interest within Our Youth and caused them to engage in obscene dancing and other sexually explicit behavior. But although the music may have embodied, and even crystallized, a movement toward more overt sexual behavior, the actual causes sprung from much bigger and broader issues in the social landscape.

Today, few people raise objections to the sexual suggestiveness of rhythms or any other aspect of instrumental music. Music is a primary as well as primal expression of a culture, and besides, sexual content in music certainly predates Ed Sullivan's conniption over the motions of a Memphis truck driver's danger zone.

From its very beginnings and up through the big-band era, jazz had many distinctly sexual elements, from the nasty wah-wah trumpet of Bubber Miley to the wild frenzy of the young Gene Krupa. Krupa's music is as good a reminder as any that the drum—probably the world's second-oldest musical instrument (next to the voice)—has carried a sexual message since the dawn of humanity.

Expressions of sex through instrumental music draw their fascination and effectiveness from the fact that the instruments themselves are physical in nature. This is expressed not only in the way that musicians play their instruments but also in the motion that is implied by a slowly bent note, the bump and grind of a pounding tom, or the naked lust of a brash, spitty trumpet.

The voice, on the other hand, while capable of all of the suggestiveness of an instrument, can also make direct verbal expressions of sex. That is where a lot of controversy lies.

Personally, I love lyrics that are sexually suggestive—the more so the better. The blues tradition offers especially fruitful pickings, whether it is B. B. King singing "I got a sweet little angel; I love the way she spreads her wings" or Lowell George growling about the rocket in his pocket.

On the other hand, I have very little use for sexually explicit lyrics. Not that I am offended by explicit language; I just don't find it very sexy outside of the bedroom.

My view is that the elements of hip-hop lyrics that some find objectionable stem more from underlying attitudes about power and misogyny than from sexual explicitness.

If the question is where should one draw the line, then the answer is easy: wherever you wish. The question of where the law draws the line is where societal pugilism comes to the fore. To even ask the question invokes the never-ending fun of battles about free speech and censorship versus community values and decency. The incredible penetration of media into the daily lives of people of all ages greatly magnifies the issue.

However, this an issue of public language, not of sex in music. Those who are offended by explicit lyrics would likely also pose the same objections in all arenas outside the musical context. Remember Lenny Bruce?

Sex at its best is one of the highest expressions of love a person can experience, and music is uniquely capable of expressing that. Sex at its worst is something I'd rather not experience at all, but it can make a great song. That alone is a good reason for sex to belong in music.

We welcome your feedback. E-mail us at emeditorial@primediabusiness.com.

STEVE CURL

Introducing the 828mkII

24-bit 96kHz resolution. DSP-driven mixing and monitoring.
Front-panel programming. Stand-alone operation.

828mkII feature highlights

- **CueMix DSP™** — the 828mkII delivers DSP-driven digital mixing and monitoring for all 20 inputs. Connect mics, guitars, synths and effects processors, and monitor everything from the 828mkII's main outs, headphone out or any other outputs with no separate mixer needed.
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- **Support for 96kHz ADAT optical digital I/O** (S/MUX) — provides 4 channels at 88.2 or 96 kHz.
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