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PAT METHENY

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Behind the Scenes

In this month's cover story on Pat Metheny's new *Orchestrion* album ("A Well-Oiled Machine," see p. 24), I mention visiting his rehearsal space, an old church in Brooklyn, where he was getting ready for the tour. Here's a little more of the back story.

I was there, along with a number of other reporters and photographers, for a press briefing about the project. We were first shown into an outer room of the church (the same room where they shot the album cover photo); after a bit of waiting and mingling, Metheny walked in and gave a talk about the history of orchestrions and how he had become interested in the subject. He showed us a coffee-table book about orchestrions, which had photos of vintage ones from the late 1800s. It was clear that Metheny really has a passion for these old self-playing instruments, and that's what led him to this incredibly ambitious project.

Next, he took us into the main room of the church, which was the rehearsal area, where all of the robotic instruments were set up. It's also where the opening photo of the story was shot, so you can get an idea of what it looked like. What you can't see from that photo is all the stuff off to the sides. There were huge road cases, computers, Metheny's old Synclavier (I'm not sure why that was there), and cables everywhere.

If I didn't know better, I would have thought it was a workshop of some mad musical scientist. But there was actually plenty of method to the madness, and Metheny and his tech crew were busy working on how to take this unique show on the road. As I discuss in the story, there were a lot of obstacles they had to overcome, and a plenty of unknowns about how the instruments would handle the rigors of the road.

Seeing the instruments themselves close up was very cool. Imagine a vibraphone and a marimba with a mallet on every note. Or an acoustic guitar with a little wheel with six picks on it—the wheel designed to spin around and pick the strings as it receives note-on and note-off messages that are translated by its electronics from MIDI data sent by Metheny's guitar or the computer. Or a drum set, separated into its various parts, all hanging on a rack with sticks attached by mechanical contrivances, ready to strike. It was a memorable visit, and I certainly recommend you check out the album and go see the tour if you can.

On another subject, I wanted to mention that we have a new back page column debuting this month called "Back Talk." In it you'll get a quick-hitting Q&A with a musician, artist, or producer. This month, it's a chat with Simon Franglen, one of the synth programmers for *Avatar*. He talks about the production of the music for that movie, composer James Horner's complex score (one cue had 200 tempo changes), and the synths that he used. For more of the interview with Franglen, you can go to emusician.com, where we have a podcast posted. Wherever possible, we hope to provide supplemental interview material online for our "Back Talk" interviews.

This new column is just one of several that we'll be introducing during the next couple of months. I'm very excited about them, and I think you will be, too. So stay tuned.

Mike Levine
Editor

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drums

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Download of the Month

Progress Audio Kinisis By Len Sasso

Progress Audio's (progressaudio.co.uk) motivation for developing its new plug-in synth reads, "Because ADSRs are so last year." A few minutes playing with Kinisis (Mac/Win, $110) will convince you the company is right. The one-screen user interface is divided horizontally into three sections: synth parameters at the top, the timeline in the middle, and global parameters at the bottom. Changes you make to any of the controls in the synth section apply at the current point on the timeline. Whenever you play a note, the sound morphs through the timeline settings. (Global settings such as effects parameters, voice mix, and so on apply across the whole timeline.) You can also freeze the timeline for a static sound, which is convenient for editing, and you can manipulate the frozen time position manually or from a MIDI controller.

A Kinisis patch has as many as four layers, whose mix is managed in the global section. (The layers are called Voices, but should not be confused with polyphonic voices, of which you get as many as 16.) The synth panel has a separate on/off switch and a programming tab for each voice, but they are all programmed in the same way using the timeline. With four layers, a single patch can get quite active (see Web Clip 1).

The signal path is standard fare—oscillator, filter, amp, and effects. The oscillator holds four waveforms, which you mix with an X/Y controller that is managed on the timeline. You can load waveforms from Kinisis' browser, draw in your own, or have Kinisis extract a single-cycle waveform from any WAV file. A five-row modulation matrix lets you route two built-in LFOs, along with MIDI note-number, velocity, mod wheel, and aftertouch to other synth parameters.

Download the demo of Kinisis from the Progress Audio website. You'll also find audio examples, an excellent manual, and a brief video, which will give you a good feel for how easy this synth is to use.

≫ OPTION-CLICK By David Battino

Visualize Your Music

Add a trippy video to your music and post it on YouTube

YouTube is today's music showcase, but what if you don't have video skills—or even a camcorder? Here's a cheap and easy way to make striking videos from your music. Just load your song into your favorite media player (iTunes, Winamp, etc.), turn on a visualizer, and record a screencast of the swirling, synchronized lights (see Web Clip 1).

Screencasts are like videos shot from inside a computer. They show what's happening in an area of the screen while capturing the audio—usually a mumbling lecturer—in sync. TechSmith Camtasia Studio (camtasia.com, $299) is the standard on Windows, whereas Ambrosia SnapZ Pro X (ambrosiasw.com, $69) and Telestream ScreenFlow

(telestream.net, $99) are big on the Mac side. There are also numerous shareware options, and I've found shinywhitebox iShowU for Mac (shinywhitebox.com, $20-$30) works well. However, Camtasia's built-in editor lets you trim dead space and add text. Songwriter Gina Fant-Saez (blueworldmusic .com) even adds synchronized

Recording the iTunes visualizer in Camtasia makes a trippy music video. See YouTube's help (bit.ly/9JdjWW) for the recommended specs.

text lyrics to her demos. —*David Battino, Batmosphere.com*

This Month on Emusician.com

PAT METHENY

Need more Metheny? Watch a video of "heads" with his "band" and listen to a podcast with more of the interview.

COMPOSER PROFILE

Available only at emusician.com is our new monthly series, in which we check in with a composer—for soundtracks, films, videogames, commercials, etc.—to find out what's in his/her gear arsenal and new projects. This month: Kusiak Music.

THIS MONTH'S SOUNDTRACK

By Mike Levine

These releases cover a lot of ground, genre-wise, including heavily improvised electronic instrumentals, classically influenced art-rock, guitar-oriented blues rock, and more.

THESE NEW PURITANS: *HIDDEN* (DOMINO)
Ambitious and highly original art rock featuring brass and woodwinds mixed with synthy electronic parts, Taiko drums laying down massive beats (and lending a cinematic feel), and imaginative and complex arrangements. Hard to explain, but cool to listen to.
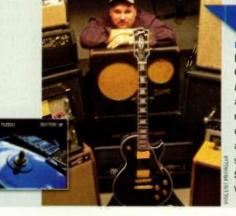
COBBLESTONE JAZZ: *THE MODERN DEEP LEFT QUARTET* (!K7)
Using old analog drum machines and modular synths, and playing tunes that span a range of electronic styles, the band—now expanded to a quartet—developed the material on this album through jamming, experimentation, and real-time, manual effects processing during mixdown.

RICH TOZZOLI: *RHYTHM UP* (R. AUSTIN MUSIC)
Guitarist, producer, and occasional *EM* contributor Tozzoli offers up a tasty collection of impeccably recorded guitar instrumentals, both electric and acoustic, on which he's accompanied by a number of session stalwarts including drummer Shawn Pelton and bassist T Bone Wolk (who died on Feb. 27).

ERNEST GONZALES: *BEEN MEANING TO TELL YOU* (FRIENDS OF FRIENDS)
Gonzales, a veteran DJ and the founder of Exponential Records, offers up a set of instrumentals that blend electronic and rock instrumentation in a beguiling sonic stew.

HIM: *SCREAMWORKS: LOVE IN THEORY AND PRACTICE, CHAPTERS 1-13* (SIRE/REPRISE)
The seventh album for the Finnish quintet features fat production of the band's high-energy, melodic rock. How can you go wrong when your drummer's nickname is "Gas Lipstick"?

AVATAR SYNTH PROGRAMMING
For more of the interview with Simon Franglen, one of the synth programmers who worked on the *Avatar* soundtrack, listen to this podcast.

IN THE STUDIO WITH MING: VOLUME 9
In his latest video, producer Aaron "Ming" Albano shows how he used Sugar Bytes Effectrix, an effects-sequencing plug-in, on a remix of the Lady Gaga song "Telephone."

By Len Sasso

MOTU DIGITAL PERFORMER 7.1

MO' TU LIKE

Digital Performer 7.1 (Mac, $795; free update for DP7 users) brings a passel of new features and plug-ins to MOTU's (motu.com) flagship DAW. The Tuner plug-in lets you tune guitar, bass, sax, and other pitched **solo** instruments with adjustable tuning reference and test-tone generation. Auto-Save will save your original or a backup copy at specified time intervals. The new Custom '59 Series of four amp models delivers preamp, vintage, classic, and modern sounds. Track folder management is enhanced with an overview inspector; contents cut, copy, and paste; and controls for play-enable, solo-exempt, lock, and more. You also get more keyboard shortcuts, faster MIDI and automation data editing, many new plug-in presets, and support for Apple Core Audio Format (CAF) files.

JDK AUDIO R22   TWENTY SOMETHINGS

JDK Audio (jdkaudio.com) has released a dual-channel, rackmount version of the compressor circuit found in API Paragon mixing consoles. The internally powered R22 ($1,195 MSRP) provides the high-frequency protection of the pat-ented Thrust switch; variable threshold, ratio, and make-up gain controls; switchable VU metering (output or gain reduction); and channel linking with RMS power summing for stereo compression. Specs include XLR and ¼-inch balanced inputs and outputs, 20Hz to 50kHz bandwidth, -88dBu signal-to-noise ratio, 1:1 to 10:1 compression range, and 0dB to 20dB makeup gain. The R22 is billed as the perfect companion to JDK's R20 mic preamp and R24 EQ.

BIG FISH AUDIO MIROSLAV VITOUS STRING ENSEMBLES

HIGH STRUNG

Powered by Yellow Tools Engine and delivered standalone and in VST, AU, and RTAS plug-in formats, Miroslav Vitous String Ensembles (Mac/Win, $1,399.95 MSRP) string collection contains two ensembles: a large orchestra (24 violins, 14 violas, 12 cellos, and nine basses) and a chamber ensemble (14 violins, eight violas, five cellos, and four basses). You get individual control of each string section, along with multiple articulations such as long sustained notes, détachés, legatos, staccato, tremolo, sforzando, sul pont, etc. Ease of use is emphasized in the library's design: Just load a patch and start playing.

ARTURIA ORIGIN KEYBOARD

Arturia (arturia.com) describes its new Origin ($3,499 MSRP) keyboard as the first laptop synthesizer. The unit combines the original Origin synth and a

BACK TO THE FUTURE

61-note, velocity-sensitive keyboard with aftertouch. The keyboard also sports a ribbon controller and the omnipresent mod and pitch wheels. The Origin synth is hinged to the back of the keyboard and folds down for protected transport. It sports a 5.2-inch LCD, along with numerous knobs, buttons, and LEDs for programming, control, and preset recall. There's even a joystick for enhanced real-time modulation. The Origin draws on Arturia's popular soft-synth models: Moog Modular, ARP 2600, CS-80, Jupiter-8, Minimoog, Prophet VS, and tonewheel and rotary organs.

Sound Advice

pop styles. You get as many as 30 velocity layers of separate left- and right-hand samples with as many as six round-robin variations.

Loopmasters
Danny Thompson Bass Samples

Danny Thompson brings 50 years of playing experience with such music greats as John Lee Hooker, Rod Stewart, and Ginger Baker, along with his prized French 1865 Gand upright bass, to the gig. Loopmasters (loopmasters.com) delivers the results, recorded using Neumann U47 and KM84 mics, Focusrite and RCA mic pres, and a bit of Neve compression and EQ. Danny Thompson Bass Samples (about $56, download) comprises 374 Acidized WAV, Apple Loops AIFF, and REX format loops and lines between

Danny Thompson
Bass Samples

70bpm and 160bpm in a variety of keys. You also get bass sample libraries for HALion, Kontakt, EXS24, NNXT, and SFZ samplers, including velocity-layered fingered, bowed, and effects instruments (see Web Clip 1).

Tonehammer Shake Vol. 1

The folks at Tonehammer (tonehammer.com) call Shake Vol. 1 ($49, download) "the fuzzy, buzzy, sandy-sounding chuff, shuck, and jingle of handheld shakers and bells." For this collection of 46

SHAKE VOL. 1

instruments for Native Instruments Kontakt 3, they've sampled strikes and shakes from 14 found and handmade instruments. Those are augmented by sustained rolls, grooves, and glissando effects, as well as a bonus collection of ambient drones (see Web Clip 2). The instruments were recorded dry, and the strikes and shakes include velocity layers and round-robin variations. The material is especially suited for orchestral, Latin, electronic, hip-hop, and rock music. Check out the demos and library details at the Tonehammer website.

AbbeyRoad | 60s Drums

Native Instruments
Abbey Road 60s Drums

The first fruit of Native Instruments' (native-instruments.com) collaboration with Abbey Road Studios is the vintage drum collection Abbey Road 60s Drums ($119, DVD and download). This sample library runs in Kontakt 4 and the free Kontakt Player, and it takes full advantage of Kontakt's GUI and scripting features to control articulation, sound shaping, mixing, and routing. The collection features Gretsch and Ludwig '60s-era drum kits recorded using Abbey Road's classic Neumann, AKG, and STC mics; the REDD.17 and EMI TG mixing desks; a tube-based Studer tape machine; and a host of authentic outboard gear. You'll also find set-ups for modern rock, urban, and

Get Smart

Sample Magic *The Secrets of House Music Production*

Sample Magic (samplemagic.com) augments its extensive collection of house-music sample libraries with the release of its first book, Marc Adamo's 144-page *The Secrets of House Music Production* (about $57). Featuring tips and tricks, walkthroughs, and insights from industry pros, the book covers all house styles: minimal, tribal, electro, progressive, soulful, jackin', and so on. The book examines drum sounds, making beats and bass lines, instruments and effects, mixing, mastering, and remixing. Step-by-step tutorials cover Apple Logic, Steinberg Cubase, and Ableton Live. You'll find 500MB of samples from Sample Magic on the accompanying CD to get you started.

Focal Press *Audio Engineering Explained*

Author and audio-amplifier designer Douglas Self has amassed a 656-page compendium of best design practices from the Focal Press and Newnes imprints of publisher Elsevier. *Audio Engineering Explained* ($49.95) from Focal Press (focalpress.com) includes chapters from books by audio-engineering pros such as John Eargle, Glen Ballou, Bill Whitlock, Philip Newell, Keith Holland, and himself. The material is presented in eight parts: fundamentals of audio and acoustics, microphone theory, preamps and mixers, power amps, loudspeakers, digital audio including MIDI and DSP, acoustics and sound reinforcement, and recording-studio design. Most topics are augmented by case histories and ready-to-go examples.

Abletoncourses.com Live Courses

Newly launched website Abletoncourses.com offers two levels of instruction for Ableton Live. The courses, which cost $245 each or $370 for the pair, are designed and presented by Ableton certified trainer Rob Jones. Each course includes online videos along with downloadable ancillary material, and both include a 750MB sample library from Loopmasters and the Novation Bass Station virtual-instrument plug-in. Level 1 covers Session and Arrangement views, recording and editing MIDI and audio, effects and instrument plug-ins, DJ'ing, mixing, and mastering. Level 2 elucidates advanced techniques, Live's premier plug-ins, and practical tips for composing, producing, and performing with Live. You can download a free module at abletoncourses.com.

ABLETONCOURSES.**COM**
the online ableton training resource

PSPAUDIOWARE PSP SQUAD

With the release of PSP sQuad (Mac/Win, $249), PSPaudioware (pspaudioware.com) delivers six VST, AU, and RTAS equalizer plug-ins inspired by the features and characteristics of classic analog hardware. Intended for treating individual tracks, ClassicQ and ClassicQex couple low- and high-shelf filters reminiscent of early Neve EQs with one and two midrange bell-shape filters, respectively. Fashioned on the EQs in consoles from the likes of Amek and MCI, ConsoleQ and McQ are designed for extensive tweaking and track separation. The

EQUAL ATTENTION

Q-factor of their midrange bell-shape filters automatically increases with gain or attenuation. RetroQ and preQursor offer PSP's filter designs with low-resonance peaks and narrow notching to provide enhanced band separation. The control panels emphasize ease of use, and extensive documentation helps you pick the right EQ for the job.

LINE 6 FBV FOOT CONTROLLERS   **POD STOMPIN'**

Line 6 (line6.com) has added two entries to its FBV line of foot controllers: the FBV Shortboard MkII ($199.99 MSRP) and FBV Express MkII ($99.99). The units are designed to control Line 6 amps, POD processors, or any MIDI software or device you can access via USB. Both devices feature a full-sized volume/wah pedal; an LCD (Shortboard) or LED (Express); and USB connectivity for updates, editing, and customization with the included FBV Control software (Mac/Win). The Express houses four footswitches, whereas the larger Shortboard has 13. Both units are powered through USB or an included RJ-45 cable. When used with Spider IV or Valve MkII amps, they offer channel switching and tap tempo, and control the Quick Loop and Pitch Glide effects. em

Home base: Oakland, Calif.
DAW of choice: Digidesign Pro Tools HD/LE
Key outboard device: Roland TR-505
Website: roguewavemusic.com

>>> Zach Rogue (left) and Pat Spurgeon

Indie Innovators

Rogue Wave's latest blends lo-fi production techniques with pop sensibilities

Oakland, Calif.–based Rogue Wave was formed in the early 2000s by Zach Rogue, armed with an acoustic guitar and influenced "mainly by a combination of late-'60s British Invasion music and mid- to late-'90s underground indie rock." Over time, the band's sound evolved as Rogue and his bandmates began experimenting with low-tech devices, toy sounds, cassette-tape manipulation, and 4-track recordings.

By Diane Gershuny

Their songs became more elaborate and ambient (finding exposure on TV shows from *Friday Night Lights* to *Weeds*). Their fourth release, *Permalight* (Brushfire Records, 2010), is in some ways a return to the early sound—short songs and lean arrangements—yet it's the most modern-sounding record they've made.

"When I started writing the record," Rogue says, "I made a conscious effort to make the music more visceral. Basically, I was hoping to make a dance record. There is still a lot of emotion in the music, but this time around, we wanted the physical component to be there, too."

Writing initially on an acoustic guitar, Rogue sketched single-track demos into Apple GarageBand, experimenting with melodic and lyrical ideas. He and drummer/keyboardist Pat Spurgeon tracked demos in Pro Tools and moved to a San Francisco studio for pre-production. They chose Dennis Herring (Counting Crows, Elvis Costello) to produce the album, and traveled to his studio, Sweet Tea Recording in Oxford, Miss., for the sessions.

Each song began with a click-track on a Roland TR-505 drum machine, with sounds subsequently added from the 505 to establish feel and tempo. Building upon that with drums, rhythm guitar, and scratch vocal tracks, the songs evolved.

The title track, "Permalight" (see **Web Clip 1**), was the only song recorded live rather than in layers. For that song, Rogue's Silvertone acoustic was miked with a Shure SM57, and bassist Cameron Jasper's Rickenbacker was taken direct and miked through a Marshall Micro Amp with a Sennheiser MD 421. The drum track was played on the TR-505 through a Lil Smokey cigarette amp miked with an SM57. Everything was run through JFL Audio preamps and into the studio's Neve 8038 console. They beefed up the drums using various drum machines, including the TR-505, a Roland TR-606 Drumatix, and a Mattel Synsonic.

Throughout the process, the band employed a host of offbeat techniques, Spurgeon recalls. "There is a lead guitar part that enters 'Permalight' that has a cool backward kind of sound, created with a Fender American Deluxe Stratocaster

run through an Electro-Harmonix Holy Grail Plus reverb pedal with an Ernie Ball volume pedal," he says. "Since the reverb pedal was first in the signal chain, the sound would be nice and lush, and then cut off clean when the volume pedal was brought down. The track was recorded using a Fender Champ amplifier miked with an Electro-Voice RE20. The signal was run through a Universal Audio 6176 Vintage Channel Strip. We also used an Akai MPC2000XL sampler to incorporate a 'Speak and Spell' toy in the pre-choruses and made use of a Korg microKorg synthesizer vocoder in the outro of the song to give the song a Led Zeppelin 'Immigrant Song'-meets-*Deliverance*-banjo sound."

Another of Herring's techniques was to record an instrument with a microcassette recorder using its internal microphone and playing it back into the track with a little wiggle here and there, as heard on the almost chime-like chorus of "Stars and Stripes" (see **Web Clip 2**).

The album was mixed at Sweet Tea by Kyle "Slick" Johnson. It was mastered at Masterdisk Studios in New York City by Howie Weinberg. **EM**

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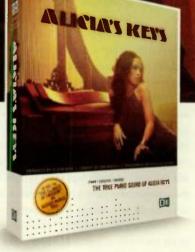
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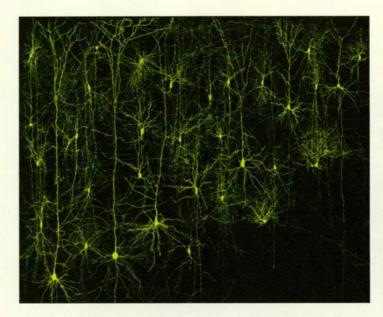
>>> **FIG. 1: Neurons** in the brain can be modeled as cortical columns, which comprise minicolumns of about 100 neurons each. The team at UCI uses this model as the basis for its research into higher brain function.

Your Brain on Music

Modeling musical cognition | By Scott Wilkinson

No one knows exactly how the brain works, but many scientists are trying to figure it out. Among them is a group of researchers at the Center for the Neurobiology of Learning and Memory at the University of California, Irvine. The team is studying higher brain function (HBF), which includes creative and analytical activities such as music, mathematics, and chess.

HBF occurs in the cortex, which forms the outer surface of the brain. Most of the brain's nerve cells, called neurons, are located in this surface layer. The neurons receive, process, and send tiny electro-chemical signals. Each neuron has a single output and thousands of inputs from other neurons. When the combination of input voltages exceeds a certain threshold, the neuron fires a pulse to another neuron.

Amazingly, there are 10 to 100 billion neurons in the brain with at least 100 trillion connections between them. Many researchers believe these connections are used for data storage, which corresponds to more than 10TB in a 3-pound package you can hold in your hand. However, the science of neurology is far from being able to unravel the workings of the brain at the level of individual cells, so many researchers use models of brain activity in which groups of neurons work together as a unit.

In one such model, the cortex is organized into basic processing networks called *cortical columns* (see **Fig. 1**), each of which is further divided into *minicolumns* of approximately 100 neurons. Each minicolumn is strongly connected to neighboring minicolumns, and some are also connected to distant minicolumns.

The electrical state of each minicolumn varies over time according to the activity of each neuron. Within a column, this behavior is called the firing pattern, which depends on the strength of the connections between minicolumns and the timing of the patterns. During HBFs, such as composing music, these firing patterns are relatively stable and somewhat periodic, evolving over seconds or minutes.

The UCI team has designed a computer simulation based on the cortical-column model. They have limited each minicolumn to three electrical states, corresponding to high, average, and low levels of neuron activity. Because of this, the minicolumns are called *trions*. Six trions constitute a cortical column in the UCI model, which gives rise to more than half-a-million firing patterns.

Some of these patterns are more probable than others in the model and in the brain itself. The most probable patterns correspond to the inherent firing patterns with which we are all born. As we are exposed to stimuli

such as different types of music, these inherent patterns are enhanced, contributing to our abilities and preferences in all areas of creative and analytic thought.

The UCI team has written software that steps through firing patterns in one or more cortical columns every 100ms, calculating each pattern from certain probabilities and the strength of the connections between trions. The software then maps these firing patterns to MIDI note numbers and channels to play notes with different instrumental timbres.

This research has yielded many interesting results. For example, the music generated by the model actually exhibits recognizable cadences and modulations. Different mappings of the same firing patterns produce different flavors of music, including waltzes, minuets, certain types of folk music, and different periods of Western art music. It is also possible to teach the model a musical theme and let it develop variations on that theme.

Although it is the most subjective of all HBFs, music was selected to study because it is highly structured and universally appreciated, even at birth. It is also very time-dependent, which is one hallmark of the trion model. The team hopes to discover common traits between music and other HBFs, leading to a deeper understanding of the brain and how it works in all its myriad tasks. **EM**

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Kontakt

Effects Processing

Unleash the robust DSP power of Native Instruments' sampler

By Len Sasso

Native Instruments Kontakt 4 presents a powerful array of effects processors, along with a flexible signal path in which to employ them. Whether you primarily use the factory and third-party content or fashion sampler instruments from the ground up, the addition of a few well-placed effects can give old sounds new legs.

Because Kontakt is a complex sampling environment and understanding its structure is key to using its effects, I'll start with a quick overview of how and where to plug in. I'll follow that with examples to serve as starting points for your own instrument construction and modification. Once you've added effects, you'll often want to add elements to the instrument's control panel so you don't have to flip to Edit mode every time you need to make an adjustment. That's a job for Kontakt's Script Processor, about which you'll find a "Master Class" in the February 2008 *EM*, available at emusician.com.

Organization

Like most samplers, Kontakt wraps each imported audio file in a zone containing infor-

mation about playing the file: MIDI key and velocity-triggering range; root key; and base volume, pan, and tuning. Each zone houses a single audio file, but the same audio file can occupy several zones, which, as I'll discuss, is useful in applying effects. Kontakt's Mapping Editor displays zones on a note-and-velocity grid where you can create, move, and resize them. In the standalone version, you can detach and resize the Mapping Editor for more convenient access. Double-clicking a zone opens the Waveform Editor (also detachable), where you can exert minute control over sample playback. Zone information is saved with the instrument, but zones cannot be exported separately.

Kontakt organizes zones into groups. By default, all zones occupy the same group, but managing multiple groups is one key to creative effects processing. Groups also carry a lot of other information, such as playback mode (internal, disk streaming, beat slicing, and pitch- and time-manipulation modes), MIDI input and audio output routing, and amp and pan settings with their modulation routings. Most importantly for effects processing, each

group has its own insert-effects chain and send-effects routing. You can save groups to disk for reuse in other instruments.

One of the handiest features of the Group Editor is the red Edit All Groups (EAG) button at the top left. When you have more than one group, turning EAG on will cause all edits—including inserting group effects and changing parameter values—to apply to all groups. For example, when you want to apply the same AHDSR envelope settings to all outputs, turn on EAG and then add the envelope and make its settings. You can then tweak individual group settings with EAG off as needed. You can also manually select multiple groups for editing using the checkboxes adjacent to the group names in the Group Editor.

Signal Flow

Signal flow in Kontakt begins with the Source module, which receives the output from all zones. When the instrument is in Edit mode (toggled by the wrench-icon button at the top left), the Source module appears below any open editors, or at the top if no editors are open. Group insert effects come immediately

FIG 1: The Group Editor provides eight insert slots (labeled Group InsertFX) for filters, distortion, effects, and utilities. Send effects are also routed from these slots.

after the Source module and are followed by the group's amplifier.

A group's insert-effects editor has eight slots, and you can toggle the last two slots to be post-amp. When you select a slot with an effect in it, the Edit button to the left of the slots reveals the effect's controls. Only the selected group's slots, effects, and controls are shown. (You can use MIDI or the mouse to select a different group.) Most group insert effects have controls to which you can apply Kontakt's internal modulators (envelopes, LFOs, and special modulators) and a variety of MIDI messages. Instrument insert and send effects support only MIDI CC control, which you set up with Kontakt's MIDI-learn implementation.

Another thing that distinguishes group inserts is that they are polyphonic: If you play a chord, each of its notes will be processed separately. If, for example, you have velocity or an envelope set up to affect filter cutoff, each note will be filtered differently. (That is why instrument and send effects don't support internal and note-based modulation.) Of course, polyphonic processing comes at the expense of extra CPU load, so instrument inserts are preferable unless you need special group-insert features.

Instrument insert effects come last in the signal path (after all group processing), and like group inserts, you get eight slots for these. Because the signal flows through insert effects in series from left to right, how you arrange the effects matters.

Going Postal

Eight send-effects slots appear below the instrument inserts, but their relation to the signal path is determined by how their inputs and outputs are assigned. Send effects are fed by insert-effects modules called Send Levels. You can put a Send Levels in any insert slot, and you're allowed more than one. Kontakt's default instrument has a Send Levels in the last instrument insert slot. That can come as a surprise when you're trying to manage effects from group insert slots. If it becomes a nuisance, save a new default from Kontakt's Files menu.

Each Send Levels has a separate level control for each installed send effect, along with a global return-level control and a return-destination selector. That lets you feed different

World Radio History

send effects from different points in the signal path **and** route their returns back to the instrument **output** or to one of the four aux outputs in Kontakt's mixer. You might, for example, use a Send Levels in the first slot to route the signal to a delay send effect and then, further down the chain, use another Send Levels to route the insert-effects-processed signal to a phaser send effect. You could then route the delay and phaser to Kontakt aux outputs for additional effects processing in your DAW. (To route an unprocessed group signal to an aux output, use a Gainer effect in a send slot.)

Kontakt offers one more option for effects processing: installing insert and send effects in the output mixer. Those apply to all instruments that use the same output. The output mixer sports four aux channels, which can hold send effects. Sliders on an Aux Sends strip associated with each instrument controls routing to those effects. Bear in mind that effects installed in the mixer are saved with the Multi, but not with individual instruments.

Effective Options

Kontakt offers 40 effects plug-ins, but not all are available in all slot types (see the **Online Bonus Material** "Filters and Effects By Slot Type"). The limitations are what you would expect: Send slots are limited to typical send effects such as flanging, reverb, delay, and the like; surround panning, which effects routing to the mixer, is not available in the mixer; and so on.

One of my favorite quick tricks is to layer a sound with itself, applying different group effects and using Group Start Options to keyswitch between groups (see **Fig. 1** and **Web Clip 1**). Kontakt's selection of synth patches is a good starting point; find an instru-

ment with a single group that spans a large key range and all velocities. (I used Naïve Synth Lead from the Kontakt 4 Factory library for this example.) Everything cited in the example is accessed from the Group Editor.

From the Edit menu, export the group (along with its samples). That creates a file with the extension NKG that you can import into any instrument. Delete the original instrument, create a new instrument, and import the group. Delete any group inserts (click on the X in their upper-right corners), then insert a 4-stage ladder filter in the first slot by clicking the slot's Add FX button (labeled +) or dragging it from the browser's Modules > Filters > Synth Flt tab.

You'll see the filter's Cutoff and Reso controls along with a filter graphic in the edit area below the row of group-insert slots. Right-click on the Cutoff knob and use the Context menu to add an AHDSR envelope to modulate the filter cutoff. (Alternatively, you could use the Mod tab's Add Modulator menu or simply drag an envelope module from the browser to the knob.) In the Mod tab, click the Quick Jump button to the right of the modulator name to see and edit the envelope settings. Use those along with the Cutoff and Reso knobs to get a contour that you like.

Use the Edit menu to duplicate the group, and then rename the groups "Layer 1" and "Layer 2." Ensure EAG is off and turn the Group Editor's Group Solo button on. Select Layer 2 and add a 2-pole HP filter as the second group insert. Add an AHDSR envelope and Sine (LFO) modulators and tweak the settings so that a slow attack takes the bottom out of the sound and a delayed LFO modulates the HP filter cutoff.

To set up keyswitching, turn EAG on, select Layer 1, and in the Group Start Options tab, select On Key in the top Group Starts slot. Turn EAG off and specify a range of one key (C1 to C1, for example) to activate Layer 1. Select Layer 2 and choose a different key range to activate it. Note that with keyswitched groups, you need to play one of the keyswitches

after you first load the instrument; otherwise, no group plays.

For send effects, put a reverb in the first slot and a delay set to eighth notes in the second slot. Turn EAG on and insert a Send Levels in the seventh group-inserts slot (the first post-amp slot). Right-click on the Reverb send knob, select Learn MIDI CC#, and wiggle the mod wheel. Turn EAG off, and in Layer 2 add another Sends Levels in the third group-inserts slot. Set the delay send to around -10dB. Notice that the delay repeats the HP filter envelope contour even after key release, which it wouldn't from a post-amp send.

Morph or Less

The new Authentic Expression Technology (AET) and the associated AET Filter group-insert effect bring amazing sound-morphing potential to Kontakt 4. AET uses high-resolution FFT to analyze and then impose the spectral characteristics of one sound on another. Furthermore, you can morph the spectrum of the sound source across several analyzed sound spectra with real-time control and modulation. AET morphing comes in two flavors: Velocity, to smooth out velocity zone transitions, and Articulation, to impart the spectrum of one sound on another. Here's how to set up an Articulation morph of a brass ensemble using the spectra of a mixed choir singing vowels (see **Web Clip 2**).

The AET Filter requires separate groups holding the sound source and each of the sounds used for morphing. For this example, I've started with instruments from the Choir and Orchestral sections of the Kontakt 4 factory library and exported the sustain group for each sound. It's important that the exported groups have zones covering most of the key and velocity range desired. The individual groups cannot have overlapping zones, but you can resolve overlaps in the Mapping Editor either manually (which is easiest in List view) or using the Resolve Overlapping Key and Velocity Ranges buttons. Be careful to avoid holes in the source group or you'll have dead spots.

After you've exported your source (the brass ensemble in my example) and morph groups, create a new instrument and import all the groups. You'll need all the groups to be resident in the instrument to create the AET morph maps, but you'll want to play only the source group, so turn the group amp volume all the way down on the other groups.

FIG. 2: To use the spectral-morphing AET Filter, you first analyze zone groups to create morph layers (left) and then arrange layers into morph maps (right).

Next, create a Morph Layer for each group. Select the group in the Group Editor, turn Selected Groups Only on in the Mapping Editor, select all the groups' zones, and then choose Create AET Morph Layer from the Mapping Editor's Edit menu. Type in a name, leave the default settings, and click OK (see **Fig. 2**).

Select your source group (ensure EAG is off) and insert an AET Filter in the first group-insert slot. Click the AET Filter's Edit Morph Map button to create Morph Maps. Make sure the source group is always the first one in the Morph Map and that the Articulation Morph button is active. When you're finished, exit the Morph Map Editor and choose the desired Morph Map for the AET filter.

The AET Filter's Morph knob determines where you are in the morphing spectrum—far-left for pure source to far-right for the last morph layer. Its Amount knob is like a dry/wet control: It determines how much effect the AET filter has. Both knobs are good targets for MIDI controllers, and I also like to apply an LFO to the Morph knob with the MIDI mod wheel controlling the LFO amount.

By the Slice

Kontakt is adept at handling pre-sliced loops in REX and Acidized WAV format, and it also offers both grid-quantized and transient slicing of its own. In all cases, you can add, delete, and reposition slice boundaries at will. I find it easiest to do slicing and other sample-based tasks in the standalone version of Kontakt because you can tear off and resize the Wave Editor. (To avoid crashes when dragging and dropping, make sure either the Wave or Mapping Editor remains embedded in the instrument.)

Start by dragging the audio file to be sliced from Kontakt's file browser to an instrument's Mapping Editor. (To create a new instrument, drag to an empty area of the Multi rack.) Open the Mapping and Wave Editors and select the zone in the Mapping Editor to reveal its sample in the Wave Editor. For pre-sliced audio, you'll see slice markers in the waveform display. For other audio, turn on the Grid panel and choose Fix for grid-quantized samples or Auto for threshold slicing. Double-click the Grid panel button to edit REX-file slices. Create the slice configuration you like. (The Kontakt manual gives full details.)

Drag slices from the Wave Editor to the Mapping Editor to create zones for those slices,
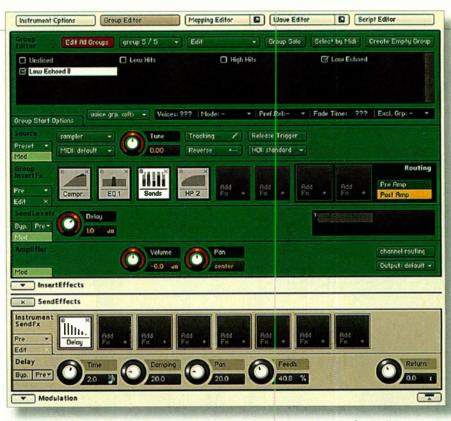
FIG. 3: The selected group (Low Echoed II) holds the fifth, ninth, and 13th zones of a sliced drum loop. After compression and EQ, the signal is routed to a delay send effect. The undelayed sound has the bottom rolled off with a highpass filter.

or use the Sync/Slice panel to create zones and a matching MIDI sequence from all the slices. In either case, Kontakt creates a new group for the slices. Creating subgroups with their own effects chains is a powerful tool for wringing new life from tired loops (see **Fig. 3** and **Web Clip 3**).
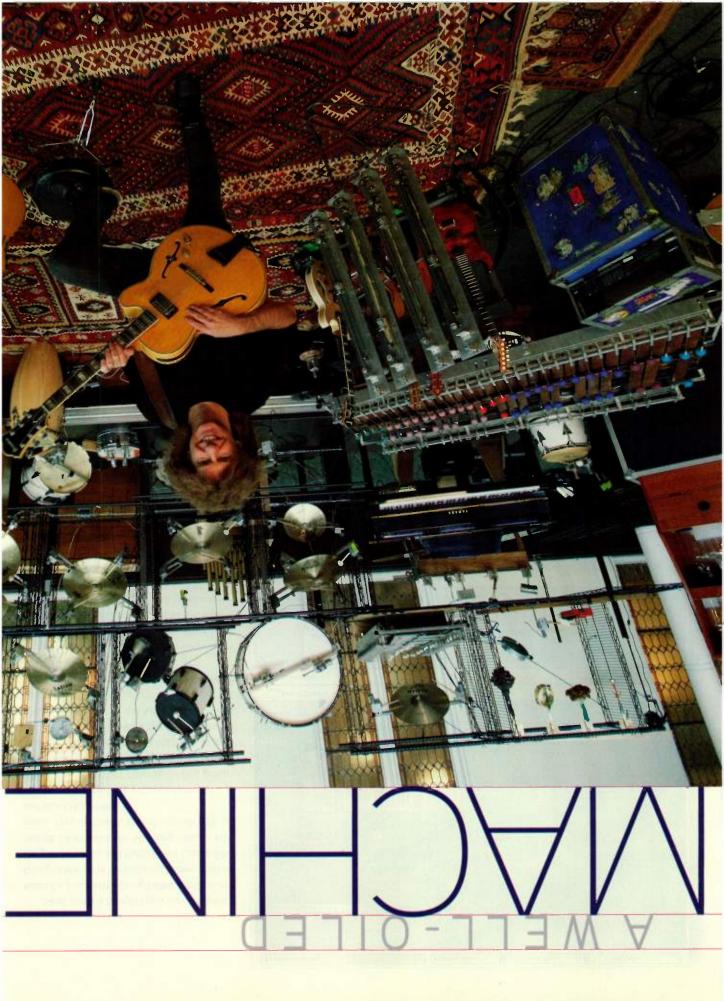
Kit Piece

Percussion and sound-effects loops are great fodder for drum kits. Simply drag individual slices to the Mapping Editor to build the kit. (Save often when working with pre-sliced loops.) Instead of using multiple velocity layers, try an AET Velocity Morph with one or more broad-spectrum sound effects. First, import the sound effects to new zones on free keys and set their velocity ranges to 64 through 127. Select all the drum-kit zones and resize them vertically to span a velocity range of 1 to 63. Now, move the sound-effects zones above the drum-kit zones, broadening or duplicating zones as desired to fill gaps in the key range of the drum kit. Try to match the sound effects with the spectral characteristics of the drum-kit zones below them.

Ensure all the drum-kit zones (drum hits and matching effects) are selected, and choose Auto Add AET Velocity Morph from

the Mapping Editor's Edit menu. Kontakt will create a Morph Layer and Velocity Morph Map from the zones and insert an AET Filter. For standard velocity morphing, you'd use this group as is, but you can also make the drum hits play across the entire velocity range while still being able to morph with the sound effects' spectra. Select the sound effects zones and set their volume to minimum (-36dB) and then change the maximum velocity of the drum-kit zones to 126 (not 127). Kontakt sets velocity to modulate the AET Filter's Morph knob, but you might want to change that to a mod wheel, pedal, or envelope (see **Web Clip 4**).
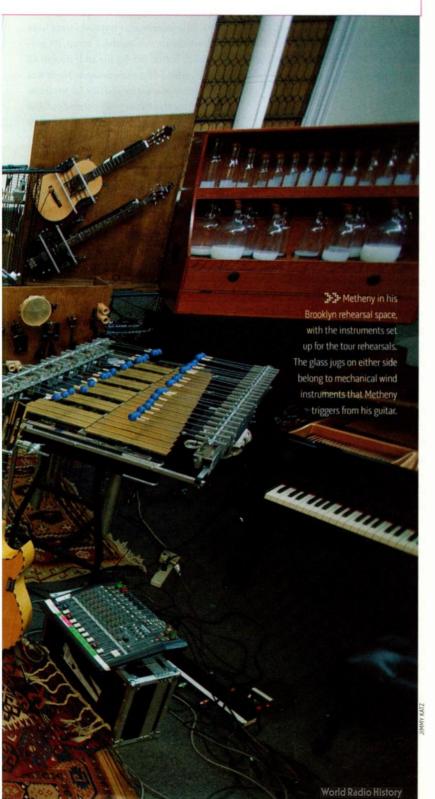
Kontakt's factory and third-party instruments are loaded with effects, and that's a good place to start searching for new sounds. Digging further down to remap and regroup zones takes a bit more effort but gives you the opportunity to create new instruments by bringing in different samples or exporting groups to new instruments. Before you shell out for yet another sound library, take a closer look at what's already on your hard drive. em

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A WELL-OILED MACHINE

Pat Metheny turns a group of robotic instruments into a jazz band on his latest album

By Mike Levine

Metheny in his Brooklyn rehearsal space, with the instruments set up for the tour rehearsals. The glass jugs on either side belong to mechanical wind instruments that Metheny triggers from his guitar.

In an old church in the Greenpoint section of Brooklyn, jazz-guitar legend Pat Metheny is rehearsing the music for the upcoming tour supporting his unusual new album, *Orchestrion*. In the church's main room, a large group of instruments are arranged in a setup similar to the one he'll have onstage for his upcoming tour. Some of the instruments are recognizable—a Yamaha Disklavier (a MIDI-controlled grand piano), vibes, a marimba, an electric bass guitar—and others aren't: two groups of glass jugs, each on a shelf-like contraption; and a metallic stringed instrument with four one-string necks (aka the Guitar Bot). A drum kit is arrayed in a very unconventional manner, with the individual drums and cymbals hanging on the large metal frame that serves as the backdrop of the stage. Most of the instruments have retrofitted electronic and mechanical contrivances attached to them.
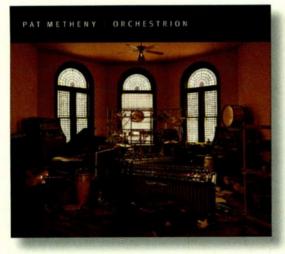
What's most intriguing to me, as I observe the rehearsal setup, is knowing that Metheny will be the only live musician onstage during his tour, just as he was the only one on the record. His "bandmates" are all robotic instruments that he controls remotely, either from his MIDI-equipped guitar (which has a 13-pin pickup connected to an Axon MIDI converter box) or from programmed sequences coming from an Apple Mac running Avid Sibelius, MOTU Digital Performer, or Ableton Live.

I find out later that many of the instruments (including the Guitar Bot and the percussion) were designed by Eric Singer, a pioneer in the field of robotic music. Singer is the mastermind of an outfit called LEMUR (League of Electronic Musical Urban Robots). The group, which until recently was based in New York City (Singer has now moved the operation to Pittsburgh), builds musical robots for performances and installations of many types. (For more on Singer's role in the design of Metheny's instruments, see the sidebar "The Man Behind the Robots" on p. 30.) Some of the other instruments were designed by a Californian named Ken Caulkins, whose company is called Ragtime, and by Peterson (the strobe tuner-maker).

Back in the Day

Before demonstrating how he interacts with his new band, Metheny gives the assembled reporters and photographers some background into why he chose such an unconventional approach for *Orchestrion* (see Fig. 1). He starts by showing us a book of photos of old, ornate instruments called orchestrions, which inspired the album title. "An orchestrion is basically a player piano that expands outward and upward into other instruments," Metheny says. "It could be a couple of drums, a snare drum, a cymbal."

⌖ FIG. 1: Before deciding to go forward with *Orchestrion*, Metheny had been interested for many years in the concept of making a record with remotely controlled instruments.

He says that orchestrions came into being toward the end of the player-piano era, which lasted roughly from the 1890s until the 1920s. "There was this interesting period where player pianos emerged, which was before recorded sound. It was really the first time musicians or composers were able to represent themselves without actually being in the room."

Back then, he explains, "If you wanted some entertainment, there were no discos, there were no record players, there was nothing like that—there were player pianos. Lots of restaurants, lots of bars, lots of entertainment, roller rinks, and whatever else had player pianos. Tens of thousands of player pianos were sold. Tens of thousands of orchestrions were sold. It was a huge business."

Metheny explains that his interest in such instruments was sparked by the player piano that his grandfather had in his basement. "I spent every hour that I visited him messing around with this player piano," he says. "I continued to have a fascination with that world of instruments, really, ever since."

About two years ago, Metheny observed that the technology for modern robotic instruments, which typically employ solenoids to control the physical actions of playing (as opposed to the pneumatic technology used in vintage player pianos and orchestrions), had progressed to a point at which they could control instruments with the dynamics that were lacking in earlier self-playing instruments. As a result, he decided to pull the trigger on what is probably his most ambitious project to date, an album on which he's accompanied only by these modern orchestrions. "I had to go about explaining to people what I wanted to do. And every single person, including my wife, thought I had completely lost my mind. I've gotten a lot of weird looks over the years for various things, but this one takes the cake," he says.

On the Record

The album itself features Metheny's clean, jazzy, electric-guitar lines and solos, backed up by the sounds from the robot instruments, which provide not only the rhythm section, but also a lot of melodic support. Instruments such as the slidey-sounding Guitar Bot (see Fig. 2) and the glass-jar wind instruments add some unusual texture to the sound.

Metheny either played or programmed all of the instruments on the record. From the almost 16-minute title cut to the five other tunes on the record (the shortest one is 7:45 in duration), you get the feeling you're listening to jazz compositions that, save for Metheny's guitar parts, were mainly orchestrated rather than improvised. Metheny has done a masterful job of playing and programming the instruments so that their performances don't sound robotic.

"People still don't understand what it is," Metheny says about the album, "because it really is something that is hard to express. So let me just say this: It's not synths, it's not samples; it's a thing—a big, living, breathing thing—a combination, actually, of a whole bunch of things. It's a very different experience—as a composer, as an improviser, as a musician—from anything else. It doesn't connect to anything else. It's not better, it's not worse; it's different. And that's kind of the main headline. It's no replacement for anything else; it's no substitute for anything else. But it really opens up a whole different set of possibilities."

Mixing in the Air

In our subsequent interview, I ask Metheny why he went through all the trouble and expense to use these retrofitted, robotic instruments when he could achieve an even wider range of sounds using synths and samplers.

"It's not like I haven't been right in the middle of everything else that's been going on," he says, "from 8-bit to 16-bit to 24-bit to, you know, the most high-def—every little step along the way, I've been very interested in what each of those little incremental improvements have offered, which are substantial. I mean, it's gotten so much better. But [in the end] it's still all getting crammed into two speakers. There was a minute there when it looked like 5.1 would be something, but that hasn't really panned out. But to me, it's still not anything like what happens when you have some stuff [real instruments] playing in a room."

That sonic interaction is what made the robot-instrument approach more appealing to Metheny than synths. "It's wildly different," he says. "It's like the difference between pureed tomatoes and picking a tomato right off the vine in a vineyard somewhere. The stuff is alive; I mean, every time it hits, it's different. It's not ever the same. If you have a sample, it's exactly the same every time. You can, of course, fudge

⌖ FIG. 2: The Guitar Bot (designed by Eric Singer) produces some of the more unusual sounds on *Orchestrion*.
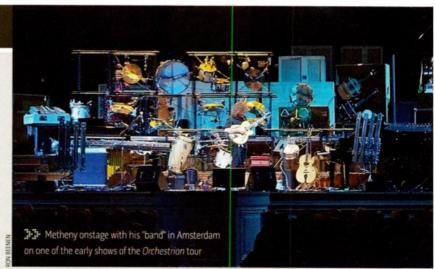
Take This Show on the Road

When Pat Metheny embarked on the *Orchestrion* project, he realized that touring with the robotic instruments would be very different from recording with them. "I knew I would have two big jobs," he says. "Make a record and do a tour. I knew the making-the-record part would be the harder part because there's no visual feedback. It's got to do what records do; it's got to tell a story and do all the things that music must do as an auditory function of our existence. Live, there's a lot more information about what's going on."

Onstage, there's also more chance for improvisation, and much of it will involve Ableton Live. "The live jamming that he does is going to be very different every night because of the way that Ableton morphs the stuff," says David Oakes, one of Metheny's two longtime techs; the other is Carolyn Chrzan.

Live, Metheny will record scenes (groups of triggerable individual parts) in Live. It will be similar to using a looper. "He'll start building the tracks up, and all of a sudden, the drums will be rocking, everything will be going," Oakes says.

Touring with the robotic instruments also opens up some unique logistical issues for the crew. They can't carry spares of the instruments (they're too big and expensive for that), but they will be stocking a lot of spare parts. "On most of these things, if the solenoid breaks, you can unscrew it, put in a new one, wire it up," Oakes says.

▶▶ Metheny onstage with his "band" in Amsterdam on one of the early shows of the *Orchestrion* tour

He points out that when a human plays a percussion instrument, he or she will hit it in a variety of different spots, whereas a robot instrument will hit in precisely the same spot each time. "You're going to wear out drum heads a lot faster," Oakes says.

"The instruments take a different kind of abuse," Metheny adds.

With so many instruments together onstage, the need to minimize mic bleed will be great. Oakes says he was considering using ribbon mics to help cut down on the leakage. "What I'm doing now is experimenting with using all ribbons, or as many ribbons as I can, mainly because, with the figure-8 [polar pattern of the ribbon mics], the side rejection is pretty total."

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that a little bit with a variety of tricks that we all know. But the fundamental thing is exactly the same. And every time you play a note and it blends with other notes, you're not really getting the sonority of those notes mixing with each other in the air. You're getting discrete samples mixing with each other in a speaker."

Playing With Vapor

So how exactly did Metheny go about recording *Orchestrion*? Did he have all the music written in advance, and then just go about orchestrating it with the robot instruments once he got them? "That would have been nice if it had gone down like that," he replies wistfully. "I essentially pulled the trigger—'Okay, I'm going to do this'—in the middle of 2008. I've been thinking about it for about 15 years, and had sort of followed who was building what and what experiments were being done. And so I made the call and told the guys that I'm going to do this thing, and, of course, everybody's response was, 'You're going to do what?'"

Once he decided to go ahead with the album, he had to commit to touring to support it. "They have to book these tours about one-and-a-half years in advance, so the call was made and we were going to do that," he says. "At that point, I made commitments to all these different instrument-makers."

So, by necessity, Metheny had to agree to do the album and the tour before he'd even had a chance to try out the instruments. He was working with a tight timeframe and had hoped to get the instruments by January 2009, but that wasn't to be.

"This was sort of my vaporware project for a while. I didn't get one instrument until March. Everybody was late with everything, partially because of me," he says. "I really wanted it to sound great. For example, the drums and the cymbals are Jack DeJohnette's drums and cymbals. Before, [the instrument-makers] had crummy little [drums], and they were like, 'Isn't it cool that it works?' I was like, 'It's cool that it works, but it sounds terrible.' There was

a lot of that. For instance, they had done some mallet instruments with really bad mallets and it sounded terrible. I said, 'That's not what those instruments are supposed to sound like.'"

So Metheny got some help from his friends. "Gary Burton gave me 37 Gary Burton vibe mallets [one mallet on each note], so it sounds fantastic. Gary also picked out all the mallets for the marimbas, so that register by register, it sounds beautifully balanced," he says.

"In the meantime, yes, I was writing a whole bunch of music that I was then planning on orchestrating, [but], essentially, nothing that I had planned worked. And all that music, when the instruments were there seemed [not] to make an enormous amount of sense. Then I really had to start over. Really, I was kind of writing for the stuff once it came in. Everything sort of emerged. In a way, it was a lot better. It was like I really knew what I was getting, and what I was working on was what it wound up being."

The instruments were designed to be controlled with a high degree of precision, allowing

for dynamic expression. (One of the problems with player pianos and orchestrions in their heyday was their lack of dynamics.) For instance, the ride cymbals can be hit in four different ways, the crashes in two. And the Singer-built instruments have an impressive dynamic range.

"The way he cracked dynamics is a pretty fascinating story," Metheny says, referring to Singer. "Essentially, he's got a box that he's built that converts the full range of 0-127 to voltage so that he can then modulate the voltage based on the velocity that it's receiving. But if you think about it, it's not just a matter of hitting harder, because if you hit something harder, it's not necessarily going to be louder. It's a question of speed. And how quickly it can get off the note, too. It's really quite a difficult engineering problem, and he found a very elegant solution. It's also very robust.

"Essentially, I had from the middle of March until the middle of September to learn all this stuff and write all this music. I wrote a lot of stuff that I just didn't get to, that I was not able to finish. Everything was in a room in my apartment. And believe me, my wife was never so glad to see me leaving [laughs]."

Tracking It Down

One of the issues Metheny ran into in the studio was the latency of the robot instruments. He discovered that there was from about 25ms to 50ms of delay in all the instruments. "I'm very groove-aware," Metheny says. "And for me, the feel of things is important. And honestly, before I could do anything serious, I had to spend weeks measuring all of those latencies and really understanding what they were from instrument to instrument, and setting zillions of software offsets [for use] under these conditions or those conditions, so that 'one' was really 'one.' I'm kind of way pickier about that stuff anyway, under any circumstances. I'm one of those guys who's really concerned about latency."

The solution to the latency issue was not an easy one. Metheny had to engage in his own manual version of latency compensation while he was playing. "You can get something in there by rushing, which I do anyway, so it fits in nicely with my thing [laughs]. Then on output, meaning when it comes back, if it's something that I'm laying there, all of the offset stuff that I've done compensates for it. That in and of itself was a pretty massive undertaking. And

it's another one of those things where people didn't exactly know what I was talking about in terms of how specific I needed that to be. It turns out that Digital Performer is really the one for that."

Software to Hardware

During the writing and recording process, Metheny used a number of different software applications. He is accustomed to working in a notation-based environment, so Sibelius was always running.

"When it comes time for just writing notes and organizing music in the sort of conventional way, Sibelius is the greatest," he says. "I always had Sibelius going in the background with ReWire. I always keep a score going with everything I'm doing because I need to see that. Sometimes I need to play this part or I need to transpose this part, or I need to do this or I need to do that. If I don't have actual music in front of me, I'm just lost."

However, when writing, Metheny often relied on the improvisational capabilities in Ableton Live. "For developing materials and being able to organize things in interesting and fun creative ways very quickly, you can't beat it. It's just a fantastic sort of playground for ideas," he says.

For getting the parts ready for recording, Metheny chose Digital Performer. "It's the deepest, most musical, and—for me, this is the most important thing—most time-locked, where you get exactly what you think you're getting, down to the sample program there is," he says. "And for me, it's the program that I have 14 years' experience with now, so I'm very fluent with it.

"Once I had everything organized, put together, and recorded in my world, going into the studio where we actually recorded the audio part of it and mixed it and everything else, Pro Tools is still the king for that. So, really, it was four platforms."

Metheny offers this description of how he worked up the material for the title cut: "I started in Sibelius, wrote a whole bunch of material," he says. "I had a huge amount of stuff. And then I started splitting it off, kind of into certain keys, certain areas, and then loading those into Ableton—kind of setting up some canons and stuff within Ableton. And then wound up with an Ableton file that had, I think in the end, 400-and-some scenes. Then I had all this material and kind of developed the form in Ableton and

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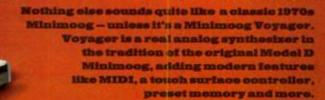
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tweaked it till I got it—not on the scene-based page, but on what they call the Arrange page—and basically nailed the form there.

"Then I dumped it from Ableton over to Digital Performer. I used that to do all the fine-tuning stuff, all the guitar-audio recording, all the musician-y kind of things."

When tracking the final parts into Pro Tools, Metheny and his crew had two somewhat contradictory aims: to record the tracks as cleanly as possible and to capture as much of the sonic interaction of the instruments as was feasible. "We had everything set up in a room," he says.

"They were divided up a little more into sections where percussion colors were all in one general area, and drums and cymbals were all in one general area, and the mallets were all in another area. But we did try to record as many things together as made practical sense, and as made sonic sense. There's a certain kind of bleed that's good, and there's a certain kind of bleed that's not good."

Depending on the song, they took different approaches. "There's everything from all the instruments playing together at the same time to each string playing individually, sometimes bar by bar," Metheny says.

Logistics Gone Wild

As mentioned, the album's writing and recording processes presented a number of challenges to Metheny, which he overcame with his musical skills, technical knowledge, and creative use of DAW and notation software. But the recording was only one part of the equation. He also had to learn to tour with his band of instruments, which presented a whole other set of logistical issues (see the sidebar "Take This Show on the Road" on p. 27). As I write this, Metheny has just embarked on the *Orchestrion* tour, which will run from early February through June and travel to cities in Europe, the United States, and Asia. That tour, which will be his most expensive ever, will be a real test of his talents, his crew, and the road-worthiness of his robotic instruments. Reports from his early shows are quite positive.

As for the music of *Orchestrion*, Metheny's efforts demonstrate that mechanically controlled instruments are capable of playing music that's complex harmonically, melodically, and, most important, dynamically. While some might consider all the time, effort, and money spent on this project to be frivolous and might argue that synths and samplers could achieve much the same result, Metheny clearly does not. He has impressively overcome myriad technical and aesthetic obstacles in the course of writing and recording this album. He views this project as just another challenge out of many he's had in his career.

"From my perspective," he says, "it's like I've been building this house for all these years, of all these musical opportunities—playing trio, playing with my band, playing with this, that, or the other thing. This is nothing more and also nothing less than a whole new addition onto the house. It's like a new bunch of things that are possible, and I honestly didn't know what the result would be. I wasn't sure whether it was going to be cool or what was going to happen, but it took me to some places that were really very different places. And the result speaks for itself." **em**

Mike Levine is EM's editor and senior media producer.

The Man Behind the Robots

Eric Singer, who invented the Guitar Bot and many of the other robot instruments Pat Metheny used on *Orchestrion*, spoke to *EM* about his creations.

Which of the instruments in Metheny's project did you invent?

Almost everything. Toward the middle and later parts of the project, he brought on some instruments from some other inventors to kind of fill out the orchestra. But we built all of the traditional percussion, some of the nontraditional percussion, all three of the mallet instruments—which were marimba, vibes, and orchestra bells—and two Guitar Bots. [We built] four ride cymbals, four crash cymbals, a couple of congas, a couple of toms, snare drums, a couple of bass drums, hi-hats, and a whole bunch of different styles of rattles. All in all, about 40 to 45 individual instruments, but maybe 150 different mechanisms.

Your instruments take the MIDI signal coming from his MIDI guitar setup and translate that into mechanical actions?

Exactly. The way all of our instruments work, and how they've worked from the beginning, they all have microprocessor boards, kind of black boxes. MIDI goes in and a variety of control signals come out to control motors and solenoids. All of the hard work is taken care of inside those boxes; they're all custom. And so the instruments are all designed to work just the way you would expect them to as a composer or performer. So you could literally plug a MIDI keyboard in and run up a scale and play the notes on a mallet instrument or play different percussion instruments that way.

Your devices are reading the note on, note off, duration, and velocity.

Yeah, we translate all of that, even like a sustain message. On the vibes, if he holds down the sustain pedal, the volume pedal goes down. All the note ons, the note offs. For percussion, you ignore the note offs, you just use the initial hit with the velocity, and that velocity is converted into a timing signal from the solenoid. It's just basically how long you pulse that solenoid that controls how hard it hits.

For those not familiar, how does a solenoid work exactly?

A solenoid is really a very simple device. It's an electromagnet and a rod, and when you turn on the electromagnet, it pulls the rod down into the body of the solenoid. Now you take that and you need to attach it to a lever because it's a very small action. They only move perhaps half an inch to an inch. So that rod goes to a lever, and that lever gets pivoted, and that pivot is attached to a mallet or a stick or a brush, or anything.

EVAN CAIRO

>> Eric Singer is the inventor of many of the robot instruments Metheny uses on *Orchestrion*.

SEE VIDEOS OF METHENY AND HIS "BAND" AND LISTEN TO A PODCAST   EMUSICIAN.COM/BONUS_MATERIAL

World Radio History

It's Only a

A recording-musician's guide to understanding and controlling phase issues

By Brian Heller

The concept of phase comes into play again and again in audio technology, from stereo mixing to microphone placement to digital sync to everyday troubleshooting and more. When it works in your favor, it can be responsible for some cool results; when it works against you, it can take whole instruments out of the mix. Understanding the consequences of phase will help you grasp a great variety of issues that recording musicians encounter every day.

What Is It?

A wave's phase describes where it is in its cycle at any given moment. While phase is related to both frequency and time, it does not measure wavelength, speed, or absolute distance. What it does measure is where a wave is relative to the beginning of its cycle. From that, its phase can tell us something about what kind of energy the wave has at that moment, and thus how it might interact with other waves it comes in contact with. Although we're focused on sound here, phase is a basic characteristic in other media that travel in waves (wave mechanics), so by understanding phase in sound, you can also start to understand something about how it works in light, water, and even quantum mechanics.

As you probably know, sound waves are the patterns of changes in pressure we hear as sound

moves through the air. The length of this pattern is called a *period*, and when a wave has made one trip through the pattern, it has moved through one complete *cycle* of the wave. When a wave pattern repeats in the same shape continuously, it is called a *periodic* wave. These periodic waves carry the sense of pitch in the sound, while waves whose shapes do not repeat regularly carry the sense of noise we hear. I will be referring primarily to periodic waves here as they are relevant to music and are easier to analyze.
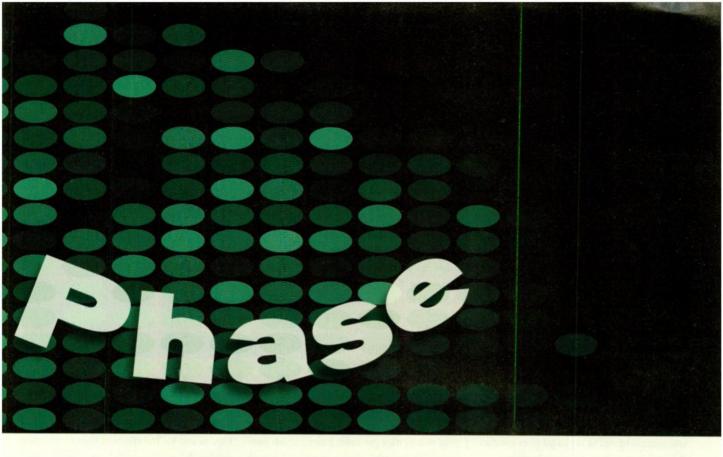
It can be useful to describe exactly where a sound is in a cycle at a given moment, and because of the repeating nature of a periodic wave, it is common to measure its period in degrees, like a circle might be (see Fig. 1). Although we've been describing acoustic sound waves, electricity is also represented with waves, and so phase also applies to analog audio—and, by extension, digital audio (after it goes through D/A conversion).

When Waves Collide

Waves interact with each other when they collide acoustically or are combined electrically. When either of those things happen, their individual energies get mixed together and the result is a new combination. In AC electricity, this is called *summing*—simply adding things together.

Because their energies are different at different points in their cycles, the phase of one wave relative to another is a critical factor.

Electrically, this summing usually happens when they are combined in the same bus of a mixer. So when two channels are both panned hard-left, they both go to the left mix bus, their full signals are summed, and thus the phase characteristics of those signals will interact. On the other hand, if they are hard-panned in opposite directions, one goes to the left mix bus and one to the right. If those buses go to left and right speakers, then the two signals are not summed and never meet (until they get back into the acoustic world), and so their electrical energy remains separate.

It is important to understand that sonically, phase comes into play only when waves are summed together in some amount. If each wave happens to reach the mic at exactly the same place in its cycle, they are fully in phase with each other. In that circumstance, when they are summed together in a mixer, the total result will be louder than either of the two by themselves. This is called *constructive interference* (see Fig. 2a), and reflects the common sense that if two summed microphones pick up the same sound at the same level, the result will be an increase in the overall volume.

Phase

The opposite effect, *destructive interference*, occurs when that same wave arrives at the two mics at different points of its cycle. When that offset reaches 180 degrees, they are then completely out of phase with each other. This effect can actually cause the summed sound of the two mics to be quieter than just one because waves that are 180 degrees out of phase with each other cancel each other out and produce no sound at all (see Fig. 2b).

How Problems Arise

Taken by itself and with simple sound waves, phase can seem pretty basic and much harder to hear than other aspects of sound, like amplitude and frequency. But in the real world, waves are not simple and they hit each other at any point in their cycles, not just 0 degrees and 180 degrees. Because sounds comprise more than a single simple wave, their harmonics interact, causing some frequencies to add and others to cancel. When you consider that there are thousands of them interacting at a time, this is where the real complexity and richness of sound occurs. When severe constructive and destructive inference is spread across all frequencies, the result can create an odd effect called *comb filtering*, or what people tend to call *phasiness* (see Web Clip 1).

The most common place phase problems start is related to microphone spacing. Because the distance between two mics changes the time a sound arrives at each of them, the differences in phase between all their harmonics interact, causing some amount of comb filtering. While it can be used as an effect, it's not usually a desired sound. So engineers often listen to a combination of different mics and try to find (and avoid) places where phase interactions have a negative effect on the recording.

One common situation where many people overlook checking phase is when miking guitar amps, where it is common to use more than one mic in front of the speaker. These mics may or may not be exactly the same distance from the speaker, so the combination of them should be checked for the negative effects of phase. This can also happen when recording two channels of an instrument—bass, for example—using a miked amp and a DI box. So depending on where the phase of the waves happens to fall when they hit the mic, some fairly significant changes in tone can result (see Web Clip 2).

One of the most critical places to be aware of phase is when the typical combination of close and overhead mics are used on a drum set. The sound from the snare drum hit is heard through its close mic first and the overhead mic slightly later because it's slightly futher away (see Fig. 3). The typical spacing of these mics is in the range that can often cause cancellations between them. Add that there are usually two drum overhead mics that are often spaced apart, and the chances for audible phase interaction between them all is even greater.
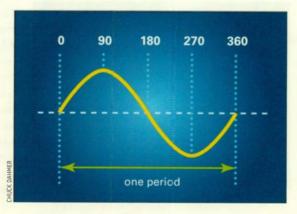

FIG. 1: The process of a periodic wave moving through time is measured in degrees because it's cyclical.

World Radio History

Staying Out of Trouble

There are a few suggestions to avoid the problem in the first place. One way not to invite extra "phase-ghosts in the machine" is simply to use the fewest number of mics you need to get the job done. The fewer mics on the floor, the fewer entry points for sound, and thus the fewer places they have to potentially conflict with each other.

Another commonly known method is by following the 3:1 rule. This principle says that for every unit of distance (whatever it may be) between a mic and its source, a nearby mic should be separated by at least three times that distance. Put another way, the distance between a mic and a source should be one-third of the distance between that mic and the next one closest to it. The idea is to avoid placing mics too close together, which greatly increases the chances for some destructive phase effects between them. In practice, the 3:1 rule is a good guideline to start with, but not be tied to, as you get more comfortable with how to listen for phase problems.

This is also one reason some engineers avoid using spaced stereo techniques in certain situations (particularly those that may involve broadcast) in favor of coincident stereo setups, such as X/Y, M/S, or Blumlein because, by definition, these coincident techniques position the left and right mics close enough together that there's no phase differences between the two. (For more on stereo miking, see "Double Your Pleasure" in the June 2000 issue of *EM*, available at emusician.com.) If you are using a spaced omni pair of mics for drum overheads, you will also want to check the phase between the two of them, and between them and at least the kick and snare.

How can you spot ill effects of phase problems between mics? They don't always occur, so it's something that you need to check for each time. To find them in a drum kit recording, you need to be able to hear just the snare and one overhead together in mono (which sums them, as discussed above). This can be done simply at any mixer, or with a combination of a mixer and a monitoring controller, which makes it easy to hear the results in mono without disturbing your mix. If you have a console, the PFL Solo button on each channel will also do just want you need. (However, it won't reflect actual fader levels in your mix as it's a pre-fader listen.) If you hear comb filtering (the phasey or wooshy sound mentioned earlier), then you likely need to make a small but important adjustment.

Often there isn't obvious comb filtering, but the mix of the two mics causes the drum to sound significantly thinner and weaker than it might normally. An odd frequency response, such as a hollow sound in the midrange, can also be indicative of destructive phase interference in the two mics. Many engineers check these things by habit as they're setting up their mix and listen for one of these problems to jump out.

Phase vs. Polarity

It might be useful at this point to quickly address a common misconception. Phase and polarity do not describe exactly the same thing. As we've looked at already, phase is specifically about timing. When you have a phase problem, you have a problem between the timing of two things, such as the distance between two mics or latency between two DAW tracks. The term *out of polarity* refers to two waves that may actually be in phase (i.e., they started at the same time), but whose energy is moving in opposite directions. Polarity does, indeed, relate to the shape of a wave and how it combines with others, but a problem with polarity is not related to a time offset
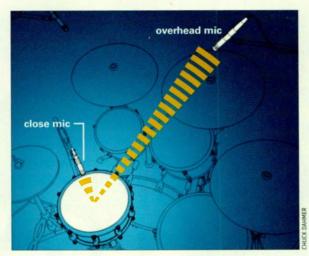
FIG. 3: Mics spaced apart can capture a wave in phase in one mic and out of phase in the other, leading to comb filtering and an odd frequency response.

between two similar waves the way phase is (see Figs. 4a and b). The effects of the two are often similar: When two sounds have opposite polarity, they look and behave like two sounds that are 180 degrees out of phase. But the solutions to the two problems are different, so it can be useful to differentiate them.

A common example is the way a snare drum works: When the drum is struck from the top, the head is pushed down. This creates low pressure on the top of the drum and high pressure on the bottom. So now we have the same (or similar) waves from the same source and at the same time, but radiating in opposite polarity with each other. This is a recipe for sonic weirdness because any frequencies that are the same in both sounds (which is many of them) will cancel each other out. This is why a polarity-reverse (often mistakenly called a *phase-reverse*) switch is used.

This function simply turns the wave upside-down (sometimes also called reverse or invert) without changing its timing so that all the points that were cancelled now add constructively, and vice-versa. It flips the sound from the bottom of the snare to become in phase with the sound at the top, and so they add constructively instead of destructively. The result is a snare sound that is usually much more full than if the two mics were kept out of polarity. This technique is often used with open-back guitar amps because similar principles are at work.

That said, using the polarity-reverse switch can still be useful in more drastic cases of phase
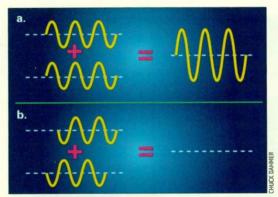
FIG. 2: Waves that are combined in phase are constructive, increasing total amplitude (a). Waves combined when they are out of phase are destructive, decreasing total amplitude (b).

bugs or as a coarse quick fix (see Web Clip 3). This can be done easily on many hardware mixers and mic preamps—look for a Ø symbol. All major DAW platforms have some way to invert polarity, as well, either file-based (which changes the audio file itself) or real time (as a plug-in). Sometimes there is a dedicated plug-in, but sometimes you need to look for it as a function of another plug-in. For example, Pro Tools LE doesn't provide a dedicated polarity-reverse switch as a real-time RTAS plug-in; there is one in the file-based AudioSuite menu. But it is a function included on many other plug-ins such as trims, delays, and EQs. In general, try engaging the polarity-flip on the mic with the least amount of leakage in it, which usually means the close mic. If the sound of that works, you should now check that close mic with other mics that share sounds in it.

For the snare, check it against the hi-hat mic because their polarity relationship is now opposite of what it was before. You might also need to change the mic under the snare back to its original polarity if you flipped it earlier. It is unlikely things will end up perfect with this method, but it can produce big changes in sound quickly, and so might very well be an improvement.

Making It Right

Fixing a phase problem with a time-based solution can happen in a number of ways at any point in the process, although the earlier the better. During recording, the most direct solution is often to move the correct mic just a bit closer or further away from the other one, listen, and

FIG. 4: The difference between two waves in phase but 180 degrees out of polarity (a) and two waves 180 degrees out of phase (b).

repeat. Because of the variables and complexities in the acoustic world, it can be difficult to predict which exact move will completely fix a phase problem. But as a general guideline, if the cancellation seems to be occurring in th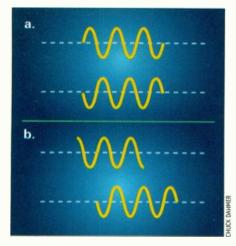e high-mids, subtle movements of less than one foot can do the job, and hopefully with a minimal change in the tonal quality of your setup. Lower frequencies usually mean larger changes are needed, so listen and check carefully to balance correcting the phase problem with keeping your desired sound.

As it fundamentally has to do with time, anything that lets you change the timing between the two tracks during or after recording can be used as a fix. We're talking about very small increments of time here, so you'll need tools that allow you control at the millisecond or even the sample level. This is commonly done in a DAW, which typically lets you see the waveform and get some visual help in lining things up. Be aware that this method is not foolproof either, and that your ears must prevail over your eyes.

Because the sound gets to the overheads later than it does to close mics, one approach is to slide the overheads slightly earlier in the timeline. This will put them more in phase with the snare and possibly other close mics to varying degrees. You'll need to zoom in far enough to be able to see individual waveform cycles. It can also be helpful to use a quantized editing mode in the DAW so you know exactly how much of a change you're making. For example, the Relative Grid mode in Pro Tools is ideal for this because the grid can be set to milliseconds or samples, and it lets you move the sound relative to where it actually started, as opposed to relative to the timeline (see Fig. 5).

A potentially faster (but less-visual) method can be to use a simple short delay on the close mic, tweaking the delay time slowly while listening. Again, the increments here are small, much smaller than most conventional delay plug-ins will allow. There are often other delay plug-ins included for this purpose (or ones for the purpose of compensating for latency, which do the

FIG. 5: Snare and stereo overhead tracks recorded out of phase. Pro Tools' Relative Grid mode and Nudge function in samples can be used to slide the snare track later in time, putting it in phase with the overheads.

same job). This is one use of the Digidesign Time Adjuster plug-in included with Pro Tools.

Take Advantage of Phase

Finally, it is worth noting that every track not being perfectly phase-aligned may not necessarily always be a bad thing. It is precisely these phase and timing differences that create spaciousness and can bring out the sense of the room in a recording. These minute timing differences can also lead to a bigger drum sound or thicker sounds from an amp due to that split-second delay between the mics.

One idea to use these phase anomalies to your advantage could involve positioning a second mic on a guitar amp so it will add or cancel at a specific frequency. (This gets nearly impossible above about 900Hz, but could work below that.) Use a high- or lowpass filter to remove ranges where the effects of this phase manipulation are unwanted, keeping the positive effects. Again, it is critical to at least check your complete mix in mono to be sure you know exactly what these delays are doing to the mix in the various situations in which it might be listened to.

Overall, when dealing with phase—whether trying to eliminate anomalies or using it as an effect—knowing what to listen for will give you the control you need. **em**

Brian Heller is a freelance engineer, composer, and educator in Minneapolis. He teaches in the Sound Arts program at Minneapolis Community and Technical College.

Tighten it Up

Clean up sloppy playing with your DAW's piano-roll editor | By Jim Aikin

'm not a good keyboard player. When I try to record anything complicated, my MIDI sequence tracks are full of little problems. Fortunately, my sequencer's piano-roll editor has some slick tools for making performances presentable. But be warned: The process is not magic. Close listening and hand-editing are required (see Step-by-Step Instructions below).

Before hitting the Record button, I practice the part a few times. If I had an 88-note weighted-action keyboard, the raw recordings would be better, but I have to make do with a 5-octave synth keyboard. It's harder to control finger velocity with an unweighted keyboard, and that means more editing of velocity data. I usually record only a few bars at a time, then stop and practice the next few bars. As a result, my average finger velocity may change between takes—another reason to edit.

Quantization

Several things can go wrong during quantization. If you quantize the start times of the notes completely, the track will sound rigid and robotic. That can be a good thing, but not for tracks that you want to sound as if they were played by a human. If you need to quantize human-sounding tracks, use strength-based quantization set to about 50 percent. This will move each note halfway from its recorded position to the nearest quantize gridline. If the track contains various rhythms (such as a passage in triplets), you'll need to set the quantize grid to the appropriate rhythm values and then quantize groups of notes rather than quantizing the whole track at once.
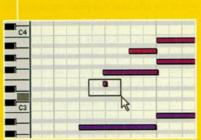
Notes that were recorded too far from the desired gridline can go further astray when you apply quantization because the computer will drag them over toward the wrong gridline. After quantizing, solo the track and listen to it entirely all the way through—these errors are easy to miss when you listen to the full mix.

Quantizing note lengths always sounds bad to me, so I don't do it. I always set the quantize preferences to leave the note durations unchanged.

Changing the note start time (either by quantizing or by hand-editing single notes) may affect the sound of any MIDI control-

STEP-BY-STEP INSTRUCTIONS

STEP 1: Select and delete any short notes that your fingers brushed accidentally.

STEP 2: Select the whole track (or a portion) and use 50-percent strength quantization to bring the notes closer to a tight rhythm without locking the performance completely to the beat.

STEP 3: Listen to the track while watching the piano-roll window, checking for sloppy rhythms. After turning off the snap grid, use the mouse to drag individual notes closer to the desired gridline.

change or pitch-bend data that you've recorded into the track. If the control data stays at its original location while an associated note moves, the musical expression may be warped. Conversely, if your sequencer automatically changes the timing of controller data when notes are moved, editing a chord can make a mess of the controller data. Check your owner's manual for details on how your sequencer handles this. (In **Fig. 1**, note the Move Controller checkbox, which tells Cubase whether or not to move controllers when the sustaining notes associated with them are quantized.)

Velocities

Some sequencers have macros for editing the velocities of a selected group of notes or of the entire track. For instance, you may be able to set all velocities to 50 percent of their current value and then boost them by 40 or so. If your original velocities vary from 12 to 90, after this process they'll vary from 46 to 85. This type of data compression will give you a smoother-sounding performance, but it can produce a sort of "steel fingers" effect, especially if the synth preset you're playing doesn't have a deep velocity response to begin with. All velocity

editing should be done while listening to how the synth responds. This is especially true with sampled sounds such as electric piano, where a small velocity change can cause a different sample to be triggered.

You can create crescendi and diminuendi very easily by drawing with the pencil tool in the velocity lane below the piano-roll. Another good use of the velocity lane is to leave most of the velocities unedited but examine the graphic display for isolated notes that stick out because the velocity is too high or are getting lost because the velocity is too low. You may be able to get a good performance by editing a few wayward velocities and leaving the rest of them alone.

Note Lengths

If you record a few bars at a time, either connecting your takes end to end or punching into the middle of long takes, check the piano-roll display closely for overlapping notes. A short note can be completely hidden in the piano-roll behind a longer note of the same pitch. On playback, the note-off message for the short note may cut off the long note before you expect it to. If you suspect

this problem, drag the long note up or down (being careful not to change its start time) to see if something is hiding behind it.

Note lengths are often irrelevant with sequenced drum tracks, but not always. If the envelope release time of the drum sound is short, shortening the note can cut off a portion of the sound. This can be a cool way to do gated drum sounds or just tighten up a flabby kick drum sample, so you'll need to know the features

of your drum sample player to make the best use of MIDI drum editing. **⟨em⟩**

Jim Aikin composes and records in a PC-based home studio. He also teaches cello and writes fiction.
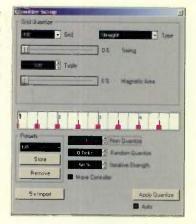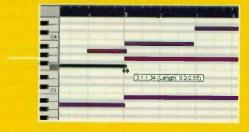
⟫⟫ **FIG. 1:** In the Quantize Setup box in Steinberg Cubase 4.5, you can set the percentage strength of the iterative quantization (a Cubase-specific term), select a non-quantize area (visible in red) within which note-on events won't be moved, and more.

STEP 4: Adjust velocity values as needed. In chords, you may need to edit individual notes by entering numeric data because it can be hard to grab a single note in a chord for graphic editing in the velocity lane of the piano-roll.

STEP 5: Leaving the snap grid switched off, lengthen or shorten individual notes with the mouse to produce uniformly played chords and smoothly articulated legato or staccato in melodies and bass lines.

STEP 6: If you're recording a piano-type part that was played with the sustain pedal, you may need to move pedal-down and pedal-up events to conform to the newly edited rhythms of the notes.

≫ Malström's Mod A and Mod B LFOs are cabled to modulate pan and feedback on the DDL-1 delay.

Back of the Rack

Use Reason's rear panel cabling to tweak up your sounds | By Len Sasso

The rear panels of most Propellerhead Reason and Record devices offer an enticing array of control voltage (CV) and gate inputs and outputs. Taking advantage of those is an excellent way to add motion and interest to your patches.

Most of the outputs are virtual-instrument modulators (LFOs, envelope generators, sequencers, and so on), but notable exceptions include the Gain Reduction CV on Record's Audio and Mix channels and the MIDI-converter outputs on the RPG-8 module. Along with the inputs provided on the rear panel, you can target almost any parameter by housing the module in a Combinator, using the Combinator's programmer to access that parameter, and then using the CV input of the associated Combinator knob or button.

One More for the Road

Many synths and samplers have limited modulators and routings. (Matrix-modulation synths such as Thor are the exception.) The NN-19 sampler, for example, doesn't have an LFO routing to control amp level (tremolo), but it does have the necessary rear-panel inputs and outputs—just cable them together and use the input-amount knob to control the modulation amount. (LFO outputs are tapped before the LFO's front-panel amount control.)

If a modulator that has an output is being used internally and has settings that don't match your purpose, add an instrument to the rack that has a similar modulator. Hold down the Shift key when you add it to prevent automatic audio cabling; instruments without audio cabling use a miniscule amount of CPU power.

Subtractor is the simplest to configure, and it gives you an LFO and two envelope generators. Malström provides outputs for its more versatile Mod A and Mod B generators, and its filter envelope. You can use Thor's modulation matrix to route most of its control and audio modules to four CV outputs, and you can scale the output by another module or input. For example, you could route a Malström Mod A to a Thor CV input, scale it with the MIDI mod wheel, and then route it to a Thor CV output to modulate another synth or effect.

When you want to modulate several targets with the same source or mix several sources, use a Spider CV module. The splitter section even provides an inverted output; try cabling the straight output to filter cutoff and the inverted output to level, for example. When you find it hard to discern the shape of a CV signal, temporarily routing it to modulate pitch will instantly reveal its contour.

When cabling envelope generators between modules, pay attention to how the envelope will be triggered. Keep in mind that the envelope will not be applied polyphonically to the target instrument. I like to combine the envelope source and the target instruments in a Combinator and limit the source module's keyboard range to a single key (which you do at the bottom of the Combinator's programmer). That way, I can trigger the envelope monophonically while playing the target instrument from the same keyboard. Alternatively, cable the gate of a Redrum pad (in Gate mode) to the Gate input of the instrument containing the envelope generator, and then dedicate that pad in Redrum sequences to triggering the envelope (see Web Clip 1).

Just for Effect

Reason's effects modules (even the basic half-rack effects) have modulation inputs that you can use to make tracks evolve and mutate. Here are a couple of examples for which you'll find Combinators in Web Clip 2.

The half-rack DDL-1 delay offers modulation inputs for both pan and feedback. I like to patch Malström's Mod A to pan with a random curve and a quarter-note rate, and to patch Mod B to the feedback with a square-wave curve and a very slow rate. That causes the delays (but not the dry signal) to move around in the stereo field while turning feedback on and off every few bars (see Web Clip 3).

The Thor synthesizer's flexible step sequencer makes it an ideal envelope-triggering source. With Thor's matrix modulation, you have access to all four of its envelopes. I often use those envelopes to modulate Scream 4's Damage Control and Body Reso knobs while using LFO 2 to modulate the Damage parameters and Body Scale knobs. (Use a Combinator to target the Reso knob.) The sequencer's Gate Len settings are especially useful when triggering envelopes (see Web Clip 4). ⌐ᴄᴍ

LISTEN TO EXAMPLES OF REASON EFFECTS ▶ EMUSICIAN.COM/BONUS_MATERIAL

⠶ According to ReverbNation's Jed Carlson, differentiation is the key to standing out among the multitude of musical artists on the Web.

Q&A: Jed Carlson

The record-label monopoly is broken; now what?

I n a world where fans can get music from anywhere in the world within moments and musicians are competing with bands all over the globe, it can be hard to break through. It usually starts with putting music up on the Web, but what should a musician do from there? Is it possible to turn a casual listener into a fan? And how can musicians turn that fan into a paying customer? To answer these types of questions, we turned to Jed Carlson, founder of the website ReverbNation (reverbnation.com). Although not a musician himself, Carlson's business and marketing background has helped him devise helpful strategies for using the many tools and websites available to connect musicians with their fans.

By Jason Feehan and Randy Chertkow

How has the Internet changed music promotion and marketing?

I think fundamentally it has leveled the playing field. The Internet has made global local. Bands can easily develop a fan base in Japan while sitting at home in North Carolina. I've asked a lot of local bands around here about what they think of the fact that they can now easily develop fans all over the world. Their immediate response is, "It's the greatest thing in the world." But then I ask them, "What do you think about the fact that

bands from all over the world can take the attention of your local fans in North Carolina away from you?" And they always say, "We didn't think of it that way." This is neither good nor bad, it just is. It just changes what musicians have to do to break through.

Powerful marketing tools like the ones we offer at ReverbNation are now available to pretty much everyone. Compare this to the old model, where the struggle was to get into the record-label pipeline. If you could just break into that monopoly, you could win

and get noticed all over. But now, everyone can run a sophisticated marketing department for their small business thanks to the Internet. Congratulations to all the musicians out there who wanted to break the monopoly! It's broken. Now everybody is in the same boat and there are a lot of people.

You just hit on one of the latest complaints by musicians: There are too many musicians now fighting for attention.

Yeah, I think that there's a lot more noise now than ever. And I think that is thanks to all of the tools and services and technologies that are now available to everybody. At the very least, the old record-label monopoly system was a filter. Whether you think that filter was good or bad is up to you, but it was a filter nonetheless. And I suspect the number of bands and music in the world is only going to grow.

To break through, you need to do more than just write good music. You need to have a good strategy and understand how to market and promote. That's how you become successful today. Part of that strat-

a little driveway underneath it, a cul de sac at the end of it, or an off-ramp so that people can pull over and actually deliver value for you such as joining your mailing list, which MySpace isn't set up for. The same applies to Facebook, Twitter, and all the social nets: Understand what they're good at and how to best use them in that context to get them into the funnel.

Now that the playing field is level and crowded, how do bands stand out?
It comes down to one thing: differentiation. When I talk to musicians, my advice to them is to make

that they have never been well connected with independent musicians is because there has never been a way to facilitate those relationships in a low-cost fashion. Just the legal expenses of licensing a song into a commercial merits staying away from independent music because you want the most bang for your buck. So you are going to pick a popular band that is going to move the needle. But that's changed.

I think the Internet enables matchmaking at a grand scale and at low cost that can sort of open the door for indie music that otherwise would not

Bands that are starting out have got to create that water-cooler experience.

egy is to develop a pipeline of fan relationships. And that's what we mean when we talk about the *marketing funnel*.

What is that?
The marketing funnel is a good way to understand that not all of your listeners are fans, and not all fans are customers. At the top of the funnel are the people who are exposed to your music; they're just listeners. They've stumbled on you through the Web, came to a show via a friend, et cetera. You will have a lot of these. However, listeners are not fans. Your goal should be to convert them into fans. This is done through your music, creating a memorable experience, and giving general listeners a reason to take the time to engage with you and your music. As you start converting listeners into fans, the funnel narrows. Fans can become customers. (You'll have more fans than customers.) And at the narrowest part of your funnel, some fans become promoters for the band. That is the ultimate: getting people to promote you and your music through word of mouth, the Web, the social nets, email, instant messaging.

How do you advise artists to effectively use social-media tools to build a following?
Use them with a purpose in mind. I think that's the missing link for most artists. They don't think strategically first. For example, consider MySpace: MySpace is like having a free billboard on a superhighway where there's millions of cars driving by every single day. But you can use ReverbNation tools to develop

yourself stand out. Deliver great experiences for your fans. For example, when you play live, make it something people are going to talk about at the water cooler the next day. The jam band provides some of the best models. Phish, Widespread Panic, and the Grateful Dead—I don't think they knew it at the time, but with CDs as a dead revenue source, these type of bands always focused on things beyond the CD. They peddled their merch, tickets, experience, and bundled packages. They gave the gift of an experience and they created word of mouth. Word of mouth is the most powerful marketing on the planet—and it's free.

How do you foresee musicians making money in the future?
I think the future of revenue in music for most bands is monetizing the experience musicians generate. I'll never forget this particular bottle of wine because my wife and I drank it when we got engaged, but I otherwise wouldn't remember that brand. Music is a context for everyone's life, creating that experience and selling that memorabilia piece of it to the über-fans, that's important. Bands that are starting out have got to create that water-cooler experience. Bands that have fans need to use the advanced techniques to harvest that fan base.

Beyond this, there are other big opportunities such as licensing and brand sponsorship. There's been a paradigm shift when it comes to the marriage between brands/entertainment properties and bands/music. I think the reason

have had access to these kinds of opportunities. We've already been helping with this. For example, in 2009, we started a program called Sponsored Songs with Microsoft Windows. Microsoft essentially underwrote a free song giveaway from 1,000 artists over a three-month period. They paid the artists $0.50 apiece for every song they could give away. These artists presented their music on a Microsoft-branded page and their MP3s have a Microsoft advertisement embedded in the digital album cover art. So when you play the song on your iPod, you see the ad come up and a "brought to you by your friends at Windows" kind of thing.

So it's not just about live shows then?
No, it's about the experience. The music, CDs, packaging, your website, videos, your newsletters—all of it. There's no denying that for a band just starting out, live performances are critical. It's a great way to connect to listeners and convert them to fans and customers. But you can do this on the Web, as well. My point is that a big reach is something you can buy; word of mouth is something you can't. When you know you've got an experience that can keep your fan pipeline full and growing, you have a business that anyone would want to invest in. <em>

Randy Chertkow and Jason Feehan are authors of The Indie Band Survival Guide: The Complete Manual for the Do-It-Yourself Musician *and* The D.I.Y. Music Manual, *and founders of the open and free musician resource IndieGuide.com.*

Cakewalk
SONAR 8.5 Producer (Win)
More than an incremental update

By Brian Smithers

Breaking with a longstanding tradition of releasing new integer versions each autumn, Cakewalk dubbed this major update to its Windows-only DAW SONAR 8.5. Reading between the lines, it seems the company is reluctant to get caught in the trap of measuring all of its engineering feats against a 12-month development yardstick. SONAR 8.5 is, in fact, the next major version, but existing version 8 users can download it for a reduced price as an incremental update. As before, there are two versions: the full-on Producer and the value-minded Studio. For a full list of the distinctions between the two versions, check Cakewalk's website.

Whatever Cakewalk might have planned for down the road, since the release of 8.5, the company has been at work enhancing and refining SONAR's functionality with free updates. You'll want to download and install these updates at your earliest opportunity as they make a big difference in certain features.

It's a Feature

The biggest news in SONAR 8.5 is the Matrix, a cell-based sequencing and live-triggering environment based on Cakewalk's Project5 sequencer, which is no longer in development (see Fig. 1). Cakewalk reports that the Matrix was not merely ported over from Project5, but was rebuilt from the ground up as an integrated SONAR feature. Still, Project5 users will feel right at home, as will Ableton Live users. The ability to have cell-based (nonlinear) triggering of audio and MIDI loops within SONAR's sequencing environment brings significant new workflow possibilities.

Between Beatscape, introduced in version 8, and its longstanding support of Acid-style looping, SONAR already does an impressive job of marrying linear sequencing and recording with loop-based production. The Matrix greatly enhances this by allowing you to assign loops to individual cells within a grid and then trigger cells in turn or in combination by MIDI notes. These notes can be recorded to a SONAR MIDI track and then edited. You could use the Matrix in live performance, although its most important trait is that it lets you make a musical performance out of your studio production. Copying and pasting loops to build an arrangement is a technical act that takes you out of the

FIG. 1: The Matrix view, modeled after Cakewalk's Project5, allows for nonlinear cell-based triggering of audio and MIDI loops.

creative moment, whereas triggering your samples in the Matrix feels more like playing an instrument.

Also new to 8.5 is an arpeggiator available on every MIDI or instrument track. This, too, bears a familial resemblance to Project5, and it brings a great deal of new power and flexibility to SONAR MIDI tracks. The Cakewalk FX Arpeggiator plug-in has been around for a while, but the new track-based arpeggiator is better at most things. It offers the parameters you expect from a full-featured arpeggiator: a variety of shapes; variable swing; control over note length, velocity, and pitch; and a variable flam. You'll find several hundred pattern-based presets, but although many of them are quite interesting, it's difficult to sift through them all in search of the perfect sound. It would be helpful if you could create and edit the patterns to fine-tune them for your song.

Lines of Succession

Session Drummer—which combines a multi-sampled instrument with pattern-based playback created from live performances—is now in version 3 (see Fig. 2), which includes 20 kits, including drum machines and classic rock kits from such names as Ocean Way Studios, Steven Slate, and Sonic Reality. Although Session Drummer doesn't give you all of the refined control over mic position and blend that, for example, FXpansion's BFD does, it does give you basic level, pan, width, and tuning controls, plus the option

FIG. 2: Session Drummer, now in version 3, offers new and improved drum kits, pattern-based sequencing, and flexible mix routing.

of routing instruments to individual SONAR tracks for authentic acoustic drum-style mixing. The kits sound great, and the patterns are quite usable. If a pattern requires refinement, then you can drag it directly to a track and edit it.

SONAR's Step Sequencer has been upgraded to version 2, bringing a number of refinements over the initial release. Although it still doesn't directly expose soft synth parameters, its handling of controllers has improved, with controllers and other parameters such as velocity displayed as a bar graph beneath a row of notes, and there are hooks to parameters for the synths that ship with SONAR. One of the most interesting additions is step probability, which allows you to vary patterns semi-randomly by specifying that certain notes are to play only occasionally. Although it doesn't absolve you of the responsibility for programming fills, it certainly allows you to take advantage of the convenience and power of the step sequencer while managing its potential for monotony.

AudioSnap, SONAR's beat-detection and time-correction utility, has been updated to version 2, but it still has some problems. Cakewalk is working on bug fixes, which the company plans to deliver in an incremental update.

Various other functions have been improved to smooth workflow, including the Send Assistant, which now offers the option of matching the track's gain and pan settings. Freeze and Archive functions now appear as dedicated buttons on each channel strip, and each plug-in offers a track Solo button. The Clips pane now displays vertical grid lines behind or on top of clips. When editing at a high level of zoom, you can prevent the window from scrolling by clicking on a clip immediately after starting playback. I'd still prefer a shortcut key to disable scrolling, but this is an improvement.

Plugged In

Even after all of these functionality improvements, Cakewalk saw fit to add to SONAR's plug-in complement. The PX-64 Percussion

Strip combines five processors—compressor/limiter, expander/gate, transient shaper, tempo-synchable delay, and 4-band semiparametric equalizer—with input and output saturation stages (see Fig. 3). The order of the five processors is customizable, and all parameters are fully automatable. Whether your percussion track needs a subtle tweak or a radical facelift, PX-64 just might be all you need.

Similarly, the VX-64 Vocal Strip combines a variety of processors designed to bring the best out of your vocal tracks. Its interface is a dead ringer for the PX-64, with the same bright, clear graphic display of the currently selected processor and drag-and-drop arrangement of processors. It includes a de-esser, a doubler, a
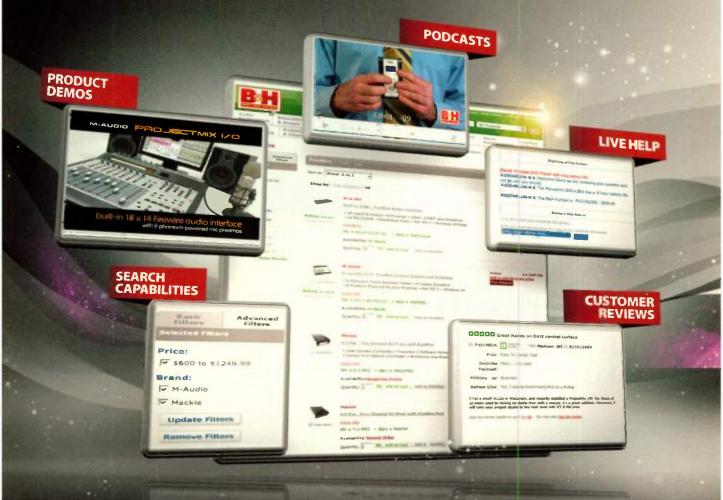
FIG. 3: The PX-64 Percussion Strip features five essential processors that can be arranged in any order, sandwiched between two stages of tube-style saturation.

synchable delay, a compressor/expander, and a 3-band equalizer, all surrounded as in the PX-64 by input and output saturation. The EQ includes variable tube saturation that can be selectively engaged on each band.

If the focus of this upgrade seems beat-heavy, that's because it is. If beats make your music go 'round, SONAR 8.5 is made for you. Although SONAR was already a strong platform for groove manipulation, its new arpeggiator and Matrix view, along with the improved step sequencer and AudioSnap function, bring it to a new level of flexibility and power. Add to that the Session Drummer 3 improvements and the PX-64, and SONAR 8.5 is sure to bring a smile to the drummer in you. For the other sides of your musical personality, there's plenty to smile at, too. ◄m

Brian Smithers is department chair of workstations at Full Sail University and the author of Mixing in Pro Tools: Skill Pack, 2nd Edition *(Cengage).*

Spectrasonics
Trilian 1.2 (Mac/Win)
Sampled bass takes a giant step forward

By Geary Yelton

Spectrasonics has plenty of experience sampling basses. The company first made its mark with Bass Legends, an acclaimed CD-ROM library for rackmount Akai and Roland samplers, earning a 5-star review in the April 1996 *EM* (available online at emusician.com). The plug-in Trilogy launched in 2002, pairing a top-shelf software instrument with 3GB of electric, acoustic, and synthesized basses. Spectrasonics has now upped the ante considerably with Trilian Total Bass Module, a virtual bass plug-in based on Steam, the sound engine at the heart of the company's flagship Omnisphere. (To get the most out of this review, I strongly encourage you to read my February 2009 Omnisphere review at emusician. com.) Trilian comes loaded with 34GB of samples that encompass the sample content of its predecessors and much more new material.

Trilian runs in AU, RTAS, and VST hosts; like Spectrasonics' other instruments, it isn't available standalone. It supports both 32-bit and 64-bit versions of Windows and Mac OS X. I used the AU version in Apple Logic Pro 9.1 running in Mac OS X 10.6.2 on my 3.2GHz 8-core Mac Pro with 10GB of RAM. Installation was straightforward and took less than two hours. For each of the five install discs, I had to specify the location of my hard disk and type in my administrator password.

Steaming Hot

Trilian's implementation of Spectrasonics' Steam Engine is practically identical to that of Omnisphere, but with slightly fewer synthesis capabilities and a few parameters optimized for bass, including release-layer triggering and round-robin playback. If you have both plug-ins, all samples and patches must be located in the same folders. Core library integration lets you open and play all of Trilian's sounds in

Omnisphere, where you can modify them using functions that Trilian lacks to create unison voices, for example, or apply ring modulation or granular synthesis. You can even create multis that combine Trilian and Omnisphere sounds. You can't, however, open Omnisphere's sounds in Trilian.

Like Omnisphere, Trilian is 8-part multitimbral. Eight parts make up a multi, each part contains a patch, and each patch has two layers. Each layer comprises an oscillator, two filters, and envelopes for filter and amplitude, and the two layers share six LFOs and two modulation envelopes—half as many mod envelopes as Omnisphere. Oscillators play multisamples called Soundsources. You can go deep into modifying any patch on its Edit page, and you have easy access to a more limited number of parameters on its Main page. Live mode lets you instantly switch to any of eight patches by clicking on it or using MIDI CCs or keyswitches. Stack mode lets you graphically arrange splits and layers containing up to eight patches by note range and velocity.

I was thrilled to learn that Trilian, unlike Omnisphere, lets you create a custom control layout on each patch's Main page by assigning knobs and buttons to any parameter. You can name your custom controls anything you want, essentially giving every patch its own user interface. The controls may be different every time you load a new patch; for example, the factory patch Bashing Punks in Rotterdam has knobs for Dimension, Bit Reduce, and Distortion, and Hell's Disco has knobs for The Abyss, Flames, and Pitch Twister. If you open the same patches in Omnisphere, however, those custom controls aren't available, but there's nothing Trilian can do that Omnisphere can't.

Trilian takes full advantage of the Steam Engine's sophisticated arpeggiator, which has more than 50 new presets in the latest version and now supports triplets and dotted notes. By dragging MIDI drum loops or other MIDI files into Trilian's arpeggiator, Groove Lock lets you play drum and bass patterns in perfect synchronization (see Web Clip 1).

I should mention at least two other new features in Trilian and Omnisphere. The Edit page offers several new filter types—most notably the Juicy filters, which quite effectively emulate real analog electronics. Additionally, the new sample file server allows 32-bit Mac systems to

break free of previous memory restrictions, so if you have sufficient RAM in your computer, you can load much larger multis and multisamples than previously.

Basses Loaded

The variety of bass patches in Trilian is stunning. I counted more than 600 synth basses, nearly 450 electric basses and articulations, and more than 75 acoustic bass patches. And that's not even counting more than 180 presets that furnish arpeggios and rhythm patterns or the more than 100 monophonic synths that extend beyond the bass range. All the patches draw from a pool of nearly 1,400 Soundsources comprising millions of individual samples.

Legacy patches include all the notes and articulations from Bass Legends—played by pre-eminent bassists Abraham Laboriel, Marcus Miller, and John Patitucci—but not any of that collection's runs, riffs, or grooves. All 3GB of Trilogy's core library is in Trilian. If you're upgrading from Trilogy (special pricing is available), you won't be able to import your user patches so you'll need to either re-create them manually or keep both plug-ins installed if you want to keep using them. Note that Spectrasonics doesn't support Trilogy in the latest versions of Windows or Mac OS X.

The selection of bass guitars is certainly comprehensive. You'll find 4, 5, 6, and 8-string electric basses—both fretted and fretless—played with techniques that include fingered, picked, muted, slapped, popped, and pulled, as well as legato, staccato, harmonics, x-notes, glisses, and various slides. I especially liked the Studio Bass patches, which were made from sampling Matt Bissonette playing the Music Man Bongo, a 5-string bass with dual humbuckers (see Web Clip 2). You get 4-band EQ and compression knobs on the Main page, and you can even dial in the mix of signals from

a DI and an Ampeg amp. And if you're a fan of the Chapman Stick (a fretted electric instrument most associated with bassist Tony Levin), there's an extensive set of articulations, both with and without effects.

Trilian's upright acoustic bass is everything you'd want a sampled bass to be. Trilian Ac 1 combines recordings made with a Neumann U47 and Wilson K-Pick-Ups, whereas Trilian Ac 2 was recorded through an AKG C 12 and Schertler STAT-B pickups. Another favorite is a Martin acoustic bass guitar with an extensive range of articulations. With dozens of articulations on hand, Trilian's acoustic basses are exceptionally versatile and lifelike.

Thirty sampled synths range from classics such as the Roland TB303, Korg MS-20, and Moog Taurus to recent entries such as the Dave Smith Instruments Tetra and Cwejman S1 MK2. Of special note is the large collection of sounds routed through the Metasonix KV-100, a lo-fi effects processor that can make the sweetest sound absolutely horrifying. And thanks to custom controls, real-time edit access is tailored to individual instruments.

The Real Lowdown

Trilian has many thoughtful features that solve problems and make using it a pleasure. For example, some of the patches are huge, and that's a problem if they consume more computer resources than you have to spare. Sample

FIG. 2: Trilian's powerful browser speeds up searching for patches, multis, and Soundsources. You can audition, rate, filter, and tag them using any search terms that help you find them most efficiently.

REVIEWS   TRILIAN 1.2

>> In our reviews, prices are MAP or
street unless otherwise noted.

thinning allows you to minimize a patch's resource consumption by limiting the number of velocities or round-robins it plays. You can also specify that a patch load samples for only certain scales or intervals, or train it to load samples for only the notes you play.

An instrument with keyswitches needs some visual indication of where those keyswitches are. Live mode gives you eight slots, enough to simultaneously load ample articulations for a single expressive instrument. Keyswitches for each part are displayed along with their patch names, and because they can be momentary switches, you only need to press F1 and play a note to slide up, for example, and release it to continue playing normally. In addition, Stack mode lets you remap articulations across the keyboard and velocity-switch between articulations.

In the latest version, Trilian's browser lets you rate patches on a scale from one to five, making it easier to relocate sounds you like at a later date (see Fig. 2). You can also group patches that work

THE SELECTION OF BASS GUITARS
is certainly comprehensive.

well together into projects for later recall. I only wish you could separate the Browser window from the rest of the GUI and expand its size.

Big Bottom
This was one of those reviews that kept sidetracking me while I was writing it. It's easy to lose track of time exploring so many incredible sounds, blending them and reshaping them into something new and different. I could easily spend entire days trying out patches, layering them with other patches, and processing them with effects and synthesis parameters to make sounds I could never achieve with a real bass (see Web Clip 3).

If you record music with a computer, I can scarcely imagine any situation in which Trilian

wouldn't be immensely useful. A bass player would have more immediate expression playing a real bass, of course, but it would take a roomful of gear and expert technique to get many of the sounds that Trilian supplies at the click of a mouse.

If you already own Trilogy, you should upgrade right away. The new synths or electric basses alone are well worth the money. Trilian is an outstanding bargain and a technological tour de force. I recommend it without hesitation. **EM**

Senior editor Geary Yelton has been playing bass for nearly 40 years and writing for EM for nearly 25. He lives in North Carolina and telecommutes globally.

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>>> FIG. 1: Pianoteq Pro gives you access to tuning, voicing, and design aspects of the physical model along with output and effects settings.

Modartt

Pianoteq Pro 3.5.3 (Mac/Win)

No longer the new kid on the block, but still the 900-pound gorilla

By Len Sasso

Modartt Pianoteq has come a long way since I reviewed version 1.0.3 in the March 2007 *EM* (available at emusician.com). In the interim, version 2 won *EM*'s 2008 Editors' Choice award for Most Innovative Product. In version 3.5.3, the underlying physical model has been significantly improved, and although many of the controls are the same, you'll find a host of user-friendly enhancements.

Pianoteq now comes in Standard and Pro versions, and the difference is in the new feature called Note Edit. The Pro version gives you note-by-note access to most Pianoteq parameters, whereas the Standard version gives you note-by-note access to only tuning and volume. (For full details on Note Edit, see the sidebar "Note for Note" on p. 52) The other difference is support for sampling rates up to 192kHz in the

Pro version vs. 48kHz in the Standard version. Both versions are now provided in standalone, as well as AU, VST, and RTAS plug-in formats.

Here I'll revisit a few of Pianoteq's more unusual features, but I'll mainly concentrate on what's new and improved. Because it is so different from sampled-piano virtual instruments, I strongly recommend checking out the previous review if you're not familiar with Pianoteq.

Model Behavior

Compared with sampled instruments, Pianoteq offers three advantages that bear repeating (see Fig. 1). Dynamics gradation over the full MIDI velocity range is transparently smooth because there is no transition between layered samples. Half pedaling—the effect of partially pressing the sustain pedal—

is realistically reproduced, again because it's not contingent on layer transitions. The third advantage is Pianoteq's odd but charming fourth pedal (called the Harmonic pedal), which simulates the effect of the sustain pedal, raising the dampers on all notes except those played, which allows you to play staccato notes with sympathetic resonance from other notes' open strings.

As a natural consequence of miking, sampled acoustic instruments always have some amount of room ambience, whereas for a physical-modeled instrument, dry really means dry. Pianoteq's reverb, although not yet on a par with the best convolution reverbs, is a significant improvement on the original.

Tremolo and a basic limiter have been added to the output-effects section. Tremolo might seem like an odd addition, but Pianoteq offers electric-piano and vibraphone add-ons for which tremolo is essential, and it's also a neat effect for acoustic piano.

The configuration dropdowns have been replaced by more extensive audio, MIDI, and preferences dialogs, along with separate Action sliders for damper position and duration, highest note with a damper, and key-release and sustain-pedal noise levels. Together, these give you much more control over the piano's mechanical noises. As in previous versions, some of the hotspots for the GUI's smaller controls are hard to hit with the mouse.

Most importantly, instead of simply choosing mono, stereo, or headphones, you now have control over the placement and output routing of up to five virtual mics (see Fig. 2). That allows for surround-sound miking and lets you mike from the player's perspective, which was previously not possible.

Modern Grands

Pianoteq has redesigned and renamed its two grand-piano instruments. The warm, rich-sounding C3—aimed at classical, romantic, and lyrical material—contrasts nicely with the bright, present M3, which is suitable for jazz, rock, and pop. Although it's billed as a pop/rock piano, the Chamber and Studio presets

of the YC5 add-on instrument (about $40) deliver an even less-edgy sound. Whereas the C3 and M3 are not models of specific pianos, YC5 is fashioned after a popular Japanese stage piano.

For each instrument, you'll find a variety of presets designed for specific situations.

(*Preset* refers to a configuration of Pianoteq's user-assignable settings. Instruments have many under-the-hood settings not accessible to the user, so in Pianoteq parlance, the same instrument can have many presets.) Pianoteq's Voicing and Design sections give you wide latitude to modify an instrument's sound, and the manual contains some helpful tutorials on modifying the presets. You can also use the Random button, and that often produces quite pianistic results (see Web Clip 1).

Looking Back

Modartt is an ardent supporter of the Keyboard Instruments Virtual Restoration (KIViR) project, which, in collaboration with several German museums, endeavors to create digital reproductions of historic instruments. These are usually in such delicate condition that restoring them to full playability is not possible (or too risky). The strategy, then, is to restore some of the notes as much as possible (without risking the full load of tuning all of the strings), and to then create a physical model from an analysis of the tuned strings. The resulting computer models are coupled with

digital keyboards to provide playable instruments in the museum. Modartt provides these models as free add-ons to Pianoteq, and the collection is continually growing.

The KIViR historical instruments are downloadable individually or as a bundle, and they are a must—you'll never find any-

MODARTT'S FREE KIVIR COLLECTION of HISTORICAL INSTRUMENT MODELS is **a must-have add-on.**

thing like these in a sample library. Whatever style of music you play or sequence, you'll have great fun trying it out on these vintage instruments, and they'll often inspire some new insight into your music.

Two of my favorites fall at opposite ends of the history range. The Kovács Cimbalom is a variant of the ancient psaltery; it became a common component in gypsy orchestras and was used as a concert instrument in the late 19th century. It has a 5-octave range, and the dampers are not raised unless the pedal is depressed—otherwise, notes die quickly. (Pianoteq includes a preset in which the keys do raise the dampers.) The Yamaha CP80 electroacoustic piano, popular in the late '70s and early '80s, featured a grand-piano action
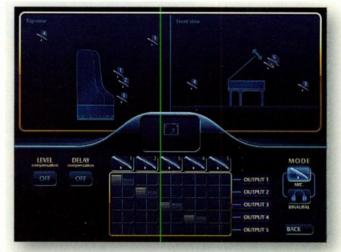

FIG. 2: The output-configuration panel lets you set the placement and delay of as many as five mics. That's also where you set the lid position.

and electronic pickups. It has a distinctively hybrid sound—not quite an acoustic piano but not an electric piano, either. Best of all, the Pianoteq version doesn't weigh 150 pounds.

I like to layer the Cimbalon and CP80 (using two instances of Pianoteq). I map the sustain pedal to Pianoteq's Harmonic pedal on the Cimbalon and map aftertouch to sustain. I leave the Cimbalon dry but use a little reverb and stereo tremolo on the CP-80, and I map a mod wheel to crossfade between the two (see Web Clip 2).

The remaining KIViR instruments are spread across three categories: harpsichord, pianoforte, and grand piano. Of the two harpsichords, the Grimaldi, which dates from 1697, has the brighter tone; the Pianoteq model lets you play the two registers separately or combined. The 1733 Blanchet has a soft, intimate sound with a stronger low end.

You get five modeled piano-forte instruments: Schmidt (1790), Schantz (1790), Walter (undated), Schöffstoss (1812), and Graf (1826). Each has its own Baroque-period sound. The presets replicate the authentic tuning reference, and temperament (which you can change).

The classic grand pianos are an 1896 Bechstein and a 1922 Erard. They provide a nice contrast to the C3, M3, and YC5, and are offered with a similar selection of presets. Finally, don't miss the new Bells and Carillons add-on—free but not in the KIViR bundle.

Note for Note

Pianoteq Pro's Note Edit window lets you set 22 physical parameters on a note-by-note basis (see Fig. A). You open it with the Note Edit button or by double-clicking the slider for any note-by-note-editable parameter. That reveals an 88-note bar chart, but fortunately, using it is a good bit simpler than setting the 88 bars independently. Click-dragging the window's title bar will detach the window, and you can then resize it for easier editing.

In most cases the default bar graph has three or four handles, indicated by blue markers along the yellow parameter curve and on corresponding keys of the mini keyboard at the bottom of the Note Edit window (not the main Pianoteq keyboard). The simplest adjustment is to grab one of the blue handles and drag in any direction. Dragging up or down creates a smooth curve of bars between the dragged handle and the closest handle on either side. Dragging left or right moves the handle to a different key and reshapes the transition between handles. You can create additional handles by double-clicking in the mini keyboard.

Here's where it gets both slick and tricky: If you click in the editing area but not on a handle, an individual yellow bar is created, affecting the setting for the corresponding note only. (In Octave Draw mode, bars for all notes related by an octave are affected.) If you click and drag a handle with that bar in its range

(meaning between it and the next handle), the individual bar's height retains its relative position to the handle and all other affected bars. That lets you mix note-by-note and handle-scaled edits. Four handy buttons along the top affect the curve globally: Random, Smooth, Scale, and Reset. For the first three, click and drag in any direction to increase the effect.

Editing a note's Spectrum Profile, which sets the levels of all the note's partials, works a little differently. Clicking on a note in the mini keyboard creates (or deletes) a yellow marker. With that marker selected, you set the level of each partial in the main bar graph.

>> FIG. A: The Note Edit window provides note-based settings for most physical-model parameters. Handy drawing modes and a Help button simplify the job.

A variety of draw modes (harmonic, comb, major, hair cut, and so on) are provided. Once you've created spectra for at least two notes, the spectrum for any note is interpolated from the spectra for the closest marked note on either side.

If the foregoing sounds daunting, a few moments mousing around will make it transparent. Of course, the devil is in the details, but a little randomizing, smoothing, and scaling will produce interesting if not always melodious results (see Web Clip A).

Looking Forward

In addition to YC5, the three other commercial add-ons (about $68 each) are also well-worth having, especially because Pianoteq's programmability lets you easily craft new sounds that retain the essence of the original instruments. The Electric Pianos add-on offers models called Rhody and Wurly—'nuff said. The Vibes add-on includes models V-M and V-B, and their range has been extended to five octaves. The Clavinet CL1 add-on is fashioned after the Hohner D6, but updated with an extended keyboard range, a sustain pedal, and continuous pick-up mixing. In these instruments, the new tremolo, wah-wah, and limiter effects come into their own.

Pianoteq is an outstanding virtual piano. It differs in sound, feel, and features from sampled pianos and will not necessarily replace your favorite. What it will do is give you a huge new arsenal of virtual pianos and keyboards. *em*

Len Sasso is an associate editor of EM. For an earful, visit his website, swiftkick.com.

»» FIG. 1: Notion 3 can easily produce large orchestral scores. Grouping of staves is automatic, and dozens of standard symbols are included. The floating tool palette is always available.

Notion Music
Notion 3 (Mac/Win)
Notation-based composing and performing software

— By Jim Aikin —

For much of his career, Johann Sebastian Bach was a church music director. He would have felt right at home with Notion 3. In the secular world, stage shows' music directors may appreciate Notion, too. Its feature set, while not power-packed or groundbreaking, is aimed squarely at people who are comfortable with notation-based composing and arranging, need a good library of orchestral sounds, and plan to use computer-generated sounds while performing with live ensembles that aren't playing to a click track.

The big news in version 3 is that Notion now offers support for third-party VST instrument and effects plug-ins, as well as MIDI output and ReWire. The program ships with a decent library of sampled orchestral instruments, but VST support allows you to augment the Notion library with third-party sounds from Vienna Symphonic Library, Garritan, and other developers.

Also new are Sequencer Staffs. (Notion's manual consistently uses *staffs* as the plural of *staff* rather than the more usual *staves*.) Though very limited, these resemble the tracks in a MIDI sequencer and are useful for recording parts that will be played by VST instruments or external hardware synthesizers. The ReWire support will be a lifesaver if you need to use Notion with sampled loops or audio-file playback. Notion will play back 16-bit, 44.1kHz WAV files, but it won't record them. In playback, it can only play the file starting from the beginning, which makes developing an audio-based piece awkward.

Peter Hamlin reviewed Notion 2.2 in the July 2008 *EM* (available at emusician.com). That review remains substantially accurate with respect to the notation-editing features. This review will take a close look at the version 3 features and offer a second perspective on some of the older features and functionality.

Duly Noted

Notion's tools for editing and printing notation are not fancy, but they're certainly good enough to get the job done quickly and efficiently. I use Avid Sibelius for my scoring needs, so I'm used to having complete control over the placement of symbols and the appearance of the page. Notion's approach is simpler—and to be fair, Notion costs far less than Sibelius.

You can enter your music with the mouse or by real-time or step recording from a MIDI keyboard. The keyboard shortcuts for choosing rhythms during step input (Q for quarter-note, E for eighth-note, and so on) are easy to learn. Placement of dynamics, articulations, and other symbols will naturally have to be done with the mouse. MIDI velocities are recorded during real-time input. Velocities are ignored during step entry, but will be controlled on playback by dynamics symbols placed in the part.

When using real-time entry on a grand staff (usually a piano part), you'll need to play both hands at once. Notion can't overdub a left-hand recording into a grand staff that already has right-hand notes, or vice-versa. However, you can record one hand in real time and then enter notes for the other hand using step or mouse entry.

Notion's default placement of symbols is generally good, and you can move them vertically or flip them from above the staff to below

it if needed, but horizontal placement and niceties like the curvature of slurs are not editable. Horizontal placement becomes a real issue in bar 1 if you have a metronome marking, a performance indication such as Swing, and a chord symbol such as D7 because they'll all have to be stacked vertically above beat 1.

Music directors who work with live ensembles will appreciate the included chord symbols, guitar tablature, guitar fretboard diagrams, alternate noteheads for drum parts, and multiple verses of lyrics. The fretboard diagrams let you choose from among a number of ways to voice any given chord, and you can edit the finger placement if you need to. If your lead vocalist needs a piece transposed to a different key, Notion will do it with a few quick clicks. Even the chord symbols will be transposed (although the fretboard diagrams will go blank after a transpose operation until you reselect them).

Printing out parts is quick and easy, and multibar rests are supported in parts. A nice-looking, pseudo-handwritten jazz font is supplied. To control the page layout, you can separate bars or force them to stay together. Automatic bar numbering is available.

Notion is not designed to handle the needs of classical sheet-music publishers. It won't do cross-staff or cross-bar-line beams, for instance, nor nonstandard key signatures. But it does clef

changes (with automatic staff transposition of already entered notes that follow the clef change), compound time signatures, tuplets, repeat bar lines, and most of the other things that you may need to put together and print out charts.

Under-Staffed

Notion's Sequencer Staff feature gives you a little more control over parts that are mainly intended for synthesizer playback. You can edit the start times and durations of notes freely, a function that traditional notation doesn't allow. You change start times, durations, and velocities by selecting single notes and tapping the computer's arrow keys; it's slow and fiddly, but it works. Overdubbing new notes into a Sequencer Staff that already has notes is supported. (Overdubbing doesn't work in ordinary notation staves, as mentioned earlier.)

Note velocities in normal staves can be edited numerically, one note at a time, by switching to Sequencer Overlay mode. The velocity-overdub feature, which you can use to replace velocities in already recorded notation tracks, doesn't work with a Sequencer Staff, but you can convert a Sequencer Staff to an ordinary staff, after which velocity overdubbing is allowed.

If you record MIDI pitch-bend or controller data, you'll see it displayed graphically above the Sequencer Staff, but again, you can't edit this data. If you didn't get it right the first time, your only option is to undo and record it again. Recording program-change data is not supported, so you'll need to manually switch your external hardware synth to a new preset before the next song starts.

I'm mystified why Notion's MIDI implementation isn't more robust. MIDI sequencers with features far more powerful than what Notion's Sequencer Staffs offer were common 20 years ago. Fortunately, if you need better MIDI editing, you can buy a real sequencer and take advantage of Notion's ReWire implementation. Using ReWire to connect the two programs, you can use Notion for composition and printing and get far more musical performance out of your VST and hardware instruments.

On Tap

If you need to do a live show using sampled orchestral instruments, you'll find Notion's performance features quite useful. Common symbols such as dynamic indications, slurs and

FIG. 2: Each channel in the Notion mixer has a level fader, pan pot, and mute and solo buttons. Each channel has four sends, one for each of the mixer's four aux buses, and can have up to four insert effects.

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World Radio History

ties, accents, and staccato dots will be executed on playback. (According to Notion Software, properly programmed third-party libraries will respond to these symbols automatically, without any special programming on the user's part.) You can vamp until the singer is ready or jump forward or back to a rehearsal mark to coordinate with spontaneous or unplanned changes onstage. A large display shows the current bar and beat, the current rehearsal mark, and so on.

Using tap tempo, which Notion calls NTempo, you can tap either a MIDI key or a computer key in the ASDF row. Notion will play a quarter-note's worth of data and then wait for your next tap. You can also switch from manual tapping to Auto-Cruise mode; I found that this requires a bit of practice. If your last two taps are a little too close together, when you switch to Auto-Cruise, Notion will gallop off at a faster tempo than you intended.

If your music contains a passage in quarter-note triplets or some other situation where the pulse changes, you can set up an NTempo track. Your tapping will then advance the clock in terms of the note values in the NTempo track. For a dramatic pause in the middle of a piece, you can tap the Q key, which will shut off any sounding notes. For real-time volume control of Notion's audio output, your only option is to grab mixer faders with the mouse (see **Fig. 2** and the Online Bonus Material "The Notion Mixer").

Sometimes a Great Notion

The new features in Notion 3 will be welcome to music directors who need higher-quality sounds, audio tracks, or control of external hardware synthesizers. The program's MIDI sequencing is primitive, but thanks to the ReWire support, you can use a better sequencer along with Notion. The score and part printing is more than good enough for printing out parts for your players or conductor to use at a gig. Being able to use a flexible tap-tempo option in performance is bound to appeal to anyone who works with mixed groups of traditional instruments. Notion will have less appeal for technologically savvy pop musicians, but it fills a definite niche in the market, and does so in a reasonable manner. **em**

Jim Aikin writes about electronic music technology, plays classical cello, and sometimes programs text-based computer games.

AKAI

LPK25 and LPD8

By Asher Fulero

One company refining another's idea is a common occurrence, so it's not a huge surprise to see Akai stepping out to refine Korg's idea of the low-cost, miniature MIDI controller. The company's new LPD8 and LPK25 (Mac/Win, $69.99 each) are surprisingly solid, reasonably affordable, and, though not as small as Korg's Nanos, undeniably portable. But the units' biggest selling point is that they are more playable than their competition. This focus on playability is evident throughout, and it makes them fun to use.

YOU CAN TAKE IT WITH YOU

I found the 25 piano keys on the LPK25 to be significantly more usable than those on the Korg NanoKey; the black keys are raised, and the mechanism doesn't feel like it's going to snap. The keys are similar to those of a Hohner Melodica in size, and anyone used to playing a Melodica will feel right at home. The keys are also surprisingly responsive, with a gently increasing resistance as the key descends, offering a nice imitation of a real action. That helps make the dynamics more reliable and makes it a lot easier to play by feel. Lighted Octave buttons on the left of the unit let you quickly navigate to whatever note range you desire, and a Sustain button offers basic pedal functionality.

The LPK25 includes an arpeggiator with eight beat divisions, six patterns, and four octave settings. With the included software editor, you can sync the arpeggiator to your host DAW or to the LPK25's internal clock (controlled by the Tap Tempo button), and you can set a default latch position. You can save these settings, as well as arpeggiator parameters, global MIDI channel, transposition, and the number of taps required to set a new tempo, and then load them into one of four preset programs. You access

these programs on the fly by holding the Program button and pressing one of the top four keyboard notes. A USB port lets you save and load presets on your computer. On Mac OS X 10.6, I was able to do this in the background while Ableton Live was running, thereby swapping presets on the fly.

NEW PADS ON THE BLOCK

The LPD8 is similarly focused on playability, with eight drum pads and eight knobs just begging to be tweaked. The pads feel great, offering an even response across the pad, just the right amount of give to keep your fin-

included. The drum pads each have three modes, switchable on the fly by pressing the buttons for pad, MIDI CC, or program change. Using the included software editor, you assign each drum pad a custom MIDI Note, CC (output value determined by velocity as you play), or program number, and you set their default to be momentary or latch. Similarly, you can assign each of the knobs a MIDI CC number and value range. As is the case with the LPK25, you can save this data, along with a global MIDI channel, as a preset and load it into one of four preset banks.

>> The LPD8 (top) and LPK25 are surprisingly versatile and playable for their compact size.

gers in time when playing quickly, and bright backlit rings that illuminate when the pad is triggered (or latched, depending on the pad setting). I wasn't thrilled with the choice of knobs; the LPD8's low-profile plastic knobs are too squat and wide for making quick, precise movements. They are passable for basic use, but endless rotary encoders with LED rings would have been fantastic. But the eight drum pads are of high enough quality to justify the unit's price.

I was hoping the LPD8 would include roll and flam settings like Korg's NanoPad (which lets you play rolls and flams using its X/Y pad), but sadly this feature was not

I found both the LPK25 and LPD8 fun to play, and I was able to create far more usable results than with their Korg counterparts. I was a little disappointed with the LPD8's knobs, as well as with the confusing design decision to have the pad numbers count up from the bottom left but the knobs count up from the top left. These complaints aside, I would recommend both units to anyone looking for an inexpensive, ultrasmall MIDI controller that they can carry around without sacrificing playability or sturdiness.

Overall rating (1 through 5): 4
akaipro.com

SOFTUBE

Passive-Active Pack

(Mac/Win)
By Michael Cooper

Three equalizer plug-ins comprise the Passive-Active Pack software bundle. The Passive Equalizer is modeled on the passive equalizer in the vintage Neumann PEV 930-00 console. The classic, Swiss-made Filtek Labo mk.5 console's equalizer was modeled to produce the Active Equalizer plug-in. The models for both hardware equalizers were combined and modified to produce the hybrid Focusing Equalizer.

The Passive-Active Pack is cross-platform and available in three formats: AU, RTAS, and VST. I reviewed Version 1.0.2 of the AU versions of the plug-ins in MOTU Digital Performer 6.02, using an 8-core Apple Mac Pro running OS X 10.5.4. All three plug-ins operate in either mono or stereo configuration and at standard sampling frequencies up to 192kHz.

START A BAND

Passive Equalizer is a 3-band affair. The filters for high and low bands are so broad that they act similar to shelving filters. The mid (*presence*) band has a bell-curve filter whose variable bandwidth, which is automatically set and not user-controllable, is widest at lower center frequencies. As in passive hardware equalizers, adjusting one filter affects the others.

Passive Equalizer centers its low-frequency adjustments around 60Hz for its

low band and 10kHz for its high band. In these bands, you can boost (up to 9dB) or cut (as much as 15dB) in 3dB steps. In the presence band, you can boost (in 2dB steps up to 8dB) at any of seven center frequencies ranging from 700Hz to 5.6kHz; you can't cut in this band. As with the other two plug-ins in this bundle, an output-volume control is provided, which ranges continuously from 12dB gain to infinite attenuation.

Active Equalizer also has three bands, not counting additional low- and high-cut filters. You can boost or cut up to 16dB at any of 12 frequencies in each band. Select a fixed broad or narrow Q for each band. The low- and high-cut filters each offer five corner frequencies and an 18dB-per-octave slope. All bands and filters can be independently bypassed.

With Focusing Equalizer, begin by adjusting corner frequencies for high- and low-cut filters. Within the resulting frequency range, the center frequencies for low, mid, and high bands of EQ are automatically chosen for you. You can only boost the low and high bands, whereas the mid band provides both boost and cut. Choose either passive or active EQ mode, or bypass both. Active mode provides narrower filter responses and steeper high- and low-cut filters (18dB per octave vs. 6dB per octave for passive mode).

Focusing Equalizer also provides saturation controls, which you can use with or without all EQ bypassed. Dial in the amount of tube-like saturation you want, and apply it to just the low or high frequencies or the entire spectrum.

TEST TONE

Passive Equalizer is one of the sweetest, most musical digital EQs I've ever heard. It uncannily produces the smooth, sweet, round, focused, and silvery tones of a passive analog equalizer. Passive Equalizer's broad filters are not meant for surgical tweaks. The wide gap of frequencies between its low band (60Hz) and its lowest mid frequency (700Hz) make it inadequate for use on tracks in need of upper-bass and low-midrange tweaks. That said, Passive Equalizer gave me absolutely magical results on bass guitar, snare drum, and lead vocals.

Active Equalizer, with its overlapping bands and variable Q, allowed the most precise tone sculpting of the three plug-ins. That said, I wish its low-cut filter went lower than 80Hz to roll off rumble. Active Equalizer lent pleasing coloration to lead vocals. By boosting low and high bands and cutting mids, I made a kick drum sound meaty and nicely clicky.

Focusing Equalizer was the most aggressive and colorful-sounding of the three plug-ins. It provided gorgeous, creamy tones that sounded slightly veiled compared to those produced by the pristine Passive Equalizer. The saturation controls added wonderful size and luster to vocals, electric guitar, and bass.

Focusing Equalizer's lack of frequency and gain readouts in all three bands made equalization tasks a mostly by-ear exercise, and the inability to cut gain in low and high bands was somewhat limiting. Focusing Equalizer nevertheless brims with character, making most other EQ plug-ins sound boring and dimensionless by comparison.

All three plug-ins imposed very negligible CPU draw. Despite its ergonomic flaws (which are partly due to faithful modeling of the original equipment designs), I gave Passive-Active Pack a "5" rating because of its superb and multidimensional sound quality. At $179.99, this bundle is an outrageous steal. High-quality equalization doesn't get better than this.

Overall rating (1 through 5): 5
softube.com (distributed by MV Pro Audio, mvproaudio.com)

>> Focusing Equalizer is one of the three high-quality equalizer plug-ins included in Softube Passive-Active Pack.

PLATINUM SAMPLES

Jim Scott Rock Drums, Volumes 1 and 2

By Marty Cutler

You can never be too rich, be too thin, or have enough drum sounds. Third-party developers that capture the seemingly endless variety of sounds drums can make, the remarkable attention to detail embedded in the instruments, and the high level of expression the software affords make FXpansion BFD 2.1 one of my go-to software drummers.

Platinum Samples, one of these third parties, is responsible for drum-kit collections from recording engineers Andy Johns and Joe Barresi, whose credentials include work with many high-profile artists. Platinum Samples' most recent contributor, engineer Jim Scott, has won several Grammys for his work. His Rock Drums collection ($319.99) is a two-volume set totaling 11 kits. Volume 1 is available separately for $249.95, and volume 2 for $129.99. You've probably heard the kits sampled for this

listed as a Pearl-Ludwig-Tama Kit on Acid. Platinum Drums says this kit is in the works as a free download for registered users, and you can already download the cymbals.

Because I play drums on a keyboard rather than a set of pads or a MIDI drum kit, I appreciated the extensive velocity layering in a very tactile, musical way; this was especially true for the hi-hats, which can sound lifeless and *ticky* when there aren't enough samples at varied dynamic levels.

REACT NATURALLY

I find it hard to make music with drums that sound as if they were recorded in an anechoic chamber. Thankfully, not much is pristine about these kit pieces; for instance, the snare samples from Kit 1 (a Ludwig Black Beauty) are as real and attitude-laden as you can get. The lightest velocities produce natural-sounding ghosted taps. You can really hear the strainers react to more forceful hits, and the open hats ring sympathetically (see **Web Clip 1**). Round-robin mapping is very much in evidence, keeping successive hits varied and interesting.

꛴꛴ To create Jim Scott Rock Drums, Jim Scott, one of L.A.'s busiest recording engineers, meticulously sampled his personal collection of classic drum kits and formatted the massive library for FXpansion BFD 2.1.

room ambience gives it a slightly edgy flavor that is better suited to alt-country and more aggressive jazz.

HEAVY-HITTERS

It's difficult to list my favorite kits here, but my favorite snares include Chad Smith's snappy, tight snare and Steve Ferrone's Ludwig Black Beauty (see **Web Clip 3**), along with the bell-brass Tamas from Kit 4 (played with Blast Sticks), the beefy but tight Chad Smith kick, and its polar opposite, the thunderous orange-sparkle Kent kick from kit 4. Kits 2's brushed toms and hi-hats are unhyped and intimate—as are its cymbals, which would fit in just as well with less-ambient brush kits.

Although the volumes are clearly labeled Rock Drums and they drip with rock attitude, I'd have no problem using them in any style that required aggressive, ambient drum sounds. Scott's notes on the accompanying poster detail the stories behind his kit-piece acquisitions in loving detail. It's clear that he's proud of his collection, and it's obvious he lavished a lot of attention on the sampling and assembly of this product. Check the demos out on Platinum Samples' website. **em**

Overall rating (1 through 5): 4
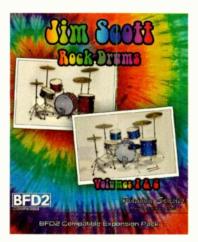
plantinumsamples.com

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collection used on recordings by such artists as Barenaked Ladies, Robbie Robertson, the Dixie Chicks, Red Hot Chili Peppers, and Tom Petty. Scott sampled the kits at his own PLYRZ Studios through a Neve 8048 console and plenty of analog outboard equipment.

GREAT SCOTT

You'll need to be exceedingly patient to install all seven DVDs in volume 1 and four in volume 2. Luckily, you can choose to install smaller sound sets at the cost of fewer Velocity layers. I chose the full 118GB installation. One thing I noticed after installing the volumes was a discontinuity in the numbering system; there was no Kit 5, which is

Kit 2 snares (a Ludwig Supraphonic Chrome) show similar artifacts, but the snares are less ringy (see **Web Clip 2**). Higher velocities bring in a more pronounced metallic edge. Hats have a slightly less-percussive attack, but with an odd, pingy resonance that becomes a little too pronounced for my taste at higher velocities. On Kit 2 played with brushes, the kick uses the same bass drum and felt beater as the regular Kit 2, but it's set up with an additional resonator track to ring out in all its boomy glory. If you need a kick that gets out of the way a little faster, BFD 2 makes it easy to swap out with one of the others. I doubt that that this kit is intended for cocktail-hour club dates; its overall

World Radio History

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GREAT SCOTT

You'll need to be exceedingly patient to install all seven DVDs in volume 1 and four in volume 2. Luckily, you can choose to install smaller sound sets at the cost of fewer Velocity layers. I chose the full 118GB installation. One thing I noticed after installing the volumes was a discontinuity in the numbering system; there was no Kit 5, which is

Kit 2 snares (a Ludwig Supraphonic Chrome) show similar artifacts, but the snares are less ringy (see **Web Clip 2**). Higher velocities bring in a more pronounced metallic edge. Hats have a slightly less-percussive attack, but with an odd, pingy resonance that becomes a little too pronounced for my taste at higher velocities. On Kit 2 played with brushes, the kick uses the same bass drum and felt beater as the regular Kit 2, but it's set up with an additional resonator track to ring out in all its boomy glory. If you need a kick that gets out of the way a little faster, BFD 2 makes it easy to swap out with one of the others. I doubt that that this kit is intended for cocktail-hour club dates; its overall

Overall rating (1 through 5): 4
plantinumsamples.com

ACOUSTIC PRODUCTS

ANALOG SYNTHS

FURNITURE

MASTERING

SOFTWARE

Customize your all-in-one MOTU studio

Run DP7 on Apple's latest multicore iMacs, connect the UltraLite-mk3 with HYBRID technology, then customize your system with the very latest desktop studio technology.

Euphonix Artist Series
High-end console technology

Together, **MC Transport**, **MC Control**, and **MC Mix** give you access to almost every facet of your Digital Performer 7 project for an unprecedented "hands-on" editing and mixing experience. In-depth, custom-programmed support for the EuCon protocol brings Euphonix' high-end console technology to your MOTU desktop studio in a compact design that fits perfectly in front of your iMac.

Apple iMac
The all-in-one that's one of a kind.

Apple introduced the first iMac more than a decade ago. And the design philosophy remains the same: an all-in-one computer that's as powerful as it is beautiful. A 21.5-inch or 27-inch display with edge-to-edge glass covers nearly the entire front of the unified enclosure. This latest iMac has a 3.06GHz Intel Core 2 Duo processor and comes loaded with 4GB (up to 16GB) of RAM and a 1TB hard drive. It's the perfect all-in-one centerpiece for your MOTU studio.

MC Transport   MC Control   MC Mix

MOTU UltraLite-mk3 Hybrid
The world's first hybrid FireWire•USB2 I/O

Connect to any current Mac or PC via either FireWire 400 or hi-speed USB2 and enjoy the same pristine audio quality and zero-latency on-board mixing performance either way. Includes classic reverb, modeled analog EQ, vintage compression, and advanced signal analysis tools like a full-screen real-time FFT display, spectrogram "waterfall," oscilloscope, X-Y plot, and phase analysis.

HYBRID Technology

Presonus ADL 600
2-channel tube preamp from DeMaria

The ADL 600 is a high voltage two-channel mic preamp designed by world-renowned analog tube circuit designer Anthony DeMaria. A unique high-voltage Class A discrete design includes three vacuum tubes per channel for maximum headroom and tonal character. Built by hand in the USA with the finest components including military grade tubes, custom transformers, and analog metering. The ADL 600 is big, warm, smooth, and clear; a sound like no other for your MOTU studio.

Neumann TLM 102
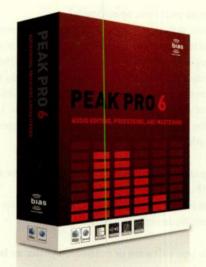### Sweet little microphone — huge Neumann sound

Don't let its small size fool you — only 4.5" tall — the new TLM 102 large-diaphragm microphone makes a perfect addition to any studio, from project to professional. Featuring transformerless circuitry, a foam-lined grille to reduce 'p' and 's' sounds and an SPL rating of up to 144 dB, the TLM 102 brings the legendary Neumann sound home at a price everyone can afford.

BIAS Peak Pro 6
Evolution of an award-winning standard

Whether you're a musician, sound designer, audio editor, multimedia producer, or mastering engineer, Peak Pro 6 offers more creative potential than ever before. Used side-by-side or launched directly from within DP7, Peak Pro 6 streamlines your workflow with industry-renowned sonic quality and precision. For additional mastering, restoration, and DDP 2.0 delivery power, step up to Peak Pro XT 6.
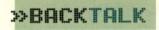
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Q&A: Simon Franglen

Programming synths for the *Avatar* soundtrack

By Mike Levine

Simon Franglen is a respected British producer, keyboardist, and synthesist. He was one of the synth programmers working with director James Cameron and composer James Horner on *Avatar*. He had worked with them previously on *Titanic*.

>>> Franglen also worked on the *Titanic* soundtrack.

COURTESY WETA

When in the production process did you get involved in *Avatar*?
What happened was that James Horner had started the film already before I was brought in. He had his team already put together.

Who was on it?
There was Ian Underwood [Horner's longtime synthesist], Aaron Martin—who'd been working with James over a period of time, who was there to do more of the technical synth programming—and Simon Rhodes, who was [Horner's] engineer for the last 12 years. They'd been holed up in a house in Calabasas, [Calif.,] where they had put together a rather ad hoc studio, with the control room being the master bedroom, and the master bathroom being the machine room—I wish I was kidding, but you had a bathroom with about 30 different computers in it—all put together, and then a control room setup with everyone having their own substation feeding into Simon Rhodes' master Pro Tools rig.

What was your role, specifically?
The idea was that I was going to arrange and create the non-orchestral portion of the score under James' direction. So I was sort of allowed to be the guy in the corner who said, "Hey, why don't we try this?" James would write out and sketch out a particular cue, let's say the flying sequence—which is, I think, one of my favorites on the film. And he'd say, "Have you got any rhythmic ideas?" And in this case, I was then given a free hand to program up what I thought would happen in terms of the rhythmic side of the film.

What program were you working in to do the sound design?
There was only one program; all the programming on the film was done in Pro Tools. All the sequencing, everything.

Did you use software or outboard synths, or both?
Between the three people who provided synthetic [parts] on the film,

there's not a single hardware synth used. It was entirely soft synths. Probably the vast majority of what I was using was [Native Instruments] Kontakt 4.

What others?
[Spectrasonics] Ominsphere, Stylus RMX. I used Trilian a little at the end; it hadn't come out until I was getting to the end of the project. [FXpansion] BFD, I was using BFD2 on some of the stuff. I like [Native Instruments] Battery, especially for drum work. It's very nice the way I can build things in that. And there's some little boutique ones like [FXpansion] Strobe. I used a bit of [Native Instruments] Massive. But it wasn't really a synth score.

So you were doing a lot of sample manipulation: repitching samples and changing their timing?
Absolutely. There's a bit where you see all the "wood sprites," as we called them, floating down onto Jake. He's in the night forest. And I remember taking a lot of things like vocals—I had some ethnic vocals—turning them backward and putting them through [SoundToys] Crystallizer.

What's it like working with a composer like Horner?
James likes writing long cues. There are some guys you work with where you're working on two- or three-minute sections and you paste it together. James is one of those guys who likes a nine-minute battle cue. And it's written out as a nine-minute piece with maybe 350 bars and 200 tempo changes (see **Web Clip 1**), and a lot of different meters, and so on. So you need to have tools that allow you to adapt quickly. ⟨em⟩

Mike Levine is EM's editor and senior media producer.

"Make Money with Your Music No Matter Where You Live"

Martin Haene – TAXI Member

My name is Martin Haene, and since I was a kid, I dreamed of earning my living with my music.

Sound familiar?

I live in Switzerland, so I thought it was nearly impossible. After all, who wants to hear music from a guy with a home studio who lives 5,920 miles away from Hollywood, in a country the size of West Virginia?

What Didn't Work…

I tried sending out hundreds of unsolicited CDs, but it was expensive and unproductive. Just like you, I needed a "vehicle" for my music.

What *Did* Work!

I discovered TAXI in 2006 and quickly understood the value of having a well-connected U.S. based company "filter" my music and get it to the right people. I thought the people on the receiving end would be much more likely to listen to well-targeted music from a trusted source, rather than the *hundreds* of unfiltered songs they get every day.

The number of my Film and TV placements has increased *dramatically* since joining TAXI: In just 2 ½ years, my music has been used in international TV Shows like *Lincoln Heights* (ABC Family), *Stylista* (CW Network), *Degrassi* (CTV) as well as feature films like *Graduation* (Independent) and *Bring it on – Fight to the Finish* (Universal). I've also licensed four of my tracks for a Sony/BMG, Latin Music CD that was released internationally.

Your Music as Your Retirement Fund

Having also signed deals with several top publishers through TAXI, I have little doubt those will turn into an ever-increasing number

of Film & TV placements down the road. I'm beginning to see how my music could easily become my retirement fund. How many musicians can make that claim?

You can if your music is top notch and you use your TAXI membership well.

More Than Just Great Contacts

TAXI is also the perfect way to learn what the market needs and how to produce it. Every member gets two free tickets to TAXI's annual convention, the Road Rally. It's such a great learning and networking experience that it's *more* than worth the price of the plane ticket – even from Switzerland!

I've also made friends, contacts, and met co-writers by becoming part of TAXI's online community. The possibilities are limitless if you take advantage of them. And the best part is, you can make money with your music no matter where you live.

Where do *you* live and what are you waiting for? Call TAXI!

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