

OCTOBER 2007

ProAudio Review

The Review Resource for Sound Professionals

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In This Issue!

- **Lipinski Sound PowerStand**
Transparent Studio Wattage!
- **Steinberg Sequel Software**
Powerful Recording Tools!
- **Neumann KM D Microphones**
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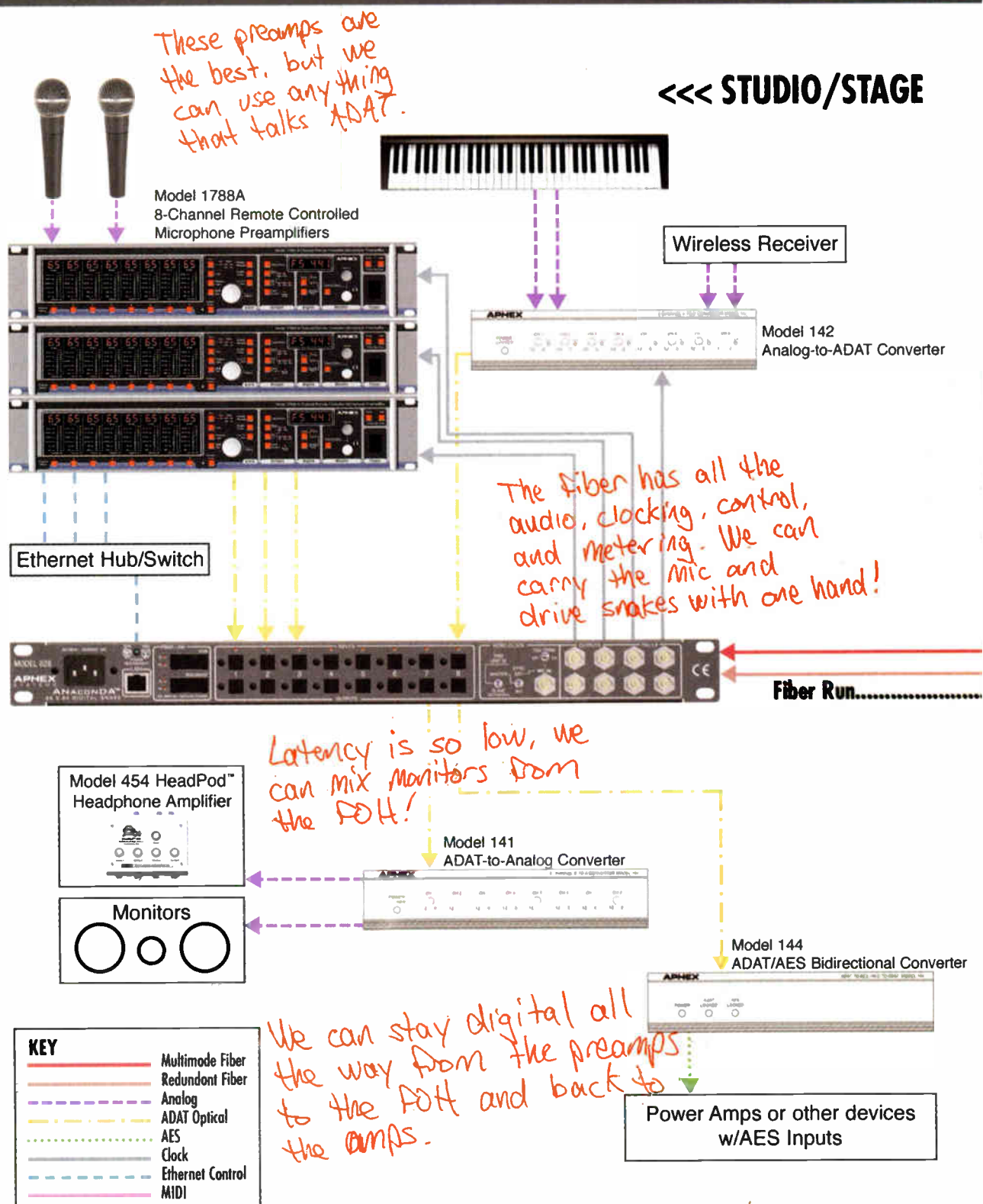
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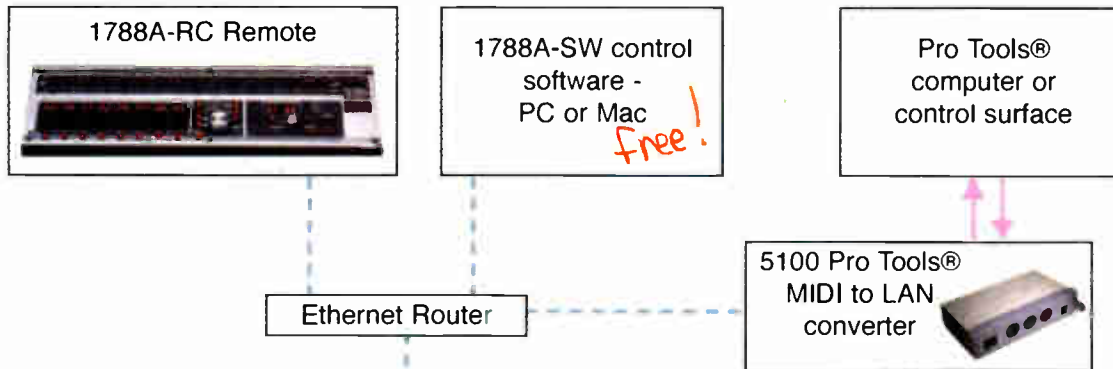


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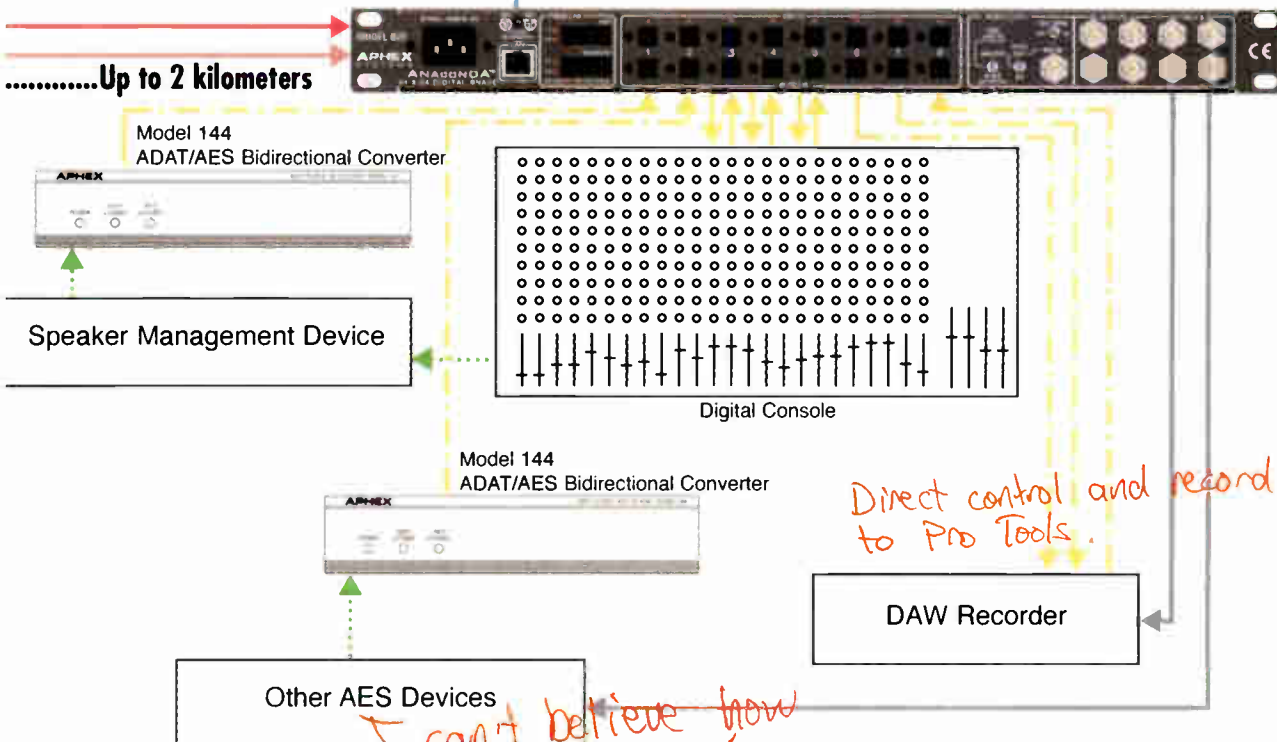
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The Review Resource for Sound Professionals

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FROM THE PUBLISHER

John Gatski

The Excellence of Gear

There is something about buying a new piece of audio gear. It's that emotional connection all of us audio fanatics feel when we first see a new product. Imagine that same feeling at a trade show, like the annual AES, where there are hundreds of new products on display from the various manufacturer booths. Man, that is nirvana for me.

As you read this column, our crew of *PAR Excellence Award* judges will be under the intoxicating spell of pro audio electronics — searching

for the hot new products to bestow our award. The *PAR Excellence Award* began in 1996 as a way to recognize the new products at AES that were being shown for the first time.

Over the years, many manufacturers have told me they really appreciate our award because it is based on the potential of a product's usefulness to pro audio — and because it acknowledges the design effort that goes into making a good product.

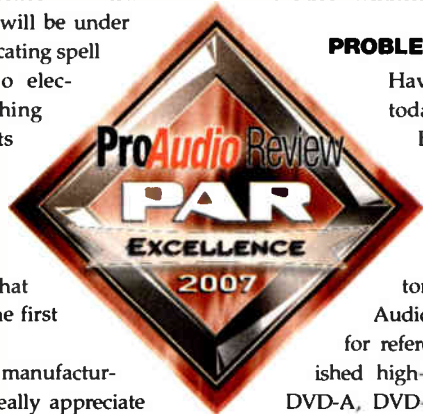
I recently took a look at the products that were awarded the *PAR Excellence Award* at the 1996 AES. Looking back, what is interesting to me is how the products reflected *PAR's* horizontal readership. There was gear, software/peripherals for studio, live, post, broadcast, installation and contracting.

For example, JBL's HLA touring speakers received an award, as did QSC's Powerlight 8.0 amplifier, in the live product category. For broadcast, there was the intuitive 360 Systems Shortcut editor, which is still easier to use than most workstation software. The contracting side was represented by such products as the Rane Balance Buddy and Audio-Technica's introduction of Uni-Tools. And Furman's why-didn't-we-think-of-that "Plug-Lock" power strip was also an easy pick. The high-end for film, broadcast and music applications was represented by such new products as the Euphonix CS3000 console and Pro Tools'

Version 4.0.

In 1996, there were state-of-the-art-product award winners that are, of course, no longer made today: Sony's PCMR500 lower-cost studio DAT, for example. And remember all those multitrack MiniDisc desktop machines that the big boys brought to market in the mid-1990s? Try finding one of those today.

The *PAR Excellence Awards* are essentially an innovation's barometer of what is happening in pro audio at the time, and when a product goes on to great success in the market, it kind of validates our little 'ol award. Look for the 2007 winners in the November issue.



PROBLEM SOLVED

Have you noticed that in today's DVD players (and Blu-ray and HD DVD, as well), all functions and setups are video menu-dependent? In my high resolution monitor rig, I use an Esoteric Audio DV-50 high-end player for reference listening of my finished high-resolution recordings on DVD-A, DVD-V, DualDisc and SACD.

Almost always, however, especially with DVD-As and DualDisc, you need to see the menu in order to select the right option for Group (which contains either the stereo high-res track, the PCM-multichannel high-res or Dolby Digital multichannel track). You don't know which Group contains the correct track — unless you have a video monitor connected.

I have come up with a way to view those critical video menu options without hooking up to a bigger LCD or CRT that I don't have room for in my rack.

I bought one of those portable DVD players, a \$120 closeout Craig seven-inch LCD. I run a composite RCA-to-1/8th-inch video cable from the Esoteric player to the video input of the portable DVD player and, viola, an instant compact monitor to allow me to select the required option from the on-screen menus. And, since it runs from a battery you don't have to hog up outlet space for the wall-wart — except to charge the battery once in awhile. What a smart guy I am.

John Gatski is publisher/founding editor of *Pro Audio Review*.

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Letters

ZOOMING IN ON PORTABLE RECORDER

Dear Sirs,

Thanks for the great article — "High Res/Low Cost Shootout" [PAR Aug. 2007].

I've had the Zoom H4 for about nine months and I absolutely love it! I have recorded my bands, brass ensembles, pipe organ concerts — even Brahms' *Requiem* with full chorus and orchestra. Most of these recordings were done surreptitiously, so placement was not always optimal. But the results have always been acceptable at the least and fabulous at the best!

I use rechargeable metal hydride batteries that have only run out in mid-recording due to my inattention to their charging or changing before hand. At full charge this has never been a problem, which leads to my one big complaint about the H4. It has no programming in the software to safely and adequately save the file being recorded

when the batteries die. I recently lost the whole set of my band's performance because of this. The unit does save the file, but incompletely, causing it to be corrupted. I've asked Zoom about this and am awaiting a reply. The owner's manual could be a bit more technically specific, too.

Other than that, I've found the H4 to be a wonderful location recorder. It is very forgiving of placement. I most often have no choice but to quickly choose a spot, set it down and hit record. Either the Neumann mic emulation or no emulation is my favorite choice. The auto gain function really works better than any other I've tried — no pumping. I keep the limiter engaged, which will pump when the input gets way too hot — but oh well.

All in all, no real complaints. Yes, it is to be treated gently, but shouldn't we be treating our favorite gear that way, anyway?

Thanks for the great magazine!

Scott E. Shuster
Oxford, Maine

The author replies: The few times my batteries have died the H4 correctly saved the file and powered down as it should. I haven't had any corrupt files in almost 100 recordings. I use a dedicated card reader for faster downloads to the computer.

Davis White

Feedback

We want to hear from you. Send your comments to letters@proaudioreview.com. Please include name, city, state and job title and firm in the email. For product submissions, contact Tony Ware at tware@imaspub.com.



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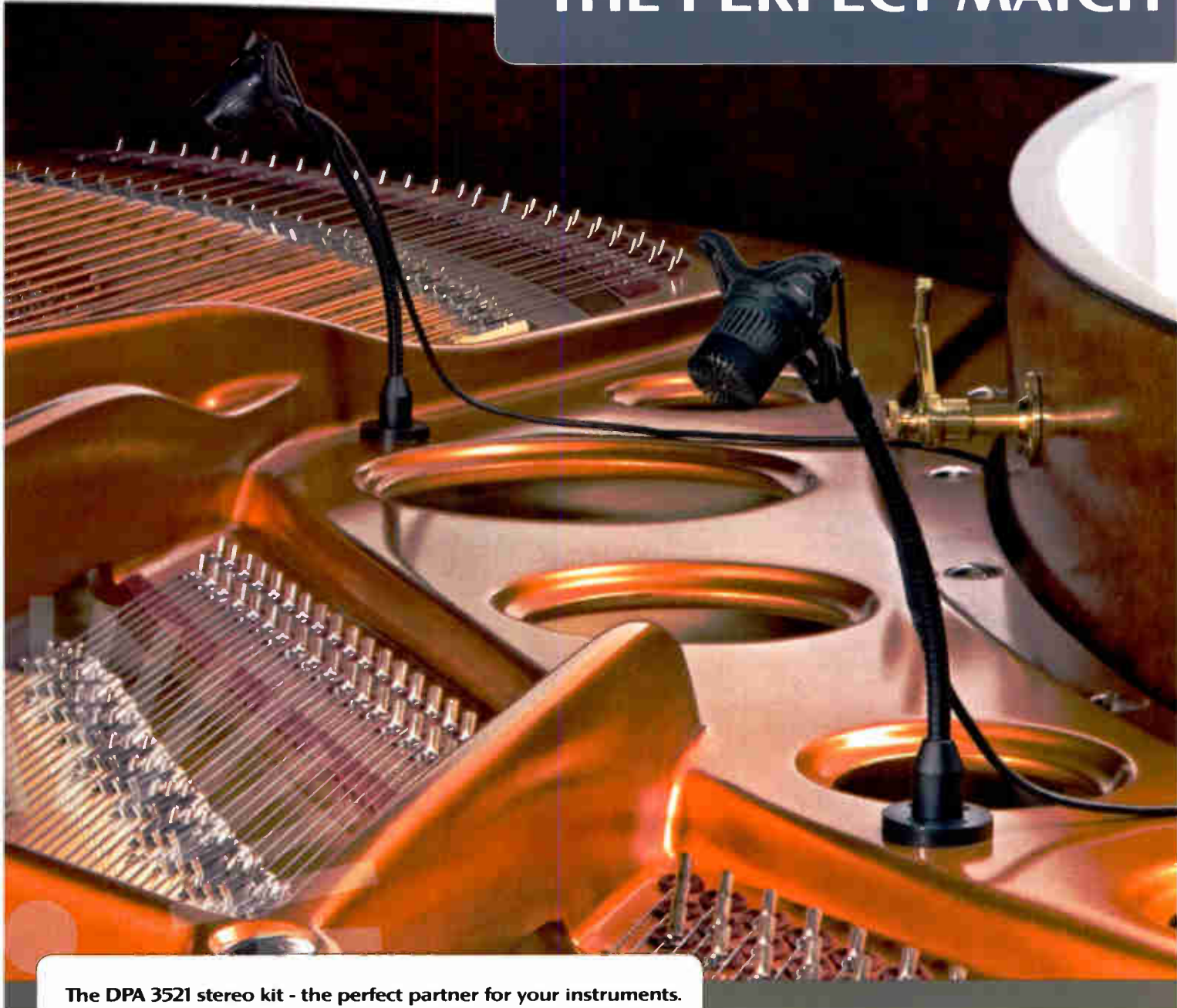
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Several seasoned professionals have recently found a new Reason to be creative. Artists including Arabian Prince, Mocean Worker, DJ Pierre and Aux 88 underwent the transition from **Propellerhead Software's Reason 3** to Reason 4 and proclaimed it "painless" and intuitive, with the new features such as the Thor polysic synthesizer module dubbed "innovative" and "phat."

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by Russ Long

Neumann KM D Digital Microphone

Neumann may have improved upon perfection with the new KM D series.

The concept of a digital microphone opens the door to a whole realm of possibilities ... as well as potential problems. Yet Neumann is the perfect candidate if someone is going to do it right. Neumann — a historic innovator in modern microphone development — is on my short list of companies that continue to excel in the microphone production arena.

The Solution-D project was Neumann's initial introduction into the small world of digital microphones. It resulted in the D-01, a large-diaphragm variable-pattern digital mic. The D-01 has the look of a traditional Neumann microphone, plus it provides a direct digital output. Now the technology from this mic has been applied to the new modular small-diaphragm KM D mic series (\$2,149 for a starter set of a KM 184 D with choice of AES/EBU or S/PDIF connection kit).

FAST FACTS

APPLICATIONS

Studio, project studio, broadcast, postproduction, ENG, film

KEY FEATURES

Comply with the AES 42 standard; KM D body with digital head-amp and output stage; KM 183 D omni, KM 184 D cardioid, and the KM 185 D hypercardioid interchangeable capsules; patented 28-bit A/D process; Neumann Remote Control Software (RCS)

PRICE

Starter Set — \$2,149; Single mic — \$1,949; Stereo set — \$3,769; Capsule head — \$779, DMI-2 \$1,365, Connection kit \$259.

CONTACT

Neumann | ☎ 860-434-9190
➤ www.neumann.com

The series includes three models that resemble the original analog KM 180 series, both physically and in model numbering. But, unlike their analog counterparts, the digital models are modular with interchangeable capsules from the KM 100 series. The KM D is available in classic nickel or black nextel finish.

FEATURES

The heart of this digital microphone system is the KM D body, which contains the digital head-amp and output stage. It can be combined with any of three interchangeable capsules to form the KM 183 D omni (diffuse field equalized), KM 184 D cardioid or the KM 185 D hypercardioid mics. Neumann has recently added three more capsules to the line — the KM 131 D free field omni, KM 143 D wide cardioid and KM 145 D cardioid with LF roll off for speech. IN 2008, a figure-8 KM 120 D will be added, as well as various extension tubes and other accessories.

Solution-D mics comply with the international AES 42 standard that has been extended from the AES 3 (AES/EBU) standard to include phantom power for the microphone, remote control and synchronization data, as well as user bits for the receipt of microphone control data. The mics feature an exceptionally clean, uncolored sound with very low self-noise and an extensive dynamic range. A patented 28-bit process performs the analog-to-digital conversion.

The mic supports sampling rates of 44.1, 48, 88.2, 96, 176.4 and 192 kHz, and has a frequency response of 20 Hz - 20 kHz. The Equivalent noise level (listed in this order: omni capsule/cardioid capsule/hypercardioid capsule) is 24/22/24 dB, and the equivalent A-weighted noise level is 13/13/15 dB. The signal-to-noise ratio is 70/72/70 dB, and the signal-to-noise ratio A-weighted is 81/81/79 dB. The Maximum SPL at 0 dBFS is 135/133/137 dB SPL. The dynamic range is 122/120/122 dB.

The mic can be used with the S/PDIF Connection Kit (which provides S/PDIF digital output), the AES/EBU Connection Kit (which provides AES/EBU digital output), or the DMI Digital Interface (which provides AES/EBU digital output).



The ability to use the Neumann Remote Control Software (RCS) is one of the primary strengths of the KM D Digital Mic. It requires the DMI-2 Digital Interface plus a PC (running Windows 98 SE, ME, 2000 or XP) or a Mac (running OS 8.6 or higher and CarbonLib version 1.6 or higher), a free USB port, 10 MB of free hard disk space, graphics resolution of 1024 x 768 or higher, HiColor or TrueColor, a CD-ROM drive, a mouse, and Adobe Acrobat

NEUMANN continues on page 14 ►

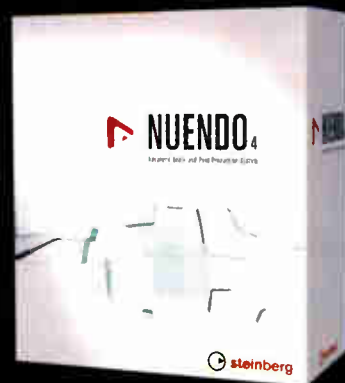
It's About Time

Anyone who works in post knows the constant pressure to get more work done under tighter and tighter deadlines. At Steinberg, we know that every extra key stroke and edit move adds up, costing both time and money. That's why Nuendo 4 has been specifically tailored to audio post professionals and gives you back something priceless: time to be creative. The ultra-smooth workflow built around a brand new routing and recording engine includes stunning new native processing tools specifically designed to streamline repetitive tasks and free you up to try new things. Nuendo 4 lets you work faster, be more creative and still deliver on schedule, no matter how crazy the deadline. It's about time.



“Nuendo has grown through a dedication to the post community, which is fantastic. The rethinking of the automation has been incredible, especially the Write-To-Punch feature, which is an extremely powerful tool for our type of work. Nuendo is absolutely world-class at this point.”

*John Ross,
Sound Supervisor and Re-Recording Mixer*



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Reader (to access the manual).

RCS is a universally applicable remote control software package for digital mics that operate in accordance with the AES 42 standard. The software displays all of the controllable functions and status indicators on the screen in easy to navigate channel strips that resemble a mixing console. The number of channels displayed is selectable from two to eight and is independent of the number of microphones controlled. All of the channel settings can be saved in configuration files for later retrieval. The current settings are automatically saved each time the software is closed, and are reloaded the next time it is started.

The software allows computer control of several parameters, including Low Cut filter, Pre-attenuation, Gain, Peak Limiting and Compression. If the Low Cut filter is activated, it can be set to 40 Hz, 80 Hz or 160 Hz. The Pre-attenuation is achieved by reducing the capsule voltage. If it is activated, the dynamic range is shifted by the corresponding value to higher sound pressure levels. The Gain parameter is carried out entirely in the digital domain, thus avoiding the additional noise and possible effects on the sound that can occur in analog processing. It is adjustable from 0 dB to +63 dB in 1 dB intervals.

The Peak limiter includes an adjustable threshold (-15 dBFS to 0 dBFS in 1 dB intervals) and a fixed attack time (-160 μ s) and release Time (< 0.1 s). The Peak Limiter provides very effective overload protection against peaks, which occur primarily as individual peaks. This facilitates easy, uncomplicated setting of the gain at the microphone, based on the effective level conditions. This is particularly helpful, for instance, when working without remote control only with the settings saved in the DMI. Even when switched off, the Peak Limiter remains active at the 0 dBFS threshold.

The Compressor/Limiter can function in a broad band, or as a high-frequency compressor/limiter (de-esser) in one of three selectable frequency ranges (> 1 kHz, > 2 kHz or > 4 kHz). The ratio can be set to 1.2:1, 1.5:1, 2:1, 4:1, 6:1, 8:1 or >100:1. The threshold is variable from -63 dBFS to 0 dBFS in 1 dB intervals. The attack time can be set to 0 ms, 0.1 ms, 0.3 ms, 1 ms, 3 ms, 10 ms, 30 ms, or 100 ms and the release time can be set to 0.05 s, 0.1 s, 0.2 s, 0.5 s, 1 s, 2 s, or 5 s. In addition, the sampling rate, the synchronization mode, test signals, the polarity of the output signal and the LED can be controlled remotely via the RCS. Internal

microphone software revisions can be updated via the Neumann DMI-2.

IN USE

I initially tested one of the mics with the AES/EBU connection kit. I put it to work recording a Taylor 514-CE acoustic guitar and it sounded wonderful. I tried all three capsules, and in each instance was impressed with the result. Although I don't own a set, I'm fairly familiar with the KM 100 series of microphones and the KM D sounds like a quieter version of the KM 100.

The most decisive factor of a digital mic is the A/D converter stage, and from listening to this microphone it appears that Neumann has nailed it.

The most decisive factor of a digital mic is the A/D converter stage, and from listening to this microphone it appears that Neumann has nailed it. This mic achieves a dynamic range beyond that of a normal capacitor capsule, yet it remains neutral, transparent and linear. The A/D converter is located directly next to the capsule, which provides the best A/D conversion available. The converter's dynamic range exceeds that of the capsule and it adds no coloration. Between the mic pre and the analog-to-digital conversion, an analog microphone's dynamic range is typically reduced by about 25 dB. Neumann's A/D in the Solution-D series is a true 28-bit conversion that provides the microphone's full dynamic range and is output as a 24-bit signal in the AES42 format. The DMI-2 interface controls the microphones and outputs the signal as standard AES3.

Using the mics along with the RCS and the DMI-2 interface is what I was really excited about, so I quickly made the move to installing the software on my MacBook Pro. The installation was quick, painless, and I was remotely controlling the mics in just a few minutes. The software has a great interface and is simple to use. The features are clearly labeled and overall it is very intuitive. I found it quick and easy to make level adjustments, add compression, adjust the pre-attenuation, insert the low cut filter, etc. It is even possible to remotely switch the power light on the microphone on and off, or change its brightness.

My hope is that mixer and computer interface manufacturers will quickly realize the importance of the AES42 interface and will begin to include it on their products, which in this case will eliminate the need for the DMI-2 interface.

During my review period, I had the opportunity to use the mics on drum overheads, piano, acoustic guitar and on a small choir. In every instance, I was absolutely thrilled with their sound. The only downside from my perspective is that I'm still not as fast with the RCS software as I am with traditional hardware, so at times I felt that I was dragging the session. This will change with more use and the software allows presets to be saved, thankfully; having a starting point that is relatively close to your desired settings is viable.

The integral digital limiter is another one of the system's strong points. It is nearly transparent yet it is 100-percent effective in preventing overloads. This is handy when doing typical tracking sessions, but indispensable when doing critical direct to stereo recording.

SUMMARY

The Neumann KM D Digital Microphone series is an amazing sounding microphone package providing unparalleled performance and amazing flexibility (especially if multiple capsules are purchased). At first glance the package appears to be a bit pricey, but when you take into consideration that the mics don't require mic preamps, analog compression or high quality converters, they are a bargain.

Russ Long, a Nashville-based producer/engineer, owns the Carport recording studio. He is a regular contributor to Pro Audio Review.

PRODUCTPOINTS

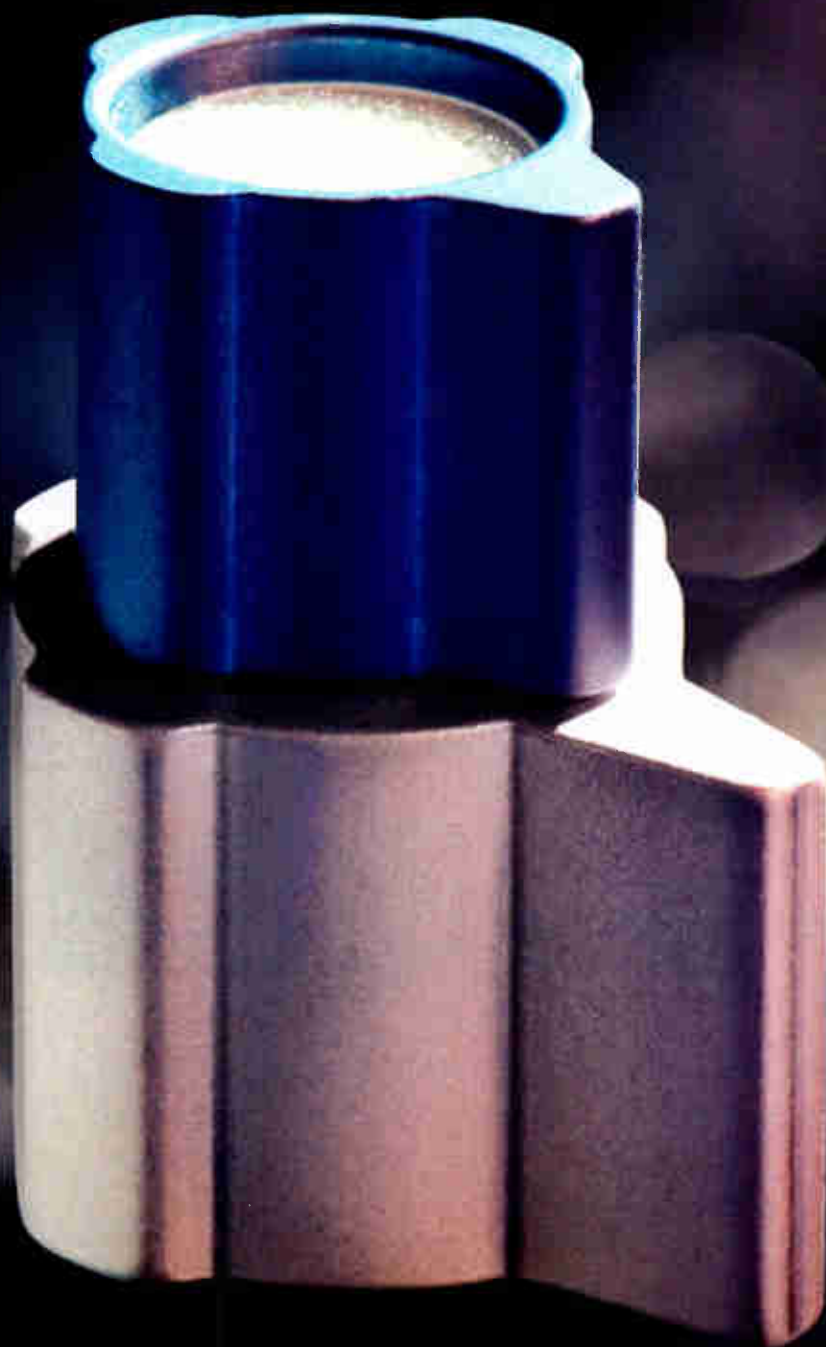
- Sounds like a quieter version of the KM-100
- Beyond a normal capacitor capsule range, yet neutral, transparent and linear
- Remote Control Software (RCS) and the DMI-2 interface

- RCS initially takes more time to control than traditional hardware

SCORE

The KM D Series is an amazing microphone package with unparalleled performance, amazing flexibility.

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AES booth 248

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World Radio History



by Stephen Murphy

Steinberg Sequel Music Studio

Even studio pros should consider this low-cost, feature-rich wunderkind.

According to the product's website, Steinberg's Sequel is "designed for first-time computer music enthusiasts," and "is the perfect first step into music production and performance" ... an "all-inclusive music production platform," to quote the manual.

In reality, those sweeping promotional descriptions are terribly misleading: Sequel turns out to be quite appropriate for seasoned professionals as well!

At a list price of \$129, this easy-on-the-curves (learning, that is), streamlined, loop-based mini-DAW is definitely poised to give a few bigger fish a run for the rapids.

FEATURES

This program and its user interface are so easy to use it seems against its nature for us to get bogged down on technical details. But before we can go out to play, we must first complete the assignment handed out by Professor PAR, so here's an overview of Sequel's specs ...

Both under the hood and in the driver's seat, Steinberg's Sequel draws heavily from the core technology within its popular Cubase DAW platform. The Sequel user interface bears more than a passing – albeit significantly hipper – resemblance to the latest version of Cubase (v4), and several of the new features from that program have been ported directly into Sequel.

Sequel's overall operation and single-screen GUI, however, are more closely related to popular loop-based programs like Sony Acid and Apple Garageband. Along those same lines, Sequel combines a clean, bars & beats grid timeline with straightforward drag-drop-and-drag (extend) event arranging, tempo auto-conforming, key transposing, MIDI sequencing and multitrack record-

ing. Included for use in the program is a large collection of loops and other audio content, sampled and synthesized virtual instrument sounds and built-in VST EQ, effects and other processing.



Sequel's main Arrange Zone

Sequel's recording/project file format is stereo WAV at 44.1 kHz (16 or 24 bits), and the program supports menu or drag & drop importing of WAV, AIFF, MP3, WMA, WMA Pro, OggVorbis and standard MIDI files (SMF), plus export to WAV, AIFF, WAV64, and OggVorbis formats.

As with Steinberg's other audio software offerings, Sequel is compatible with both Mac and PC systems. Although it could be considered a "basic" software application, the program requires a bit more hardware to meet minimum requirements than one might expect.

Here's a brief rundown (but be sure to check the Steinberg website for full specs and compatibility): Windows machines require XP (Home or Professional) or Vista Home Basic and above, an Intel Pentium or AMD Athlon 2 GHz processor and an ASIO compatible

interface (for low latency performance) though DirectX will work. Mac minimum requirements are OS X 10.4, a Power Mac G5 1.8 GHz or Core Solo 1.5 GHz and CoreAudio compatible audio hardware. Both platforms require at least 1 GB RAM, 6 GB of hard disk space, a Display resolution of 1280 x 1024 pixels, a DVD drive and an Internet connection (a must for license activation).

OK, enough of this yakity yak; there's plenty more specs in the manual, should you require them. Now it's time to play!

IN USE

Perhaps I was a bit too flip just then? Not serious enough for the hallowed pages of *Pro Audio Review*? Well, as it turns out, this is a perfect analog for Steinberg Sequel: sometimes it's good to not be overly focused on protocol and perfection; sometimes it's all about the process and the playing. Off we go ...

With a nod to Sony Acid and a wink to Apple Garageband (and a nudge-nudge to a couple others), Sequel leverages Steinberg's considerable audio and MIDI software acumen to boldly go where neither of the aforementioned apps have gone: a slick, super easy-to-use loop-based mini-DAW that spans the great divide with its cross-platform interface.

Apart from the usual protectionism regarding the DAW computer I use professionally, Sequel's standalone, all-in-one production platform was a perfect match for my new Intel dual-core (1.8 GHz x2) laptop with 2 GB of RAM running Vista Ultimate OS.

Once installed, authorized online and set to the appropriate audio interface, the program essentially led my way from there on out. Sequel successfully redirected me from my usual geek-quest survey of features and functions; its Media Bay content manager immediately beckoned for me to dive right into playing with the vast loop collections. Off to a good start, I'd say...

Speaking of content, Sequel comes with a generous (nearly 5 GB) collation of royalty-free audio content for use in your projects. The collation includes literally thousands of high-quality loops, one-shots and other goodies – all hierarchically meta-tagged across a wide range, with a drill-down progression from categories to subcategories to styles, substyles, and character. Consequently, navi-

STEINBERG continues on page 18 ►

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Nature has come up with clever ways to let some animals adapt quickly to their environments.

At Genelec our new 8200/7200 DSP Series also have the ability to adapt to their environment, by design.

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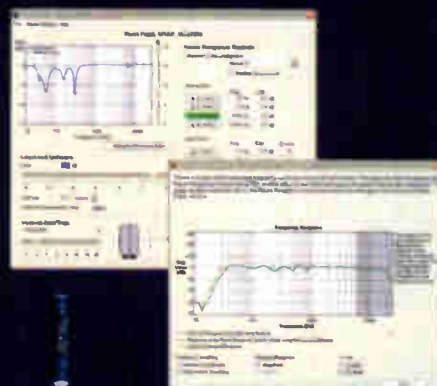
AutoCal™



GLM

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On screen, GLM software uses its *Interactive Response Editor* to give visual indication, loudspeaker-by-loudspeaker, of exactly what the response of each loudspeaker is.

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gating, searching and auditioning audio clips are completely intuitive and manageable. The program also comes with 600(!) MIDI-playable sounds from (and powered by) Steinberg's Halion One sample playback instrument. Sequel allows the user to audition instruments sounds and loops (in the correct tempo and key), all while the project is playing. Instrument tracks, by the way, are combined VSTi audio output (Hal One in this case) and MIDI data tracks that shield the user from the usual I/O and channel configuration rigmarole.

As mentioned earlier, Sequel uses a streamlined single-screen approach, with the Cubase-like Arrange Window... err, Zone occupying the upper two-thirds of the screen (as part of the hip-ification effort, all functional elements are now "Zones"). A tabbed six-zone section called the Multi Zone fills in the lower third, providing instant access to the Mixer, Track Inspector, Media Bay, Audio/Instrument Track Editor (context sensitive), Arranger (live performance) and Program Settings pages without losing sight of your project timeline or cluttering up the screen with open windows.

At the top of the screen is the Pilot Zone, a.k.a. transport bar, that also provides easy icon-and-tool tip access to the most-used operational functions (the sort usually relegated to dull menus and pop-up windows).

Besides Sequel's incredibly intuitive operation and well-appointed content and instrument collection, the most surprising discov-



Sequel's Media Bay

ery for me was that, despite its entry-level price, the program is not hamstrung by the usual entry-level restrictions. For instance: unlimited audio and instrument tracks; filter cutoff, resonance, drive, DCA attack/release, LFO modulation and effects control on sam-

At the top of the screen is the Pilot Zone, a.k.a. transport bar, that also provides easy icon-and-tool tip access to the most-used operational functions.

pled instruments; powerful (yet easy-to-use) automation of levels, processing and effects; automation track overlay and editing; Cubase/Nuendo's 32-bit internal mix engine; some really good effects and dynamic processing; Steinberg's powerful MIDI note and control editing; and Cubase/Nuendo's non-linear project arranging system (plus Ableton Live-style triggering from on-screen pads or computer keyboard); and many other common features usually stripped out in rudimentary and even LE versions.

If the \$129-list Sequel was exactly as billed – an entry-level program for first-time users, etc. – and didn't draw me in with its innovative and powerful features, I wouldn't feel that it's quite fair to raise some of these issues (nor would I be writing this review, for that matter), but it is so close to perfect for pro use, that I would really love to see a few improvements.

[Good news: As *PAR* went to press, my biggest issue was rectified, and hence removed from this article. The issue was the lack of external compatibility, specifically ReWire and Cubase/Nuendo import. The latter is now included in the Cubase 4.1 update and built in

to the newly released Nuendo 4. Crossing my fingers for ReWire and VST plug-in support, but those issues are not nearly as important anymore. – SM]

OK, I don't want to seem like I'm bullying a school kid, because I really do appreciate to

whom Sequel is being marketed and its unequivocal value, but for the love of Geffell, please allow the option to transfer the Sequel's Syncrosoft authorization to a USB dongle (like all other Steiny products) so I can take it with me or use it on my production DAW.

[Good news on this front: Though I couldn't get official confirmation, there are strong indications USB authorization may become a reality for Sequel's near future. OK...Steiny really took the wind out of my "issues" section – a really good sign for future Sequel development. – SM]

SUMMARY

As a longtime user of Steinberg's sequencers (and I mean Commodore-64 long-time), and a big fan of the company's Nuendo postproduction platform, I approached the prospect of reviewing "the perfect first step into music production and performance" Sequel software with what could best be described as an unbridled lack of enthusiasm. But once I climbed down from my Nuendo horse, gave the install DVD a spin and fired up the program, I discovered the wild fun that can be had with this fast and friendly mustang.

Sequel strips away most of the impediments to creativity imposed by full-scale DAWs and in many respects, beats the established loop-based programs at their own game. At an astounding street price of \$99, Sequel and its huge content collation are a super value, and a refreshingly fun experience to boot. I look forward to what Steinberg has in mind for Sequel's future (Sequel Sequel?) – I am especially keen to see ReWire support and the option for authorization to a USB dongle for those of us who work in multiple locations.

PAR Studio Editor Stephen Murphy has over 20 years production and engineering experience, including Grammy-winning and Gold/Platinum credits. His website is www.smurphco.com.

FAST FACTS

APPLICATIONS

Studio, project studio

KEY FEATURES

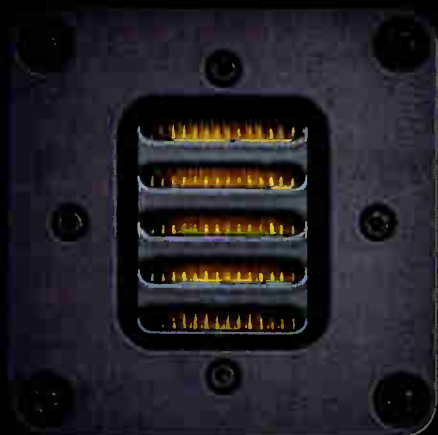
Easy-to-use loop-based music creation and live performance software; Mac/PC compatible; unlimited audio, midi and instrument tracks; 32-bit floating point internal processing; nearly 5GB of audio content included; streamlined single-page interface (with lower-third six-zone Multi-Zone interface).

PRICE

\$129

CONTACT

Steinberg | ☎ 877-253-3900
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by Russ Long

sE Titan Multi-Pattern FET Condenser Microphone

A titanium driver gives this versatile, affordable beast of a FET the flavor of a classic '47.

sE Electronics is one of the few companies that manage to make great sounding microphones at reasonable prices. From the beginning, sE founder Siwei Zou has been determined not to fall into the common practice of re-branding cheap Chinese microphones. Unlike most Chinese microphone companies, sE Electronics has its own factory and manufactures only its own products; the sE mic capsules are built entirely by hand.

Since launching its condenser mic line in Europe in 2003, sE has become the number one microphone brand in the UK and the fastest growing brand in all of Europe. I reviewed the sE R1 ribbon mic a couple of years ago and found it to be a respectable mic worthy of attention, so I was happy to put the Titan to the test and see how I felt about sE's condensers.

FEATURES

The sE Titan is a beautifully designed, multi-pattern, transformerless, Class A FET condenser microphone (According to the manu-

facturer, the transformerless design enhances low end and assures no low frequency distortion.

— Ed.). The \$1,499 mic (special discounts forecoming) uses a center-terminated titanium diaphragm that does a wonderful job of capturing clarity, detail and transient response. Like gold, titanium doesn't tarnish. It is a lighter and stiffer metal than gold, so it is able to move faster and provide a superior transient response. This translates into clear high frequencies with less distortion and a tighter low frequency response. The Titan is equipped with a -10 dB pad and a low frequency roll off, which are both activated by a switch on the mic. The mic includes a shock-mount, a wooden storage box, an XLR cable and an aluminum flight case.

The Titan has a frequency response of 20 Hz - 20 kHz and a sensitivity of 40 mV/Pa - 34 ± 2dB (0dB=1V/Pa 1000Hz). The mic's selectable polar patterns include cardioid, omni and figure-eight. The microphone's impedance ≤ 200 ohms and the equivalent noise level is 16 dB (A weighted). Max SPL for 0.5-percent THD@1000 Hz is 128 dB. The mic uses a standard 3-pin XLR connector and it requires 48 volts phantom power for operation.

IN USE

I have had the opportunity to use the Titan in a wide variety of circumstances and have been very pleased with the results. Although the mic arguably has a sound of its own, I found it to somewhat resemble the Neumann FET 47 in its overall performance characteristics. The mic can handle a high SPL and it works exceptionally well on per-

cussion, especially on kick drum.

During a tracking session, I placed an AKG D112 inside the kick drum slightly off center but pointed at the beater. I placed the Titan outside the kick drum about three inches from the back head. I blended the two mics about 50/50 and ended up with a wonderful kick sound. I usually use a Yamaha SubKick in addition to the two mics and, surprisingly, I was able to get enough bottom end from the Titan where this wasn't necessary. I had good results using the mic on the bottom of a djembe, and it also worked well on an 18-inch floor tom.

As a vocal microphone, the Titan works well in most situations, though overall I tended to have better results using it with male vocals than with female vocals. I found that using minimal or no equalization in most circumstances resulted in a full and rich vocal sound.

The microphone has very little proximity effect making it a great contender for the broadcast market.

I had the opportunity to use the Titan to record viola and cello, and in both instances had good results. The

mic tended to get a bit harsh when it was closer than 24 inches, but as long as it was 2 - 3 feet away from the sound source it sounded smooth and present without getting brittle or edgy. I also had good results using the Titan to record bass guitar (on an SVT cabinet), plus acoustic and electric guitars, and in each of these instances had nice results.

SUMMARY

The SE Titan is a versatile microphone that isn't necessarily a bargain, but is easily worth its price tag. The mic has a smooth and natural sound with nice top end sheen.

Russ Long, a Nashville-based producer/engineer, owns the Carport recording studio. He is a regular contributor to Pro Audio Review.

REVIEW SETUP

Apple Macintosh 2 GHz Dual Processor G5 w/2 GB RAM; Digidesign ProTools 7.3; Lynx Aurora Converters; Lucid Gen-X-96 Clock; PMC AML-1 monitor



FAST FACTS

APPLICATIONS

Studio, project studio, broadcast, postproduction

KEY FEATURES

Titanium diaphragm; transformerless; -10 dB pad and 100 Hz roll off switches; cardioid, omni & figure-eight polar patterns

PRICE

\$1,499

CONTACT

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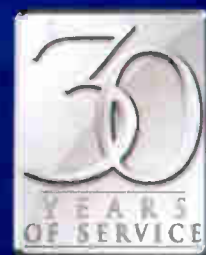
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by Strother Bullins

KRK Systems VXT8 Powered Studio Monitor

The VXT8 offers KRK excitement of old to a new generation of powered monitor buyers.

BENCH TEST | Page 25

Back in January of this year, a colleague and I visited the demo room of KRK Systems at the Winter NAMM Convention in Anaheim, CA. Within, I was thrilled to discover that KRK was displaying betas of the company's latest moni-

Of course, this news piqued my interest, as well as encouraged my doubts; was it true that for \$1,200 a pair street I could have a pair of monitors that sounded even close to the Exposé 5.1 setup that Chuck Ainlay used to bring to Nashville's Masterfonics' the Tracking Room (the studio where I worked back in the day)? If so, I was excited to try them out. After waiting patiently for months, then listening to a VXT8 pair for another couple of months, I was convinced: KRK is back.

As I observed while peering into Nashville pro engineer circles in and around 1998, using powered closefields then usually meant Genelecs, or possibly the then-new and budget-friendly Mackie HR824. Later, innovative folks like Ainlay, amongst others, started to roll up with extra flight cases holding two or more Exposé E8 monitors.

To my ears, the KRK E8s were markedly different; they weren't sweet and zippy as I personally felt Genelecs were, just incredibly natural, organic and powerful. Certainly the VXT8 isn't the same monitor as the Exposé is (or was), but it does hold those same qualities that had me coveting the E8s of others.

| FEATURES

The dual-amped KRK VXT Series includes three models (essentially something for every powered closefield monitor shopper): the VXT4, VXT6 and the aforementioned VXT8, with list prices of \$399, \$599 and \$799, respectively. The number in each model name refers

to woofer size; the series offers four-inch, six-inch and eight-inch Kevlar woofers, each in classic KRK yellow. Each model features a one-inch silk dome tweeter (in comparison, the new-design KRK Exposé E8B comes with an aluminum/beryllium composite inverted dome tweeter). Main features of the new series include active filter crossovers (subsonic, low

Most interesting is its sleek "Resonant Free Extended Low Frequency Enclosure Design," built from ABS Structural Foam, which is touted to be far stronger than cabinets made of common ABS solid materials.

pass and high pass), a defeatable limiter, a new design with radiused faceplate for reduced diffraction (*According to the manufacturer, the greatest benefits are enhanced imaging and a huge sweet spot.* — Ed.), and more. The VXT Series is manufactured in China.

VXT8 technical specifications are as follows: Reported frequency response is 37 Hz – 22 kHz +/-1.5dB. Maximum SPL is 111 dB music, 114 dB peak. Dual power comes from 60 W high-frequency and 120 W low-frequency amplifiers. (For more VXT8 tech specs, see Tom Nounsaine's full bench test following this review.)

KRK continues on page 24 ►

FAST FACTS

APPLICATIONS

Studio, project studio, broadcast, post

KEY FEATURES

60 W and 120 W dual amps; eight-inch Kevlar woofers; one-inch silk dome tweeter; filter crossovers; defeatable limiter; Neutrik Combo input connector; Resonant Free Extended Low Frequency Enclosure Design

PRICE

\$799 each

CONTACT

KRK Systems | ☎ 954-316-1580
🌐 www.krksys.com



tor range, the VXT Series. At first glance, the VXT Series looked a lot like the Exposé Series; as it turned out, the VXT Series is intended to possess many of the aural attributes of the Exposé Series, which so many of us fell in love with during the late '90s, but at a far more affordable price point.

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ALLEN & HEATH

World Radio History

On the VXT8's face, the woofer resides just above a rectangular slotted port, which, according to KRK, reduces "port turbulence and distortion commonly found in poorly designed round ports." I've never been much of a fan of

front panel round ports, so this was an appreciated feature. Above the woofer is the tweeter, and to its right is the KRK logo, which glows warmly when the VXT8 is on. A "clip limit" LED resides just below the logo.

On the rear, a Neutrik Combo connector accepts XLR, 1/4-inch TRS and TS inputs; it is accompanied by a variety of useful controls. System Level Adjust allows counterclockwise-reduced attenuation from +6 dB to -30 dB. The factory preset gain is +6 dB. A high-frequency adjustment toggle allows +1 dB HF Shelf, Flat and -1 dB Shelf settings. A low-frequency adjustment toggle allows whole (normal), quarter-space and half-space roll-off settings. Also available are auto-mute (on/off), clip indicator (on/off/limit) for KRK's improved speaker protection circuit and ground lift (LFT is on, GND is off) toggles. Two well-conceived translucent plastic shields protect the VXT8's various metal toggles.

VXT Series' build quality, heft and design are notable features unto themselves. Most interesting is its sleek "Resonant Free Extended

Low Frequency Enclosure Design," built from ABS Structural Foam, which is touted to be far stronger than cabinets made of common ABS solid materials (and I believe it; a VXT8 physi-

PRODUCTPOINTS

- Accurate
- Great imaging
- Good value
- Unmistakable, classic KRK pedigree

- None noted

SCORE

The VXT8 is an accurate, organic-sounding, powerful and reasonably affordable bi-powered studio monitor.

My familiar audition tracks sounded as they should have, and each seemed to be more natural and, dare I say it, more "analog" (I challenge anyone to grab a pair of VXT8s, put on Tom Petty's *Wildflowers*, and resist the urge to smile goofily before the first track is over).

cally feels far pricier and substantial than its \$599 street price implies). (According to the manufacturer, the ABS Structural Foam allows for thinner yet stronger walls, providing more internal volume and bass extension, plus no parallel surfaces internally help reduce standing waves. — Ed.) The heft-as-weight of the VXT8 is considerable —

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BENCH TEST

KRK VXT8 Studio Monitor

BENCH MEASUREMENT DATA

Frequency Response:

On-axis 66 Hz to 20 kHz +/- 2.8 dB

Bass Limit:

72 dB SPL @ 40 Hz @ 2 meters
(<10% Distortion)

Control Action:

HF Adjust

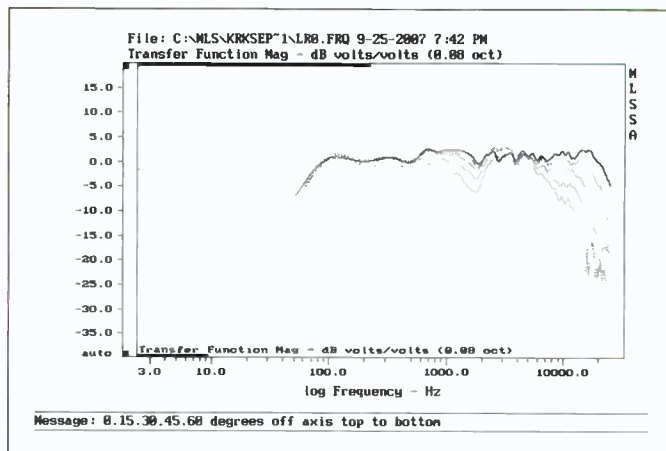
HF +1 Actual Response +1.1 dB >2 kHz

HF -1 Actual Response -1.1 dB >2 kHz

LF Adjust (relative to Whole Space)

Half Space Actual Response -1.4 dB < 200 Hz

Quarter Space Actual Response -2.1 dB < 200 Hz



BENCH MEASUREMENT COMMENTARY

The Bass Limit of the speaker is the Sound Pressure generated at 2 meters in a 7600 cubic foot room with less than 10 percent distortion. The 10 percent distortion limit is used because operating characteristics of drivers (using DLC Design DUMAX) shows that when a speaker has reached the end of its linear operating range (BL product has fallen to 70 percent of the rest position value or the suspension compliance has stiffened by a factor of 4) the unit will still sound clean, but distortion increases exponentially with further drive. However port and suspension noise and with powered speakers amplifier output or limiting may also constrain sound pressure capability.

Basic measurements here have been taken at a full 2 meters in

a large room on a 6 foot stand. Using time windows then gives equivalent anechoic results above 200 Hz including front panel reflections and cabinet diffraction; overall results give true acoustical summation of all drivers and passive radiating elements.

The KRK VXT8 has remarkably smooth response on axis marred only by a 2.3 dB elevation between 600 and 1800 Hz and a 1.6 dB rise above 3 kHz. However the HF Adjust of -1 nearly perfectly accommodates the latter anomaly. There is relatively moderate off-axis lobing beginning just above 1000 Hz when the microphone is moved outward in the horizontal plane to 30 degrees and beyond. The speaker low frequency dynamic capability is limited by port noise at 40 and 50 Hz.

— Tom Nousaine

41 pounds — but nowhere near the weight of the current E8T, which is over 75 pounds (and for those of us without the benefit of a cartage services, over 40 pounds per monitor is more than enough heft).

| IN USE

I listened to the VXT8 for a number of weeks in a fairly large control room and in a smaller, damped edit space. Source material varied widely; I mixed, "treated" and referenced completely raw 24-bit/96-kHz tracks recorded via an Alesis HD24XR multitrack, and I listened to a number of my own mixes and normal collection of personal references.

After a few days of break-in time, I sat down for some truly critical listening and found what I hoped (and, after the last few days of not-so-critical listening, expected) would be there: smooth, natural highs reminiscent of the E8s, as well as a full and room-filling, yet tight and accurate bass wallop ... the latter being exactly what I had forgotten about the E8s due to my preoccupation with its

uniquely nice top-end. Vocals were natural with smooth transients and rarely.

My familiar audition tracks sounded as they should have, and each seemed to be more natural and, dare I say it, more "analog" (I challenge anyone to grab a pair of VXT8s, put on Tom Petty's *Wildflowers*, and resist the urge to smile goofily before the first track is over).

To my ears, imaging was generally excellent, thus mixing was easier than normal while using the VXT8s. I even liked what I did more a few weeks later while listening back to the VXT8 mixes on a couple of other go-to speaker references. In my opinion, a mix made on a VXT8 pair will translate to other monitoring systems incredibly well.

Finally, the VXT8s possessed that certain "wow" factor most of us desire; they sound big, have a large, wide sweet spot and are capable critical tracking, treatment and mix monitors, as well as perfectly affordable "knock your socks off" mid-fields for impressionable clients. In my experience, a large control room with hardwood floors was no match

for the VXT8s; closefields of similar size and price are regularly underwhelming in this environment, but even when I drove the VXT8s rather hard at times they always stayed clean and full without becoming sloppy or shrill.

| SUMMARY

The KRK VXT8 should be a notable contender in the mid-priced, close- to mid-field powered monitor market. It's far more affordable than KRK's premier E8T (or the new model the E8B), yet it looks, sounds and performs like a product built by the same people that brought us the Exposé E8 (which, arguably, many KRK fans still prefer to new E8 models with its next-generation tweeters). In my evaluation, the VXT8 performance was always accurate, natural and powerful. But don't take my word for it; check out a pair and hear for yourself.

Strother Bullins is the Features and Reviews Editor for Pro Audio Review.

by John Gatski

M-Audio EX66 Powered Closefield Monitor

This mid-priced digital/analog closefield monitor accepts up to 216 kHz input for re-clocked 96 kHz playback.

BENCH TEST | Page 28

M-Audio has been making powered monitors for several years, focusing on inexpensive closefield models that sounded pretty good for not much money. The EX66 is a large step up for Digidesign's overachieving little brother, and is equipped with three drivers, a high-power onboard amp, digital processing and digital input.

| FEATURES

Priced at \$1,399 per pair retail, the designed-in-US/made-in-China EX66 sports two six-inch woofers and a one-inch titanium dome tweeter. The well-braced MDF cabinet utilizes a rear port for bass extension. Specs include a claimed 37 Hz - 22 kHz frequency response (50 Hz - 19 kHz +/- 1 db). The EX66 features two 104-watt (dynamic power) Class

D amplifier sections: one for the bass, one for the treble. The two-way driver arrangement contains a 2.56 kHz 4th order Linkwitz-Riley crossover. All signals — analog or digital — are converted to digital.

The clock accepts any sampling frequency up to 216 kHz and re-clocks it at 96 kHz. The tone-tailoring controls are DSP-based and include high-frequency adjustment, low-frequency cutoff, acoustic space bass compensation (full, half and quarter) and midrange boost.

Inputs and outputs include AES/EBU input and S/PDIF input for the digital, and 1/4-inch input balanced and XLR balanced for the analog. There are also digital pass-thru jacks to feed separate left and right signals. The cabinet measures 19-inches tall by 8.25-inches wide and 9.5-inches deep. Weight is a not-too-heavy 25 pounds each. The drivers are shielded to prevent video screen coloration.

| IN USE

I set up the EX66s in my studio, mounting them on custom Apollo stands and moving them about three feet from the wall and about six feet apart, angled in slightly for better dispersion to the listener's position. I fed analog audio to the speakers via a Legacy/Coda high-current monitor preamp and a Trident 8T console I had on hand for another review. Digital audio — 96 kHz and 192 kHz — was relayed from a Lynx L22 PCI card/Mac G5 workstation.

All audio was routed via Alpha Core solid silver XLR or Westlake Low PE XLR cables. Audio included 192 kHz and 96 kHz PCM acoustic guitar mixes, several reference SACDs and DVD-As, as well as some bluegrass music done at 44.1 kHz. I also listened

to a house dance recording that I remastered a couple of years ago.

Other speakers I had on hand to compare with the EX66s included Legacy Studios and Lipinski L-505s — both passive speakers, but about the same size. The Lipinski L-505s are about as accurate as *PAR* has ever measured, and are good speakers to compare with closefield monitors. The passive speakers were driven by amps ranging from a high-end Pass Labs X-350.5 to a \$150 Behringer amp.

I left the speaker tone controls in the flat position, though I did play with the acoustic



space control and found it to work best in the half-space mode for my room.

After a week's worth of break-in time, I sat down and listened to the EX66. Upon first listening to some reference jazz SACDs, I thought the EX66s were quite articulate on the top end. I could clearly hear room reverb tails from drum cymbals and the speakers did a good job of reproducing instrument timbre.

The bass was excellent — with depth and tightness, but without mid-bass boom I have heard in other small, ported speakers.

M-AUDIO continues on page 28 ➤

FAST FACTS

APPLICATIONS

Studio, project studio, broadcast, postproduction

KEY FEATURES

216 kHz digital input for re-clocked 96k playback; two six-inch woofers and a one-inch titanium dome tweeter; a rear port; two 104-watt Class D amplifier sections

PRICE

\$699 each

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World Radio History

Acoustic guitar recordings — especially an Audix SCX-25-miked Martin custom 00-28 — sounded really good. Imaging was wide, with good detail filled in between the speakers.

In comparison to the Lipinski and Legacy passive speakers, I heard many of the musical nuances with the EX66s — especially in the transients. The M-Audio has more low bass than the Lipinskis.

The EX66s delivered lower-to-moderate SPL pretty cleanly, but lost clarity when really

cranked up. The highly modulated recordings with horns or keyboard music got a bit congested sounding; I thought they sounded better loud with acoustic music and classical than busy pop and jazz music.

Ergonomically, the speaker controls and settings were easy to use. And, unlike its pricier Digidesign cousins I reviewed a few issues ago, the EX66s can run 192 kHz signals via the digital input and hear the audio (albeit down-sampled).

SUMMARY

At \$1,199 per pair on the street, the EX66 is priced in a very competitive niche for a powered studio monitor, and I think M-Audio has done a good job providing a full-featured speaker with good performance. It's not a speaker that sounds its best really loud, but it does a reasonable job reproducing accurate audio at low-to-medium levels and delivering precision in the stereo image.

John Gatski is publisher/founding editor of Pro Audio Review.

BENCH TEST

M-Audio EX66 Monitor

BENCH MEASUREMENT DATA

Frequency Response:

On-axis 67 Hz to 19.1 kHz +/- 3.0 dB

Bass Limit:

75 dB SPL @ 50 Hz @ 2 meters (<10% Distortion)

Control Action:

Low Cutoff:

37 Hz Actual Frequency 69 Hz

80 Hz Actual Frequency 82 Hz

100 Hz Actual Frequency 94 Hz

Midrange Boost:

+ 2dB; Actual Boost 1.5 dB 400 Hz to 7.5 kHz

High Frequency Adjustments:

+2 dB; Actual Response +2 dB > 2 kHz

-2 dB; Actual Response -2 dB > 2 kHz

Acoustic Space: (Compared to Full Space)

Quarter Space: -3.7 dB < 500 Hz

Half Space: -3.0 dB < 500 Hz

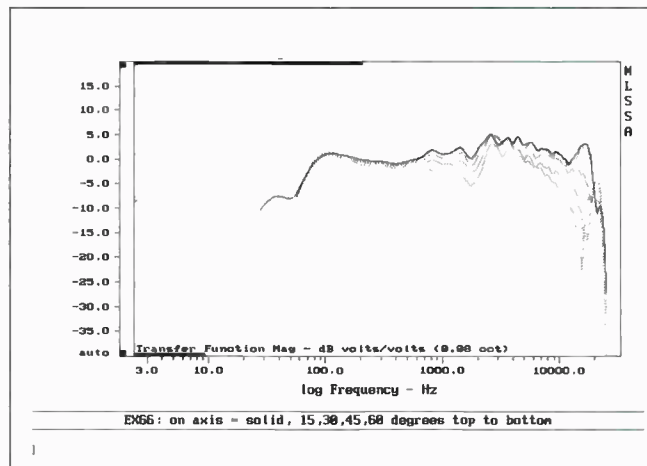
BENCH MEASUREMENT COMMENTARY

The Bass Limit of the speaker is the Sound Pressure generated at 2 meters in an acoustically large (7600 cubic foot room.) The figure of merit 10% distortion is used because operating characteristics of drivers (using DLC Design DUMAX) shows that when a speaker has reached the end of its linear operating range (BL product has fallen to 70% of the rest position value or the suspension compliance has stiffened by a factor of 4) the unit will still sound clean, but distortion increases exponentially with further drive. With powered speakers amplifier output or limiting may also limit sound pressure capability.

Basic measurements here have been taken at a full 2 meters in a large room on a 6 foot stand. Using time windows then gives equivalent anechoic results above 200 Hz including front panel reflections, cabinet diffraction and true acoustical summation of all drivers and passive radiating elements. Control action was measured using the same microphone set-up.

Of the original pair of EX66s I received one of them had a

response anomaly at 8 kHz that clearly showed the speaker was defective. Potential customers should make certain that return privilege is assured. Control action of everything except Low Cutoff matches specification closely. In the lower operating range of the speaker it can be seen that the port only acoustically adds response well below the true operating range of the upper frequency drivers at reasonable listening distance of 2 meters and 90 dB SPL. This is also true of much of the competition and loudspeak-



er specifications are frequently made under conditions that do not emulate actual usage conditions. This is why Pro Audio Review measurements are made at a full 2 meters, a reasonable listening distance, which allows true acoustical summation of all radiating elements in a real environment. The EX66 has a 4-5 dB elevation between 2 and 6 kHz that forms the basic sonic character of the system. Directivity is well controlled and the system will sound similar at all listening positions. Dynamic capability is relatively limited. The EX66 can produce 75 dB SPL at 50 Hz and 2 meters but at that sound level the system is quite sensitive to system noise with small increases in drive level. Be sure to use a woofer system with wide band low frequency program material.

— Tom Nounsaine

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by Rob Tavaglione

FMR Audio RNLA7239 Levelling Amplifier

Yes, the RNLA offers a really nice levelling amp at a really cheap price.

What if I told you of a levelling amp that claimed to be "really nice," was only a third of a rack-space wide and sold for \$250? You'd be skeptical, right...but check this out.

FMR Audio, manufacturer of the Really Nice Levelling Amplifier, builds the heralded RNC (Really Nice Compressor), known for its quality processing, clean transparency and bargain price. This review will address whether the RNLA, like the RNC, could be another FMR product destined to become a "best kept secret" for many budget conscious engineers.

FEATURES

Visually, the RNLA is of diminutive size and features striking orange knobs. The front panel, nearly identical to the RNC, has only one set of Threshold, Ratio, Attack, Release and Gain controls. The RNLA borrows from the Valley People Gain Brain II with an additional control: Log Release Contour. This helps by speeding up the release curve acceleration

under extreme attenuation. The RNLA is a stereo-only device; the two channels can not be unlinked or separately controlled.

Although sturdily built using extruded aluminum and steel, the RNLA saves money and space with an external (9 volt AC) wall-wart power supply. Potentiometer quality is not an issue here, as the front panel is only a control surface for the digital engine inside; this DSP in the sidechain path models an opto element and then feeds a Blackmer VCA. Specs are worthy with only 0.005-percent THD, a maximum output of 22.5 dBu (0.3-percent THD) and noise is typically -95 dBu over full bandwidth.

IN USE

Right on the money, the manual recommends applications with vocals, bass guitar and drums. I ran both my bass DI and mic outputs through and was quickly impressed. Smoother dynamics nicely brought all the notes closer together in volume, tempered occasional harsh note attack and added just a touch of low-end emphasis. This coloration is subtle, pleasant and very mildly distorted, much like some vintage units, but not exactly like any one in particular.

Drums, predictably, worked well with the box. I tried processing individual track outputs, but the RNLA's stereo design and inherent sound made me prefer the whole kit nicely leveled instead (via subgroup buss inserts). The circuitry made cymbals heftier, less piercing and gave a nice low-mid punch to snare and toms that EQ cannot. The Log Release shines in this application, allowing more gain reduction with seemingly more responsiveness, less distortion and smoother attenuation recovery. I was compelled to try the "soup du jour" — slamming drum room mics — and got fickle results. After much tweaking I settled on approximately 3:1, with the Log Release, a low threshold, slow attack and quite slow release (leveling, one might say). However, these settings were extremely sensitive and were the slightest twist away from undesirable pumping or mush ... so watch that release. The interac-

tion between controls does provide much fodder for experimentation and unusual dynamic envelopes, if so desired.

Vocals are handled nicely by the RNLA, imparting all of the above mentioned characteristics, which are often desirable, especially



with the common harshness of condenser mics. I really liked the RNLA on female vox, where dynamics were easily controlled with the added taming of sibilance and the chesty "filling up" of the low-mids around 200 Hz — a response that reminded of me of a dialed-in multiband compressor.

I found processing of melodic instruments to be sonically significant, but not always preferable. Electronic sources, such as synths and samples, often gained size and solidness, while guitars either got a bit too undefined and thick or nicely beefed up (depending on the nature of the source). Most acoustic instruments seemed to like the ride, particularly if they were scratchy or clicky up top. Use on the mix buss revealed too much coloration, inadequate top-end definition and too "soft" of a sound ... at least with my analog mixing desk. Those who prefer to mix in-the-box with an analog compressor inserted on the mix buss might find those qualities to be perfect for the job. The RNLA's true bypass feature, with a sealed relay, proved to be helpful in such comparisons.

SUMMARY

I found this compressor to be an incredible value with an adequate build, a thoughtful design, useful features and that ever-elusive quality: a "sound." My gripes are few and relatively minor; I could live without the wall-wart, balanced inputs would be nice (even though we'd lose the "insert" ability of the input) and a more thorough legend with attack and release times would be quite helpful. But at this price point I'm not complaining, especially for a "Made in the USA" product. (For the record, I'd pay twice as much for a "pro" version with XLR I/O, an internal power transformer, more metering and two sets of controls.)

It looks like the RNLA really is really nice after all.

Rob Tavaglione owns Catalyst Recording in Charlotte, NC, and mixes for the Charlotte Bobcats. Contact him at rob@catalystrecording.com.

FAST FACTS

APPLICATIONS

Studio, project studio, live, broadcast

KEY FEATURES

Threshold, Ratio, Attack, Release and Gain controls; true stereo operation; eight-segment gain reduction LED meter; I/O on 1/4-inch connectors with an unbalanced input and a non-differential balanced output on TRS; TRS doubles as inserts; a single TRS cable can connect the RNLA to an unbalanced insert point on your mixer/preamp. A sidechain I/O connection is also provided on another quarter-inch TRS; Blackmer VCA

PRICE

\$249

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UpSAMPLER

Capsule Reviews and Product Review Updates

Propellerheads Abbey Road Keyboards ReFill

Contact: Propellerheads | Price: \$229 |
818.575.3600 | www.propellerheads.se

As a huge fan of the Beatles and Pink Floyd, I'm naturally a fan of Abbey Road Studios; so many performances by those bands (as well as the recordings of hundreds of other groups) were documented in musical history within that facility. Additionally, though I'm not an expert user by any means, I've been a huge fan of putting Propellerhead's Reason virtual studio rack to work as often as possible over the last few years (usually within a Pro Tools session). Upon hearing that Propellerhead was putting together my two loves in the Abbey Road Keyboards ReFill for Reason, I couldn't wait to make the actual sounds of the studio's keyboard collection a part of my recordings.

This collection was recorded on location in Abbey Road – using vintage microphones, equipment and recording rooms – and provides Reason access to seven of the studio's classically captured and processing-ready instruments: the Steinway "Mrs. Mills" upright, the Challen Studio Piano, the Hammond RT-3, the Mannborg Harmonium, the Schiedmayer Celeste, the Mellotron M400 and the Tubular Bells (which are tuned and laid out like a keyboard instrument). Minimum system requirements of the ReFill for Windows are Reason 3.0.4 or later, an Intel P4 1.4 GHz/AMD Athlon XP, 1 GB RAM (2 GB recommended) and DVD drive. The recommended system includes an Intel P4 2.4 GHz processor, 2 GB RAM and a low-latency ASIO-compatible soundcard.

Minimum Mac OS requirements are Reason 3.0.4 or later, Macintosh G4 1.0 GHz, 1 GB RAM and DVD drive. The recommended system includes a Macintosh Intel, G5 or Dual G4 1.4 GHz processor, and 2 GB RAM.

The package includes two DVDs: one 16-bit, one 24-bit. Except for the bit depth, the DVDs contents are identical. This makes the two versions completely interchangeable; however, they cannot coexist in the same location. To alternate between the two, place the currently unused version outside your computer's designated ReFill location. It is possible to mix bit rates by having Abbey Road Samples 1 -16.rfl and Abbey Road Samples 2 -24.rfl side-by-side, since they have different content.

To determine which ReFill version to install, it is important to consider if the computer is powerful enough (CPU speed, system bandwidth, RAM, audio card, etc.) to handle the 24-bit format, if there should be any concern about workflow issues such as loading and browsing time, if there is any practical and/or creative need for the 24-bit format, and if the extra disk space needed for the 24-bit version (4.2 GB instead of 1.7 GB) is available.

Each instrument in the collection utilizes Propellerhead's Hypersampling methodology, making the product extremely flexible.

There are samples covering between four and five velocity levels of each note sampled, and typically there are samples of every third note across the keyboard. The keyboard samples were captured by up to 12 different microphones. This provides the user with complete control over all sound aspects relating to microphone placement and signal mix. Abbey Road Keyboards incorporates attributes such as hammer noise and release resonance samples, thus mimicking real instruments all the way from keystroke through key release. This ReFill also includes preset Combinator patches for various mic blends, style patches, and empty, pre-wired template patches.

Pull up the Mellotron flutes and it's like you are sitting in on the "Strawberry Fields Forever" tracking session. The sounds themselves are amazingly inspiring. While there are only seven instruments, each are immaculate and extremely usable.

The thing I really love most about this collection is the realism of the sounds. I work primarily with bands and real instruments, and a lot of sample collections out there are too clean and slick to be usable. The Abbey Road package is the best ReFill that I've encountered. If you use Reason, it's a no-brainer: buy it!

—Russ Long

Audio-Technica ATH-ANC7 QuietPoint Active Noise-Canceling Headphones

Contact: Audio-Technica | Price: \$219.95 |
330.686.2600 | www.audiotechnica.com

The concept of good noise-canceling headphones – eliminating background noise while providing quality audio playback – is great. Unfortunately, capturing background noise with a mic, reversing the phase so the original signal is canceled, blasting it into a pair of headphones and making it sound natural isn't as easy as it sounds. I've tried noise-canceling cans by several different manufacturers and have always been disappointed. They all have an awkward faded quality. I haven't tried a pair that remains comfortable over a long period of time, either.

I had actually given up on the concept altogether and just assumed that it couldn't be done ... well, until I tried the Audio-Technica ATH-ANC7 headphones at a recent tradeshow. A noisy tradeshow floor makes it hard as heck to demonstrate a microphone or pair of monitors, but it is the perfect place to exhibit a noise-canceling device. As skeptical as I was, I was sold the minute I tried them, and when I got home I immediately ordered a pair of my own.

The ATH-ANC7 lightweight closed-back headphones feature large-aperture 40 mm drivers with neodymium magnet systems that provide deep bass and extended treble. The noise-canceling electronics are fully integrated in each earpiece and are powered by a single AAA battery (providing 40 - 50 hours of use). Audio-Technica claims that the ANC circuitry reduces environmental noise by up to 85 percent.

These headphones fold flat for easy portability, and include a detachable cable with 3.5 mm mini-plug, a 1/4-inch adapter, a specialized airline adapter and a handy carrying case. Though it's not a unique feature to the Audio-Technica phones, the detachable cable allows them to be used simply as a noise-canceling device without any audio, making it much easier to sleep or just relax in a noisy environment.

As I write, I'm seven hours into my 13-hour flight from Hong Kong to Los Angeles and I'm once again wondering how I ever traveled without these little noise-eliminating gems. In a single flight I've used them to listen to music on my iPod, watch a movie on the plane's in-flight movie

UpSampler continues on page 34 ►



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headphone extension cable with lapel clip, power supply,
USB cable (A to Mini B) and protective carrying case with mic pouch

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system, "beat detective" a lame drum performance using my M-Audio Traveler USB interface and Pro Tools M-Powered, and, finally, take a pownap in nearly total silence.

The noise-canceling capabilities combined with the natural sound and slight top-end boost make the phones the perfect solution to doing productive DAW work in flight. I've found that with the noise-canceling feature I can listen at a substantially lower volume, allowing me to work much longer without any fatigue.

Another great feature of the ATH-ANC7 phones is that the audio functions in passive

mode even without the battery. This means if your battery dies 20 minutes before the end of your movie you lose your noise canceling capabilities but you can watch the rest of your flick (not true with most noise canceling phones).

The ATH-ANC7 headphones are in no way a flat studio reference headphone, but they do sound extremely good, are comfortable to wear for even the longest flights, and do an exceptional job eliminating noise. I'm not sure how I ever managed to travel without them.

—Russ Long



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If there's one thing that every PAR reader needs, it's a good-sounding and rugged dynamic cardioid ... or three. For that very reason, our industry has its tried-and-true standards. However, even microphones that are strong enough to drive nails must be replaced now and then. I've yet to find a replacement for the venerable Shure SM57, but when it comes to ball-grilled, vocal-friendly handhelds, I've discovered a really good, and really affordable, choice: the Samson Q8 dynamic supercardioid.

A guitar player I often work with introduced me to the Q8. I reluctantly swapped out my tried-and-true standard at lead vocal (both he and a vocalist on the gig insisted). At mix position, I was very impressed: via his channel I was presented with great feedback resistance; high SPL handling capability; and crisp, smooth transients that cut through with a full-bodied response.

After the gig, I bought a Q8. Then a month later, I bought two more. Since then I've used it everywhere: snare, toms, guitar cabinets of all kinds, even a bass guitar cabinet, studio vocals (loud rock tracking) and live percussion. And it works as a workhorse should — always good.



Workhorse dynamics quickly lose their cases and are thrown into bags with cables, etc., and carried in every way to make it to the next stop. The Q8 is overbuilt to travel without protection and, with me, it travels "baggage" and I don't worry about it. Finally, I've heard several Q8 users comment out loud about how good the mic feels in your hand. Even the sweatiest-handed screamer shouldn't lose their grip on a Q8.

At a glance, Q8 specs aren't necessarily unique: it offers a neodymium dynamic element with a tight cardioid pattern, its frequency response is listed as "60 Hz - 16 kHz," and sensitivity is -52 dBV/pa (2.5 mv/pa), 300-ohm output impedance, etc. It is a unique product, though, and good enough to make it your next workhorse dynamic purchase. Pick up one or three, as I did.

—Strother Bullins

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by Richard Alan Salz

TASCAM DM-3200

Digital Mixing Console

This digital mixer offers an excellent value for operating at up to 96 kHz.

The middle of the mixing console market demands value for money spent, flexibility, space efficiency and, perhaps most importantly, tight integration with the digital audio

left me feeling that it belonged to a much more expensive device, from the nicely finished top surface, the solid feeling knobs and faders, and faux wood panels (that I had to tap to realize



workstation world. For these needs, TASCAM offers the DM-3200 digital mixing console.

| FEATURES

At \$3,000, the DM-3200 includes 48 channels of audio along with requisite DAW controlling, plus multichannel computer audio interface capabilities via the IF-FW/DM FireWire card. Unlike some midrange digital consoles, the DM-3200 operates at sample rates ranging from 44.1 to 96 kHz without the normal halving of channels at its maximum 96 kHz rate. An optional meter bridge (MU-1000) is available for those that desire it, but the console is fully operable without it.

The physical design of the console certainly

that they weren't real wood). The basic layout consists of 16 channel strips (each controlling three layers for up to 48 channels) with a touch-sensitive, motor-driven fader, as well as mute, solo and select switches, a rotary encoder and mic/line switch with trim control and pad. A multitude of source and monitor selection switches, talkback, solo, navigation buttons, rotary encoders and pushbuttons — and, of course, a nice LCD display — round out the front panel. A dedicated dynamics/equalization switch allows the 16 channel strip encoders to form a horizontal dynamics or equalization control strip for the selected channel ... in a word, "Cool." Profiles for Mackie HUI emulation, MOTU Digital

Performer, Cakewalk SONAR, Steinberg Cubase and Nuendo, and Apple Logic are included via CD for easy DAW integration.

The DM-3200 provides 16 analog inputs (XLR mic and TRS line); phantom power is engaged in blocks of four rather than individually. Control room monitor outs are 1/4-inch TRS, while studio outs are on XLR and unbalanced RCA jacks (the latter of which is admittedly odd). Eight TRS send/return jacks are available for use as individual channel inserts, or they may be used for aux sends and effects returns.

Three TDIF and an ADAT optical I/O are included, and there are two slots for additional analog or digital I/O. There are also two AES and S/PDIF connectors. Additionally, USB, Word Clock I/O, MIDI In/Out/Thru, SMPTE in, Sony 9-pin, quarter-inch foot switch and a connection for the optional MU-1000 meter bridge is included. A cascade port allows two DM-3200s to be linked together.

Channels 1 - 32 include a four band fully-parametric equalizer (each band can sweep from 31 Hz to 19 kHz) with shelving, high-pass and low-pass filters as an option, as well as a dynamics section that includes compressor, gate, phase-reverse, aux, assignable inserts, direct out and bus assignment. Both dynamic and snapshot automation is available and function as expected. There are also two onboard digital processors, one of which is a dedicated TC Electronic TC Works reverb.

| IN USE

I had a chance to use the TASCAM console over a several months; in that time, I was

FAST FACTS

APPLICATIONS

Project studio, studio

KEY FEATURES

48 channels, 16 faders; four-band parametric EQ on 32 channels; dynamics and TC Works reverbs per channel; three TDIF and an ADAT optical I/O; maximum 96 kHz rate; IF-FW/DM FireWire interface card; optional meter bridge (MU-1000)

PRICE

\$3,799

CONTACT

TASCAM | ☎ 323-727-7617
 ➔ www.tascam.com

unable to make it crash, hiccup or misstep in any way — kudos to TASCAM for the stable operating system.

As it was, I used the DM-3200 along with my standard Nuendo rig, connected to my PC via USB ports; subsequently, TASCAM's Nuendo control surface software made performing complex mixes vastly more pleasurable than using a mouse along with my standard Behringer BCF2000 and Frontier Tranzport combination. If you have extra desk-top real estate (the DM-3200, though narrow, is

PRODUCTPOINTS



- Excellent value
- Good sound



- Reverb offering is shy of "premium" status

SCORE

The DM-3200 looks and sounds good, with a user-friendly, functional interface.

pretty deep), the console is almost worth the asking price just for use as a control surface.

It's obvious that console design (as well as digital processing) has come a long way since the first affordable digital consoles. The DM-3200 has a much more open and "easy" sound quality to it ... more analog-like, if you will.

It's interesting how certain devices (whether microphones, preamps, compressors or consoles) can subjectively make something sound either bigger or smaller, more open or more closed in. Overall, instruments tracked and/or mixed through the TASCAM had a nice relaxed quality (especially at 96k) to them, with none of the harshness and hardness that many inexpensive digital devices seem to suffer from.

For instance, drum overheads can sound totally dreadful on some digital gear, while I had good luck using a Microtech Gefell M71K pair on a GMS maple drum kit thanks in part to the TASCAM. More important than the kit itself were the mix of Zildjian A-Custom and Paiste Signature cymbals, which have a ton of mid- and upper-frequency information and can easily sound harsh with the wrong equipment chain. The DM-3200 gave a good representation of the sound of the entire kit and prevented the cymbals from sounding spitty or too "white" in nature.

The center image was not quite as solid as the very best analog (or digital gear), but it was more than close enough for all but the highest-end sources. The apparent weight and depth of

the kick drum was not quite as well defined as the 10-times more expensive specialist analog console (with two-thirds less channels) that I normally use.

The microphone preamps were more than acceptable for a device in this price range, although you may find a need to use something with more color on vocal tracks (if you like that kind of thing), but I found it easy enough to get good vocal results using nothing but the console's onboard preamps.

I found the preamps to sound very dynamic, with punch on my current favorite small guitar amplifier (the Sonic Cord Toad) when used with an EMG-loaded Telecaster. The midrange snarl of this combination was well represented!

The equalization and dynamics sections are similar, in that they have more than enough horsepower to do the job ... and then some. I would have preferred if the sweep went both lower than 31 Hz and higher than 19 kHz — as it's sometimes nice to be able to boost extreme low and high frequencies — but this is a minor quibble.

The included TC Works reverb section is just shy of "premium" outboard reverb quality, but it's very close (perhaps I've been spoiled working with convolution reverbs). About the

only effect I didn't care for was the distortion preset, but those almost always are terrible on digital processors. Most of the others are usable, or better.

| SUMMARY

I will admit to skepticism when it comes to inexpensive digital gear. I mean, come on: a 48-channel digital mixer for three kilobucks? However, the TASCAM console is, to put it simply, an excellent value at its asking price. It looks good, sounds good and has all the right interfaces, including a user interface that will not have you reaching for the manual just to complete routine tasks. Make no mistake, this is not a Neve or an SSL for 1/100th the cost, but it's more than clean and punchy enough to please those of us used to working on high-dollar gear.

Richard Alan Salz lives in Southern Vermont and operates Vermont Audio Labs.

| REVIEW SETUP

UREI 809 and Fostex NF-1 monitors; Legacy PointOne Subwoofer; Pass Labs x250 amplifier; Audio Developments AD146 console, DAV Broadhurst Gardens mic preamp, Steinberg Nuendo 3.2

by Bruce Bartlett

DARTECH DART Pro 24 Restoration Software

This affordable 24-bit/96-kHz processing package cleans up almost any audio PC users can throw at it.

If your audio is "broke," this program can fix it. DART Pro 24 is the newest version of DARTECH's popular audio restoration software, and you can consider it a Swiss army knife of audio processing for PC users.

The program supports both 16- and 24-bit sound files with sampling frequencies up to 96 kHz, and includes nine new tools and functions compared to the previous version, DART XP Pro. Running on Intel Pentium PCs with Win XP, NT, ME or 2000, DART Pro 24 sells for \$299, while Dart XP Pro (16-bit only) costs \$99. Upgrades from previous versions are available at lower cost.

FEATURES

DART Pro 24 lets you record any audio source, clean up and improve the audio, and burn CDs with custom playlists. Use it to restore old 78's, film and video soundtracks, or old voice tapes. Clean up surveillance/security recordings as part of forensic work. Improve

music recordings made with old or low-quality equipment or restore and save an LP collection on CDs.

Here are the steps in a typical DART Pro 24 restoration job: Import the sound file that you want to work on. This is called the Source file. Its waveform shows up on your screen. Select "Restore" and choose an appropriate process. Set parameters then preview the result. This takes a few seconds. After you click on "Process," the Destination file's waveform appears on screen below the Source file. Save the destination file for more processing or burn a CD.

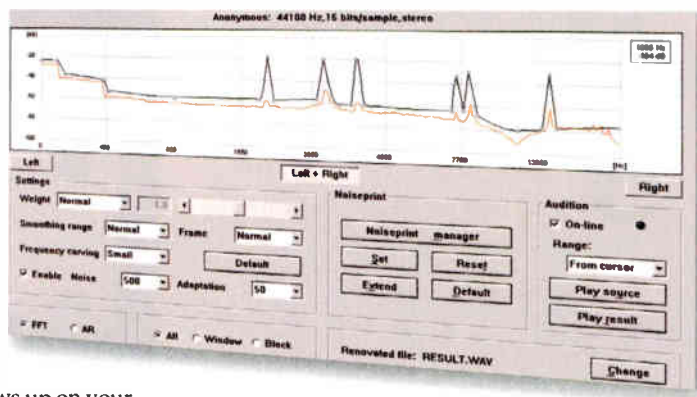
IN USE

DART Pro 24 offers a wide range of audio restoration functions. DeClick Plus removes impulsive disturbances (clicks) and low-level wideband noise. It gets rid of most clicks on 78's and LPs. DeCrackle removes clicks by using forward-time and backward-time signal analysis. It totally removes crackles on vinyl sources without affecting the music; processing time is a little longer than with DeClick Plus. ReTouch allows manual removal of overlooked clicks, or keeps signals that were misclassified as clicks. It works well to repair dropouts caused by DeClick. NoisePrint is a feature where you select a short piece of background noise before or after a recorded signal, then take a "noiseprint," or sample, of the noise. This noise is then removed from the entire program, using the DeNoise Plus algorithm that, when applied to an entire audio recording, reduces broadband noise that has the same spectrum as the sampled Noiseprint. Alternatively, you can extract the NoisePrint from one recording and use it in another.

With any de-noise algorithm, there is a tradeoff between noise reduction and artifacts. The cleaner the processed sound, the more you hear artifacts, such as swishing noise and fil-

tered speech. Typically you can reduce noise up to about 10 dB before artifacts become audible. You need to set the processing parameters carefully to get an artifact-free sound with adequate noise reduction.

Fortunately, DeNoise Plus lets you do that with surgical precision. The graphic below shows a NoisePrint spectrum taken from an audio recording using a digital camera's built-in microphone. Note the frequency spikes in



the bottom red curve.

Just above that curve is the noise-reduction threshold curve. You can set the threshold to work just on those frequency spikes. The result of this is less-audible artifacts in the processed audio than if you applied uniform processing to the entire spectrum.

When used to reduce broadband air-handling noise, DeNoise Plus was less effective than the de-noise function in DC-Art 32, a competing product. DC-Art 32 produced a few dB less noise with the same audible artifacts as DART Pro 24. However, DC-Art 32 threshold settings are at fixed frequencies, while DART Pro 24 lets you add as many frequency points as necessary so you can tailor the threshold curve more precisely.

MaskNoise, DeHiss Plus and DeHum are three important new features available in DART Pro 24. MaskNoise is a gentler type of noise suppression based on psychoacoustics. It identifies noise components in the program that are audible because they are not masked by the program. Only the audible noise is removed. In my tests, this did not work as well as DeNoise Plus. The resulting background noise and audio sounded "swirly."

When a Noiseprint is not available, you can use the DeHiss Plus noise-reduction utility based on a standard noise model. This tool eliminated the entire hiss in a cassette recording with almost no loss of highs in the audio signal.

DeHum applies a comb filter to remove harmonically related interference, such as hum and buzz. I tried it on a cassette recording with a very loud buzz and DeHum totally removed

FAST FACTS

APPLICATIONS

Studio, project studio, mastering, post, broadcast, audio-for-video

KEY FEATURES

Many audio restoration functions including MaskNoise, DeHiss Plus and DeHum; many audio processing tools; works with up to 24-bit/96-kHz audio files

PRICE

\$299 (DART Pro 24), \$99 (Dart XP Pro, 16-bit only)

CONTACT

DARTECH, Inc. | ☎ 952-844-9025
 > www.dartpro.com

the buzz without affecting the audio. I liked that I could tune the hum frequency slightly below 60 Hz, as the cassette was recorded off-speed.

More new features include the following: Batch processing performs multiple tasks in the background for lengthy unattended processing. Split Frequency separates a sound file into multiple frequency bands and processes each band independently. Add All recombines frequency-split files into a full-range file.

Divide/Combine is time division processing; it separates a sound file into different time

PRODUCTPOINTS

- Excellent reduction of hum, buzz and crackles
- Fine-tuning of De-Noise threshold
- Works up to 24/96
- File-split tools and batch processing

- PC only
- Some functions are handled better by other programs
- Some functions' GUIs could be more graphic

SCORE

DART PRO 24 does a fine job of repairing a wide range of audio flaws for PC users.

segments, and then processes each segment separately. For example, you might apply more de-noising to a quiet part of the program than to loud parts where the noise is masked. Resample converts any input sampling rate to any output sampling rate; this works very quickly and sounds great. Combine All combines multiple files end-to-end, automatically aligning the waveforms where they merge. Multi-cut tool removes a selected fragment from many files. Finally, you have MP3 import — rather self-explanatory one, no?

In addition to the restoration functions, Dart Pro offers a full palette of audio processing tools. In my use, all worked fairly intuitively and did the job quickly.

Here's an additional overview of some of the most notable tools:

Scale (adjust level), Maximize (Normalize), Reverse audio. Add two files (mix), Subtract two files (add in opposite polarity). Split a stereo track into two mono tracks, or mix a stereo track to one mono track.

Editing: cut, copy, paste and multi-cut. Smart Editing: cut, mute or crossfade highlighted sections of the program.

Graphic equalizer: It lacks the precision of a parametric equalizer, but is simpler to use.

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Low-pass, High-pass, Bandpass, Bandstop and Notch filters. Fade In, Fade Out and Crossfade: This feature's user interface is numeric rather than graphic, making it hard to use.

Change speed (program length) without changing pitch: This works without audible artifacts up to about a four-percent change; beyond that, the sound gets warbly. There are better programs available for this need. Shift pitch without changing speed: it is transparent up to about a four-percent change; with extreme settings, this is a cool audio effect, creating a harmonically rich signal.

Find Pattern: automatically detects constant noise/signal patterns. Remove vocal, add reverb: the reverb is good enough for Karaoke, but not very smooth. 2D and 3D spectral analysis: I like the Voxengo Span analyzer better because it lets you zoom in on the frequency axis.

Myfilter and Filter Builder: make custom combinations of tools and processing options. CD Rip and Burn: simple, easy and effective. Set/Drop Markers, Set/Remove frames, Divide/Combine time slices. DirectX audio plug-in support. Supports numerous SCSI and EIDE CD-ROM, CD-R and CD-RW devices. CD Player controls are as follows: re-order

tracks, add, delete, rename, move, play, pause, loop, etc.

For Realtime processing, Dart Pro 24 can process 16-bit/44.1-kHz audio input from a sound card. However, only the fastest computers can enable Dart Pro's realtime processing of 24/96 audio.

Last, but not least, Dart Pro 24's documentation is context-sensitive as an online Help and electronic manual. It is well written and easy to follow. I found most of the features easy to use after reading the Help files.

| SUMMARY

Dart Pro 24 has some restoration tools (DeCrackle, DeHum) that are extremely effective ... and some that are not so effective. To handle every situation, I'd recommend using more than one audio restoration program along with DAW plug-ins and stand-alone applications (such as Harmonic Balancer). Still, Dart Pro 24 does a fine job repairing a wide range of audio flaws, and it works with 24-bit/96-kHz audio.

Bruce Bartlett is a recording engineer, musician, tech writer and audio journalist. His latest book is Recording Music on Location published by Focal Press.

SSL Duende DSP Accelerated Plug-ins

by Dan Wothke

This console-grade 1RU module's 32 mono channels of proven, presence-boosting EQ and dynamics make ears smile.

Solid State Logic has been a standard in high-end professional audio production for decades. Smaller studios and home recording enthusiasts have long viewed SSL gear as an idyllic, yet non-realistic dream, and to incorporate their signature sounds into a production (unless a budget allowed for recording in a pro studio) was unrealistic, even while many rack-mount SSL modules became available during the last half-decade. Shelling out a few grand for any SSL signal path (or partial path) was, and is, still out of budgetary reach for many.

So, audio engineers all over the globe, and of all stripes, took notice when SSL introduced Duende about a year ago, because now anyone could buy 32 mono channels of SSL EQ and compression for around \$1,500 street. Originally introduced for the Macintosh PPC DAW market, Duende was later made available for PC and MacTel systems with the release of new drivers several months later.

Many users experienced compatibility problems and headaches with the early drivers. SSL worked hard to quickly handle these problems by establishing a dedicated section of its web-

site (www.solid-state-logic.com) for FAQs detailing common usage issues and solutions.

It was my assessment that since this was SSL's first venture into releasing its own end-user processing software, they should be extended some grace as they worked out the issues and released driver updates. Still, many frustrated users who shelled out their hard earned cash were rightfully a little reluctant to wait through the headaches.

Now, one year and multiple driver revisions later, it seems appropriate to revisit the Duende to check what progress has been made, and how viable it is for a user to consider this widely desired device.

FEATURES

The Duende hardware is a sleek, simple 1RU device with a very basic layout. It is lightweight, only four pounds, which is favorable for mobile use. On the front panel is a power button which glows with a blue rim when on. The rear of the unit consists of a power jack for the auto-adjusting power supply and two IEEE 1394 (400 Mbps) FireWire ports. The power supply is shipped with a number of international plug adaptors for use with the wall wart-type power supply. The unit can also be powered via the FireWire buss, providing the host computer has a 6-pin FireWire port and can handle such powered devices.

The Duende is not new code aimed at modeling SSL gear, but proven technology taken directly from the popular C series consoles, which were originally modeled from the 9000K, and E and G Series EQ.

The Duende Control Panel is the software for monitoring the unit's DSP usage, and is lightweight and basic with no user input required. It shows the four banks of processors built to handle eight mono or four stereo

channels each for a total of 32 channels at 44.1, 48, 88.2 or 96 kHz along with the Driver and Firmware version and the serial number of the unit. VST is the chosen interface standard for the Duende. Most audio applications that support VST will work with Duende, and SSL has a list of compatible applications listed on the company's website. RTAS and AU drivers for the Mac are supported via an integrated VST-RTAS custom wrapper developed by FXpansion.

IN USE

This is clearly one of the best DAW EQ and dynamic processing options on the market. As expected, the EQs are very smooth and easy on the ears; although this applies to all bands of EQ, I was particularly pleased with the upper frequency filters. Dialing in the perfect mix of presence and air is a cinch with the Duende.

The compressor performance was flawless: whether using it for minimal dynamics or really spanking the signal, the SSL lived up to full expectations. I quickly fell in love (again) with the SSL Stereo Buss compressor. It is simple enough for people not familiar with this particular compressor and it is like an old friend to those of us who have used it for years.

Many people have asked about the Duende compared to the popular Waves SSL plug-in bundle, and I had a chance to compare the two side-by-side. Although they are modeled after

FAST FACTS

APPLICATIONS

Studio, project studio

KEY FEATURES

32 channels of SSL C, E and G Series EQ and dynamics up to 96 kHz operation; standard VST interface; RTAS and AU drivers for Mac; 1RU hardware; Duende Control Panel software; two FireWire 400 ports

PRICE

\$1,795

CONTACT

Solid State Logic | ☎ 212-315-1111
 ☞ www.solidstatellogic.com



different SSL EQ styles, the Duende is clearly a smoother overall dynamic processor. The Waves is good for what it does, but the Duende, to my ears, is just better.

The Duende software is pretty straightforward. Color and layout is similar to an SSL 9000K channel module although more compact and includes standard routing options including where the filter is located in the chain and the ability to side chain. At the bottom of the display is a graphical layout of the signal path. The V2.0 driver provided mouse wheel control of the knobs. Users can Shift-click to fine-tune the knobs. A simple CTRL click (PC) or COMMAND click (Mac) resets the knob's position

back to default. The plug-in is missing the ability to double click and manually type in a numeric value. This is a feature I have been accustomed to having on other plug-ins, as I have found that it is sometimes easier to get the parameter to my desired setting without having to "turn" the knob. Duende also supports the ability to "freeze" tracks in Nuendo and Cubase, which frees up the DSP on the Duende. There is no graphical representation of the frequency plot when adjusting EQ; this is not critical, but has become a common feature for EQ plug-ins, and could be missed by users ... especially those not familiar with the general performance characteristics of SSL.

PRODUCTPOINTS



- 32 channels of true SSL dynamics
- Easy installation and interface
- Sounds amazing



- Compatibility
- Cannot manually type in values
- Requires dedicated FireWire buss

SCORE

Duende's 32 channels of SSL EQ and dynamics make it a remarkable performer, but beware of potential system compatibility issues.

No one will be disappointed with a Duende investment, except for when he or she encounters the aforementioned compatibility problems, and here is where the real Duende adventure began for me. I will reference some of my own experiences, as well as feedback from a few other engineers who have been active with the Duende since before the latest drivers have been released.

The first thing that someone considering a Duende purchase should note is the unit likes to have FireWire bandwidth all to itself. Problems are almost guaranteed when sharing the FireWire buss with any other device. This does not mean a separate port on the computer; it means a completely separate buss. Most motherboard-based FireWire ports use the same buss, so to successfully use the Duende when other devices are connected and in use a PCI FireWire card purchase is a must. (SSL lists compatible cards on the Duende support webpage). Potential purchasers should take this into account, because the limited number of PCI slots available on newer computers can be used up quickly (and laptops offer another set of challenges). With the late August v2.1 driver release, up to two Duende units can now be daisy-chained, pro-

viding 64 mono plug-ins at 44.1/48 kHz and 32 mono plug-ins at 88.2/96 kHz.

The Duende can also offer performance problems when interfacing with other VST/RTAS/AU plug-ins. I experienced this first hand when testing the first generation of drivers in January with a system owned by engineer David Terry of Nashville. He was running Nuendo 3.x with three UAD PCI cards and Lynx AES 16 audio cards on a professionally built custom audio PC. The Duende worked great at first try, but once we put one of the UAD plug-ins on a channel any other channel with the Duende in use was immediately muted. SSL's work-around for this was trying to reorder the UAD PCI cards, but since this was not my system I did not feel comfortable cracking open the case and swapping around his PCI cards.

With the Duende being lightweight, physically, it struck me that this would be a great piece to carry and use on different systems. However, after this problem I was cautious of this option. Once the 2.0 drivers were available I went back to David to try it on his system. He had since upgraded his DAW and put all of his PCI cards in an expansion chassis. At first, the Duende worked flawlessly on

this system, although I'm not sure if that can be accredited to the moving of the PCI cards, the new system itself, the Duende driver, or a combination of the three (I would gamble to say it was the latter). After a couple days of use, David experienced the mute bug when having a combination of native VSTs, UAD and the Duende inserted on the stereo buss. After a series of reboots did not restore the audio path, the problem was finally resolved by clearing the stereo buss and reinserting some of the plug-ins. Although we never confirmed exactly what had caused the problem, it was isolated to the Duende; whenever the Duende was removed from the buss, the audio passed through. When the Duende worked well, David found a new go-to EQ and track compressor and was reluctant to give it back to me for further testing.

On my home system — IBM Netfinity Server, VIA OHCI Compliant FireWire card, SONAR 6.0, Aardvark Q10 — everything was going great, even though none of my hardware was listed as compatible. The install was fine, SONAR didn't hesitate to recognize the Duende, and I was off to mixing.

Then I tried bouncing to disk. When listen-

SSL continues on page 42 ►

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STUDIO

REVIEW

SSL Continued From Page 41

ing back I noticed pops and clicks from the mix. I did some additional testing with Acid 6.0, which also played nice with the Duende, but had the same results when mixing down using the Render option. Any track with the Duende experienced the pops and clicks, including the stereo buss. After troubleshooting with the SSL team via the web interface, I determined that it was due to the Duende and the bandwidth needed when not bouncing in real time. When I bounced in real time, the pops and clicks were gone. The problem also occurred with David's latest setup and the Duende 2.0 drivers using Nuendo's offline bounce feature. Using the

Real Time bounce option, the pops and clicks were non-existent.

I spoke with Chuck Zwicky, engineer from New York, who is a Duende user with an Apple system running Logic 7.2.3. Chuck reported early problems with dropouts and timing issues, but suspected the problem was with the Apple FireWire driver. After running the update from Apple, everything has behaved quite nicely. Chuck also noted the quality of the Duende sound. In his exact terms, "the Duende EQ is the only digital EQ I have used that does not cause ear fatigue."

SSL continues on page 85

SSL DUENDE DRUMSTRIP:

The first add-on for the Duende is the SSL Drumstrip. The Drumstrip consists of five new and enhanced audio processing tools specifically for drums, both acoustic and electronic. These include a gate with a wide range of controls, a transient shaper, HF & LF enhancer, and a Listen mic compressor. The sections are bookended by Input — which includes metering and gain — and an Output section — with gain, metering and a Mix knob to control the overall amount of Dry vs. Wet applied to the signal.

The look of the Drumstrip is very sleek and modern. Each virtual knob changes color to represent the amount of processing for that particular parameter. The five different sections each have their own Power button to activate the section.

First, I ran the Drumstrip through the gamut with a full acoustic kit. It was very easy to dial in exactly what I wanted using the Gate. The controls were easy to navigate, and, just as importantly, the visual layout of the Gate was well thought out. There are different colored indicators on the Gate meter corresponding to Open, Close and Range parameters. Visually, this gate is a breeze to work and works well.

The transient shaper offers a way to grab the attack of a drum, adjust the characteristics of the attack and blend it back into the original signal. It is controlled by the input Gain, Amount and Speed. Audition quickly became my friend, as it acts similar to that of a side chain listen on a compressor; I could hear exactly what I was tweaking. Inverting the signal is also available, and is a great help when working to get the phase of a drum or combination in drums correct. Snare and toms were primarily used during my run. With the audition button, I could grab exactly what was needed and then mix it into the signal.

HF and LF enhancers, not the same as

EQs, work on the premise of dialing in 2nd and 3rd harmonics to the signal respective to the enhancer in use. Freq, Drive and Amount are the only controls for the enhancers, and, again, it was really simple to begin adjustments and discover the true potential of the drums. This is one of those plug-ins that I just start turning knobs to find what I like. I used the LF to round out the punch of a snare drum and it did wonders with the kick, really giving the extra bottom that was needed.



Here, the popular Listen Mic Compressor is modified for more control. Improvements include the ability to bypass the original hard high and low filtering, which allows the compressor to apply full bandwidth processing. I applied this to the mono overhead room mic, and for this particular application a little of this compression went a long way, although squashing the signal resulted in some creative results.

I found the Drumstrip to be a great tool for digging into drum sounds, naturally pulling out characteristics of the drum without adding unnecessary artifacts or over-hyped frequencies to the overall sound. SSL obviously put a lot of thought into the Drumstrip, and I was very appreciative as I did not even have to think when using these plug-ins; I just started working with the interface and enjoyed the results.

The Drumstrip is available as a free 30-day trial included in the new Duende V2.0 driver.

—Dan Wothke

AFFORDABLE LUXURY

YAMAHA'S LS9 DIGITAL MIXING CONSOLE is jam-packed with value. 2 models offer 16 or 32 channels with built-in recallable head amps, both expandable for twice the input capacity. Making life a little easier, the LS9 boasts a virtual effects rack, a surplus of EQ and dynamics on every channel, a color navigation screen and ultimate portability. An MP3 recorder/player and USB memory add even more worth to an already reasonable extravagance.

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by Rob Tavaglione

Manley TNT Microphone Preamplifier

Manley's new "dual-but-different" channel amp equals hours of tweaky and explosive fun.

The dual-channel Manley TNT is loosely named after its "Tube 'n No Tube" channels. But I contend that it could also refer to "Tweak 'n Tweak," as that's what

(nearly) a norm ... and those mics often require different preamp topologies with which to "paint."

For the TNT, the "Tube" side is based

tional 1/4-inch outputs. Output level can be switched from the rear panel, offering -10 dBV unbalanced and +4 dBu balanced or unbalanced modes.

| IN USE

With only a vented panel for top and bottom, the TNT's insides called for a closer look. I marveled at a beautiful layout and quality components throughout (12AT7 and 7044 tubes, multiple Lundahl transformers, rugged switches, sealed relays etc.). I proceeded to throw a little bit of everything into these circuits over the next few months, including close and ambient drums, amplified instruments, acoustic instruments, percussion, piano and vocals.



you'll be doing with this \$3,000 preamp's wealth of interesting options.

| FEATURES

According to the manufacturer, the TNT was born out of many users' requests for the tube mic amps of the Manley SLAM! (Stereo Limiter and Mic Preamp) without the other features. In addition, with so many engineers having access to unlimited (or at least very high) track counts, multiple mics on one source is

on a JFET/vacuum tube triode cascode circuit. The solid state side was created with a mix of discrete and op-amp amplification, as well as the Rapture Amp as the final line driver; Manley calls this channel the "Cool" side of the TNT.

Both sides offer independent phantom power, a polarity reversal, a high pass filter (80 Hz for Cool, 60 or 120 Hz for Tube), stepped gain controls (+20 to +70 dB), gain trim (from -10 up to +10 dB, allowing up to +80 dB gain overall) and impedance selection; each side offers numerous values (600, 2400 and 10 kilohms Tube; 2 MEG, 2000, 600 and 300 ohms Cool) and neither suffers from volume changes (an aid in making valid comparisons). Settings on the Cool side are actually achieved by switching between (or combining) two different stages of the solid-state amplifying circuits.

The Cool side also benefits from a Color switch, offering '60s and '70s settings. These switches engage additional circuitry, which simulates tape and guitar amp clipping of a light variety. The Cool side also allows control of its output transformer via the Iron control. This knob adjusts the audibility of the transformer from an exaggerated +3 to "bypassed" at 0 to the "opposite" of a transformer at -1.

Metering is provided from two sets of four LED's indicating signal presence, +10 dB, +22 dB and overload. Input/output is provided on XLR connections (except for front panel 1/4-inch DI inputs), with addi-

The TNT took each application with grace, revealing nuances of mic selection and placement without any signs of sluggishness, distortion or compression. Both sides offered unrestrained dynamics, openness and headroom, exhibiting linearity at nearly all gain levels. These qualities, and the TNT's quality construction, are to be expected from Manley's tradition, and are delivered here. The pleasant surprises truly lie in the additional features.

Seasoned pros know that "tube" does not always equal "fuzzy," and the tube side is wonderfully clear and clean. I found its impedance selection to be my paintbrush, with higher settings typically sounding more open and crisp (due to the lighter load) and lower settings affecting the bottom end and its tightness. In use, the subtle (and no so subtle) differences from mic-to-mic called out for much experimenting; sometimes the textural differences were merely academic, while at other times there was significantly altered balance and transient response. At times, impedance changes made me reposition my mic ... tweak 'n tweak.

The Cool side is truly the perfectionist's delight, however. The two HPF settings were useful, with the non-traditional 60 Hz setting being a welcome addition. The wide range of impedance settings dominated my curiosity with a great variety of possibilities, especially the 300C setting (this setting bypasses the preamp mixing circuit, a

MANLEY continues on page 46 ►

FAST FACTS

APPLICATIONS

Studio, project studio

KEY FEATURES

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PRICE

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purist setting if you will). When using the 1/4-inch instrument DI input the imped-

PRODUCTPOINTS



- Multiple topologies and signal path options
- Wide range of useful and creative features
- Quality construction and thoughtful design



- Expensive, even if warranted
- Limited metering
- Requires ample cooling in rack

SCORE

Although it's \$3k, Manley's TNT is a bargain by its expected longevity, resale value, quality and flexibility.

ance switch selects a different range of values for that input. The Cool side offers five impedance choices from 100K to 10

Megohms, allowing compatibility with everything from old-school single-coil pickups to modern humbuckers to line level-type sources ... more tweak 'n tweak.

The Color switch proved itself to be powerful, but not frequently called for. Both the '60s and '70s settings were fairly dramatic, altering frequency response and dynamics in obvious ways. Although I've been doing a lot of clean work lately, those of you who do highly stylized productions and work relying on grit and grain will utilize this feature on a daily basis. Combined with driving the input hard, the output trim and impedance settings, I found myself doing lots of long experimental soundchecks. Tweak ...

Topping it all off: the Iron control. Seasoned pros know that "transformer" does not always equal "fuzzy," either, so this control should not be expected to be some sort of overdrive. What it does do is impart subtle transformer coloration in the form of low frequency level and its distor-

tion, high frequency level and its dynamics. I found myself split on the use of Iron, sometimes rounding out harsh aggressiveness, sometimes getting out of the signal's way to retain clarity. Sometimes this control made me reconsider mic issues again ... you know by now.

SUMMARY

Manley quality does not come cheap, but this unit is still a bargain due to its expected longevity, resale value, quality and flexibility. Engineers who travel with gear will value its portability vs. versatility ratio. Engineers who like to double mic sources and track either similar or dissimilar options will revel in its inherent flexibility. Engineers who like to make quick decisions and aren't concerned with colorful options take heed; you may require blinders to block out features vying for your scarce attention ... my initial re-naming "tweak" is amply supported.

Rob Tavaglione owns Catalyst Recording in Charlotte, NC. You can find him at www.catalystrecording.com.

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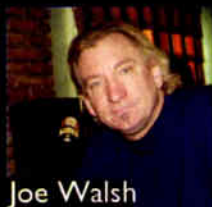


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YAMAHA

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by Bruce Bartlett

Cakewalk SONAR Producer 6.2.1 Digital Audio Workstation

This affordable, capable DAW for PC-based engineers offers 64-bit floating-point processing.

Cakewalk's premier digital audio workstation, SONAR Producer (\$619) is a high-end but affordable DAW/MIDI sequencer for Windows XP and Vista. After using it for several years I've only scratched the surface of its powerful, tightly packed features.

Often today's productions are equally packed and increasingly require a large number of tracks and processor-intensive plug-ins, but this is not a problem for SONAR Producer. It can work with 64-bit or multi-core CPUs that permit very high track counts without dropouts. This overview will describe facets of Producer's 6.2.1, as well as provide some perspective by revisiting features that appeared in Producer 5 (*Producer 7 has been released since the completion of this review, and it will be addressed in an upcoming PAR*).

FEATURES

SONAR Producer 6 screens look great: colorful, uncluttered and customizable. You can show as many or as few controls and buttons as you like, change colors, create custom toolbars, organize plug-in menus, give your drivers and ports "friendly" names, and so on. A new console view is attractive and ergonomic.

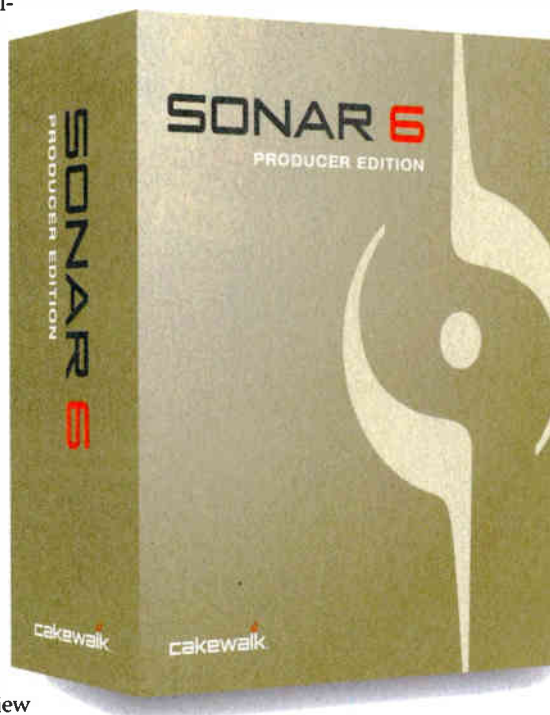
Producer 6 offers a full complement of soft synths and sample players. For example, Session Drummer 2 (new in version 6) offers professional multi-sampled drum kits, with hundreds of drum patterns included.

The new Synth Rack is a GUI that lets you conveniently view, insert, delete and configure your virtual instruments. You can mute or solo a synth, choose patches, automate changes and more. To reduce CPU overhead and free up memory you can freeze the track, which converts the synth output to an audio track.

Active Controller Technology automatical-

ly maps any MIDI controller or control surface to software effects, soft synth controls or SONAR mixer controls. ACT assigns your knobs and sliders to plug-in or synth parameters and you can customize the mapping.

When you open a new plug-in,



ly maps any MIDI controller or control surface to software effects, soft synth controls or SONAR mixer controls. ACT assigns your knobs and sliders to plug-in or synth parameters and you can customize the mapping.

The VC-64 Vintage Compressor plug-in is a cool-looking new tool with EQ, gating, compression, and de-essing. Its two compressors can be combined in series or parallel. Presets with various routing schemes are included.

Automation has been enhanced in SONAR Producer 6.2.1 thanks to the read/write controls for tracks and plug-ins. You can enable automation while playing back or recording.

Like Pro Tools' Beat Detective, SONAR Producer's AudioSnap syncs audio tracks

with each other or with a tempo map. For example, AudioSnap marks transients in a drum track and saves them to a "pool" that stores tempo data. Select the bass track, mark its transients and select Quantize to Pool. The bass now follows the tempo of the drums.

If an audio clip is made to play at a slower tempo, the clip has to stretch to a longer time. Sometimes the audio sounds glitchy if the time-stretch is done in real-time. Fortunately, Producer 6 comes with a powerful set of time-stretching algorithms that smooth out the sound during offline rendering. You need to choose the right time-stretch algorithm for the particular musical part to get the best sounding results.

Other notable features of SONAR Producer include Crash Recovery, protecting your projects from buggy plug-ins. Automatic file versioning backs up your projects with time/date stamps so you can access older versions of projects. Analyst spectrum analyzer reveals problem frequencies. And finally, SONAR now handles both VST and DX plug-ins with-

FAST FACTS

APPLICATIONS

Project studio and studio

KEY FEATURES

Fully-functional and feature-packed DAW well suited for pro and project studio use; good plug-ins; 64-bit audio engine; PC only

PRICE

\$619 list (Version 6); \$399 for an upgrade from earlier versions

CONTACT

Cakewalk | ☎ 617-423-9004
 ☞ www.cakewalk.com

out the need for an external wrapper.

The SONAR 6.2 free update adds features such as MIDI input quantize, permanent time display in the track view, a Bit Meter which monitors the bits in a track, refinements to ACT, Windows Vista support, and X-Ray Windows which allow you to see through a plug-in's GUI to control the tracks below. According to Cakewalk, version 6.2 also provides "WaveRT driver support for more efficient low latency playback and MMCSS for higher priority audio processing." Update 6.2.1 optimizes the program for higher track counts at high sample rates and bit depths.

CAKEWALK continues on page 50 ►

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A lot of composing these days is done with loops, or repetitive musical patterns. SONAR's Loop Construction view lets you edit loops, and control and automate various loop parameters. A groove clip is a loop (audio or MIDI) that can follow the tempo and key of your project. Dozens of groove clips are included in SONAR Producer, and you can import ACID-style loops. Each clip can be repeated indefinitely by click-dragging the right end of the clip.

The RXP REX Player and Groove Box let you use REX or REX2 loops in your projects and trigger individual slices of those loops.

Of course, SONAR also excels at handling sounds from live musicians. An efficient way to start recording is to open a multitrack template: a set of tracks with plug-ins already inserted. Use one of the included templates or create your own.

Comping is made easy with track layers and the Mute Tool. Each successive take is recorded as a new layer (virtual track) within a track. Then you can keep only the best parts of each take. Mute or solo parts of various takes and auto-crossfade them to create the final perfect track. All layers on the same track can share common plug-in effects.

SONAR comes with tons of plug-ins made by Cakewalk and Sonitus. There is EQ, reverb, echo, chorus, flanging, multiband compression, gating, pitch change, stereo effects, guitar-amp simulation and much more.

The program automatically compensates for sound-card and plug-in latency. All the tracks stay in sync regardless of the plug-ins you have inserted.

Need some good reverbs? Producer includes not only the Lexicon-designed Pantheon reverb but also Perfect Space convolution reverb. Cakewalk provided over 300 MB of samples — a huge variety of impulse responses of real spaces for use with Perfect Space, as well as sampled reverbs, amp cabinets and more.

Suppose you've set up a vocal track with various effects and bus routings. You can save it as a track template. Then, when you want to record another vocal with the same settings, just insert that track template. You can also clone a track: copy its settings and plug-ins to a new track. That really comes in handy when you're overdubbing vocals. *(According to the manufacturer, track templates also help automate effects on outputs in multiooutput synths like*

Session Drummer 2 — Ed.)

Are some vocal notes flat or sharp? Highlight a vocal phrase containing a flat note and call up SONAR Producer's included Roland V-Vocal. A graph will appear showing the pitches of the notes in the phrase. You simply click-drag the pitch of the flat note to where it should be, and you then adjust the formant parameter for best sound; V-Vocal is also excellent for quick adjustments to volume and timing.

Some of us are getting into surround



SONAR Console and VC-64 Vintage Channel Views

mixing, over 30 formats of which SONAR has supported since version 5. SurroundBridge technology lets you use stereo VST effects in a surround mix. The Sonitus Surround Compressor works on the complete surround mix.

Three 64-bit enhancements put SONAR Producer on the cutting edge of recording technology:

1. Starting with version 5, SONAR was the first DAW to include 64-bit (double-precision) floating-point processing, which is accessible even on 32-bit computers. The 64-bit engine is claimed to provide more accurate processing, dithering and summing than 32-bit engines — especially when used with 64-bit plug-ins (which SONAR includes).

2. SONAR can handle 64-bit floating-point audio files. You can import these files, mix and render them with the highest quality available.

3. If you have Windows XP 64-bit edition, a 64-bit CPU, 64-bit drivers for your sound card and 64-bit virtual instruments or plug-ins, you can use them with SONAR Producer 5 or 6, which are 64-bit applications. Two claimed benefits are faster processing and lower latency (depending on the application). Other benefits

are more simultaneous tracks, real-time effects and virtual instruments. Plus, the x64 platform allows up to 128 GB RAM so you can load larger, higher-quality samples into memory.

Producer works with 32-bit or 64-bit applications and operating systems. You can run 32-bit apps in a 64-bit operating system and with 64-bit sound cards, but you lose the benefits mentioned above.

IN USE

First let's address the foundations of Version 5. The included Sonitus effects sound great and are highly adjustable. The quality of reverb in Producer is extremely

high, both in the included Lexicon Pantheon reverb and Perfect Space convolution reverb.

I love the Track Templates and the Clone Track feature. They really speed workflow, because you can add pre-configured tracks on the fly during a session.

If you want to apply an effect for a short time — such as an echo at the end of a vocal phrase — use clip effects, which apply to a clip rather than an entire track. That's often easier than automating an echo send.

V-Vocal's pitch correction sounds natural. In my experience, V-Vocal is easier to use than AutoTune.

The included soft synths are powerful and flexible. However, many of the synth patches of acoustic instruments sound synthetic, so you might want to purchase or create some good samples *(According to the manufacturer, this is addressed in Producer 7 with the inclusion of Dimension LE and the Garritan Pocket Orchestra. — Ed.)*

An effective way to automate level changes is by creating a volume envelope in a track: a graph that shows the fader level over time. Adjust the fader envelope until the mix sounds right. The fader follows the envelope you made. If you change

your mind about the track's overall level in the mix, however, you can just press "O" to go to Offset mode and adjust the track fader. The envelope stays intact. Envelopes can be used to automate panning or aux-send levels as well. For me at least, envelopes are the easiest way to automate track levels and sends.

SONAR has POW-r dithering, one of the best available. When you export a 24-bit mix to 16 bits and add POW-r dithering, the

The many drum kits in Session Drummer 2 sound realistic, but take up a fair amount of CPU resources and memory. I like the attractive graphic interface. Its main panel could use a label for drum-pad velocity and a few steps for basic use.

Using Active Controller Technology is much easier than manually assigning a MIDI controller's keys and sliders to the software parameters. If your controller is in the ACT presets list, just select it and

you're ready to go.

Vintage Compressor 64 is a piece of work. The many included presets are useful for a variety of instruments, vocals and mixes. Sounds range from transparent to warm and colorful. Putting two compressors in series — each with a 3:1 ratio and half-second release time — resulted in transparent, non-squashed 9:1 compression!

In the heat of a session you might accident-

CAKEWALK continues on page 52 ►

PRODUCTPOINTS

- Powerful
- Great sounding new plug-ins (VC64, Session Drummer)
- AudioSnap tightens up a musical performance
- Can handle 64-bit files, 64-bit processing, 64-bit OS, multi-core CPUs, Vista
- Dozens of convenience features to enhance workflow (ACT, synth rack, Analyst, lock clips, fast zoom, X-ray)

- AudioSnap and Session Drummer 2 GUIs could be clearer
- Soft synth and MIDI install as two tracks rather than one

SCORE

SONAR is a 64-bit capable, affordable PC-based DAW.

result sounds almost like a 24-bit recording.

Is the 64-bit mix engine a significant enhancement? I compared a 64-bit-processed mix to a 32-bit-processed mix. The 64-bit version sounded very slightly "sweeter" or "smoother" and more transparent than the 32-bit version. In the quest for analog-like sound in digital workstations, every little bit helps. I performed a null test by adding the 64-bit and 32-bit files in opposite polarity. This resulted in continuous low-level hiss and very low-level, grainy hiss modulated by the audio signal.

Now let's move on to Producer 6, which installs easily. Once you have registered, you can download the free updated versions 6.2 and 6.2.1. They are well worth the download, especially if you plan to upgrade to Windows Vista. Many users have wanted the program to quantize MIDI notes while recording and the 6.2 update does that. If your projects involve high-resolution audio, version 6.2.1 helps to keep the track count high without dropouts.

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tally slide a clip in time (which I've done quite often). To prevent that, SONAR Producer 6 lets you lock the position or data of clips.

Fader automation has been made easy. Press the Write button, tap the spacebar to play the track and move the fader as needed. After rewinding press the Read button. Then play the track to hear the fader moves. You can rewrite the moves as needed.

AudioSnap takes some patience to use, but when it works it's amazing. The accuracy of AudioSnap's time correction depends on which transients you select to include in the pool. You can minimize the artifacts of time-stretching by rendering the stretched clips offline. I've found AudioSnap most effective in locking bass to drums, less effective in locking drums to acoustic guitar. However, if you spend enough time it can be made to work. Sometimes it's easier just to slide individual notes to adjust their timing.

The concepts in AudioSnap are confusing to many users. Fortunately, Cakewalk's website has some helpful video tutorials about

AudioSnap and other topics. Also go to acapella.harmony-central.com/forums then check out "Pro Reviews."

I'd like to see a simpler AudioSnap interface, something like, "Make track X follow track Y" or "Make track X follow the tempo map." Rather than choosing a time-stretch algorithm to suit the music, I'd rather choose "percussive part," "staccato part" or "sustained part."

I'm using Producer 6.2.1 with a modest computer: 2.2 GHz single-core AMD and 1.5 GB of memory. It has not crashed yet. I've played 100 24-bit/44.1-kHz tracks simultaneously without plug-ins and up to 40 audio tracks with plug-ins. Your results might vary, depending on your computer and sound card. I was not able to test Producer 6 with a 64-bit machine, but some users in the online forums report track counts exceeding 100 with plug-ins when using a 64-bit set-up.

How about support? Loads of helpful tutorials are on the Cakewalk website. The SONAR 6 reference manual is available for

\$49 (According to the manufacturer, a PDF accompanies the software. — Ed.). Another fine reference is "SONAR 6 Power" by Scott Garrigus at www.digifreq.com, which also has a great SONAR discussion group. Also, the online help files within SONAR are clear and thorough. Cakewalk has a number of excellent user support groups on their website. Often, the company has incorporated customer suggestions for features that add convenience and speed workflow.

SUMMARY

I've found SONAR Producer 6.2.1 to be a joy to work with. After some learning time and practice, it is fast and intuitive to use. Everything I need to do for elaborate productions — recording, MIDI, editing, mixing, effects, automation, sync to video, mastering — can be done in-the-box. And thanks to the fine plug-ins, Pow-R dither and 64-bit audio engine, the sound is superb.

AES and Syn Aud Con member Bruce Bartlett runs a commercial recording facility with SONAR Producer as the heart of the studio.

OFTEN COPIED, NEVER EQUALED

The advertisement displays a collection of Groove Tubes audio equipment. At the top, a row of microphones is shown, each with its name and model number printed below it: HARDTOP, CONVERTIBLE, GT30, GT40, GT50, GT60, GT57, GT67, MD1B FET, MD1B TUBE, and VELO 8. Below the microphones are three preamplifier units: THE BRICK, GLORY COMP, and SuPRE. To the right of these units is a large, detailed image of the VELO 8 microphone. The background is dark, and the equipment is arranged in a professional, studio-like setting. The text "OFTEN COPIED, NEVER EQUALED" is prominently displayed at the top in large, bold, white letters. At the bottom, the website "www.groovetubes.com" is listed on the left, and the slogan "Hear the Difference™" is on the right, accompanied by the Groove Tubes logo.

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The A7 finally brings the acclaimed ADAM folded ribbon tweeter to a lower price point, delivering the same imaging and extended top end as its bigger brothers. Its brand new carbon fiber woofer yields superb mid- and low-frequency details.



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ADAM Audio

One of the jewels of the ADAM line, the P22-A was primarily designed for project studios. However, the P22-A is also an excellent choice in any application – broadcast/post applications to surround monitoring – calling for a medium-sized powered monitor.



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ADAM Audio

The acclaimed S3-A (ADAM's highest profile monitor) is actually a three-way speaker in which one woofer acts as a full-range driver with the second woofer joining it only below 150 Hz. This effectively doubles the area of the woofer for deep, tight bass response.



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Adamson Metrix Multi-Purpose Line Source Series

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Visit AES Booth 254 or www.adamsonproaudio.com

Adamson Metrix WAVE

These ultra compact enclosures feature the newly designed Adamson ND8-ML 8.5-inch Kevlar neodymium mid-low frequency driver is optimized for the 2-way enclosure, and is paired with a 1.4-inch HF (3-inch diaphragm) compression driver, mounted on Adamson's patented HF wave-shaping chamber. The ultra light Metrix-i weighs only 41 lbs, and the WAVE-i comes in at 43 pounds.



Visit AES Booth 254 or www.adamsonproaudio.com

Adamson Metrix Convertible Sub

The Metrix convertible cardioid sub boasts two newly designed ND15-L 15-inch Neodymium Kevlar drivers manifold mounted within an ultra compact, fully tuned, vented enclosure. It is flyable within the same array or behind the main array when trim heights are limited. Optional are lightweight aluminium rigging frames, dollies, custom flight cases and soft covers.



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This small-format mixer features six mic/line channels, seven additional stereo channels with routing to channel strips or to L/R, USB output from L/R or Aux busses and USB input from a computer, DuoPre preamps, 69 dB gain range, 3-band EQ with MusiQ and SONAR LE. \$499.



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API A2D

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API 5500

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Conceived by Joshua Aaron and manufactured by Jim Maher of Sterling Modular, the Control Room Concept can hold most all of your studio equipment and includes two ISO boxes, room for multiple displays, speaker arms, cable management, solid-wood construction. It gives your control room a breathtaking look and feel.



192 pt., 3U, TT->DSUB patchbay with standard TASCAM wired DSUB connections (the same as the Lynx Aurora, Digi 192 and Apogee converters), large labeling strips, QuickSwitch technology allowing for the normal and ground configuration to be decided on per channel, a lifetime warranty, and, of course, outstanding audio quality.



Audio-Technica's new flagship ATH-M50 professional studio monitor headphones offer maximum isolation and comfort. Designed to provide an exceptionally natural response for professional monitoring and mixing, the closed-back, collapsible headphones feature 45 mm neodymium drivers, circumaural ear pieces (180-degree swivel) and luxuriously padded ear cushions. A gold-plated mini plug includes screw-on adapter.



Audio-Technica's groundbreaking SpectraPulse Ultra Wideband (UWB) wireless microphone system bypasses the congested RF bottleneck to deliver 14 channels of clear, intelligible audio without the performance/set-up issues associated with conventional wireless. Designed for conferences, courtrooms and corporate events. Optional encryption meets the NIST-approved AES encryption standard developed by the U.S. government for safeguarding sensitive material.



Audio-Technica's AT2041SP offers two versatile condensers for home/project/commercial studios. The critically acclaimed AT2020 side-address is ideal for vocals/instruments; the AT2021 end-fire is a natural for acoustic instruments, overheads, piano. The AT2041SP now comes bundled with Cakewalk's SONAR LE, a pivoting stand mount, a microphone clip and protective pouches.



The M1280, available in four polar patterns, is a miniaturized condenser microphone with a fully integrated mic preamp and immunity from RF.



The M1250, with RF immunity, is available with cardioid, hypercardioid, omni or shotgun capsules—allowing for a wide variety of applications.



The VX5 brings studio quality sound to the stage with pad and roll-off, resistance to plosives, off axis rejection, and ability to handle high levels of sound pressure.



The award-winning ADC1 is a reference-quality, two-channel, 24-bit/192-kHz digital-to-analog audio converter. It features Benchmark's phase-accurate UltraLock clock system with multi-function clock input and word clock output. One of the four digital outputs allows a low-resolution output to a CDR, while the other three operate independently at up to 192/24.



The DAC1 USB is a reference-quality two-channel 24-bit/192-kHz digital-to-analog audio converter featuring Benchmark's Advanced USB Audio technology, UltraLock clock system and HPA2 headphone amplifier. The 24-bit/96-kHz USB output makes it ideal for digital audio workstations and computer-based media playback.



World Radio History

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The VX-5 handheld condenser from Audix brings studio quality sound to the stage.

The flexibility, resistance to plosives, off axis rejection, and ability of the VX5 to handle high levels of sound pressure will satisfy the most demanding performers and engineers.

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VICTOR: \$169.00
Long Element Ribbon



FAT HEAD II: \$199.00
Single Element Ribbon



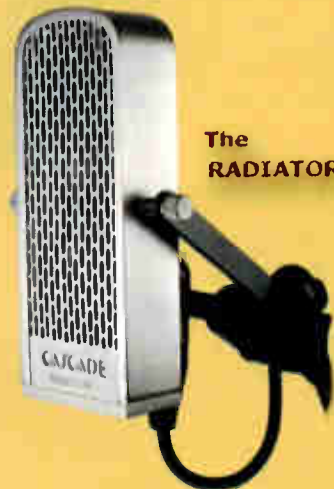
ELROY: \$299.00
Tube Multi-Pattern LC



FAT HEAD: \$159.00
Single Element Ribbon



X-15: \$499.00
Stereo Single Element Ribbons



731R: \$299.00
Dual Element Ribbon



VIN-JET: \$199.00
Long Element Ribbon



DR2-S: \$599.00
Stereo Dual Element Ribbons



L2: \$399.00
Stereo Multi-Pattern LC

CASCADE MICROPHONES
Olympia WA
cascademicrophones.com
360.867.1799

See us at AES Booth # 568

This is a select sampling of products to be displayed at the 2007 AES show, October 3-5, Javits Center, NYC

DPA Microphones 3521

The 3521 Stereo Kit is composed of two 4021 compact cardioid microphones carefully matched on frequency response, sensitivity and self-noise. The 3521 includes a hard shell carrying case and multiple mounting accessories for applications such as acoustic piano, overhead for drums, horn or string sections, choirs, and acoustic ensembles.



Visit AES Booth 928 or www.dpamicrophones.com

Earthworks PianoMic System

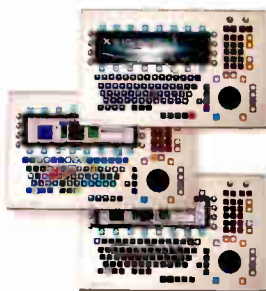
The ultimate piano microphone system that will change piano miking forever. Benefits are High Definition Piano Microphones, superb sound with the piano lid either up or down, incredible gain before feedback, virtually no leakage from surrounding instruments and invisible from outside the piano.



Visit AES Booth 658 or EarthworksAudio.com

Fairlight Xynergi Unified Media Production Center

This digital interface delivers a combination of technology, software and tactile control. Featuring Fairlight's revolutionary new self-labeling key switch technology and powered by the breakthrough CC-1 engine, the Xynergi is compatible with most widely used media processing tools. \$23,000.



Visit AES Booth 142 or www.fairlightus.com

Frontier AlphaTrack

AlphaTrack is the portable DAW control surface designed to help you mix and edit better and faster than ever. AlphaTrack combines natural controls, like a motorized fader and transport buttons, with productive innovations, like touch-sensitive encoders, LCD display, and unique jog/shuttle strip. The result is one powerful and attractive package that travels anywhere. \$249.



Visit AES Booth 756 or www.frontierdesign.com

Groove Tubes SuPRE

Two-channel stereo microphone and instrument preamplifier. High-resolution tube signal path with 72 dB of gain on each channel. Variable transformer impedance settings. Large custom rotary variable attenuators. Legendary GT sound and build quality.



Visit AES Booth 736 or www.groovetubes.com

D.W. Fearn VT-15 Recording Channel

D.W. Fearn introduces the VT-15 Recording Channel, an all-vacuum tube 3RU unit that combines the sound and features of the VT-2 Microphone Preamplifier, the VT-7 Compressor, and a sub-set of the VT-4 Equalizer controls.

D.W. FEARN
www.dwfearn.com

Visit AES Booth 858 or www.dwfearn.com

Equi=Tech Model 20WQ Wall Cabinet Balanced Power Distribution System

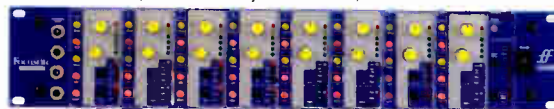
Equi=Tech, "The Pioneer of Balanced Power," introduces the newest member its family of products: the Model 20WQ wall cabinet balanced power distribution system. The 20WQ features a massive yet precisely balanced toroid isolation transformer at its heart, a 200 amp load capacity and 20 branch circuits for hardwiring AC power.



Visit AES Booth 638 or www.equitech.com

Focusrite ISA828 Eight-Channel Mic Preamp

Eight original ISA series transformer-based preamps in a single robust 2U chassis offers selectable impedance per channel (including original ISA 110 impedance setting), direct instrument inputs on the front panel, high pass filter, phase reverse and phantom power per channel. Additionally, there are convenient 25-pin D-Type connectors and insert points on every channel. \$2,999.99.



Visit AES Booth 136 or www.focusrite.com

Furman Sound IT-20 II Balanced Isolation Transformer

Designed for the most critical, ultra-low noise installations, the IT-20 II can supply 20 amps of balanced AC power. The IT-20 II's ultra-low noise balanced isolation transformer provides over 80 dB of common mode noise reduction from 20Hz - 20kHz, ensuring the lowest noise floor possible for today's sensitive recording equipment.



Visit AES Booth 615 or www.furmansound.com

Groove Tubes VELO-8

Classic ribbon mic with a modern twist. Includes extra field-replaceable ribbon assembly. Advanced dynamic design with Neodymium for maximum sensitivity and wider response. Selectable impedance and roll-off switches. Velvet-smooth response that all mic lovers will appreciate. Includes hard mount and shock mount.



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See us at AES Booth #845

This is a select sampling of products to be displayed at the 2007 AES show, October 5-8, Javits Center, NYC

Heil Sound The Fin

The Fin is a large diaphragm dynamic microphone that brings a one-of-a-kind visual element to the stage. List Price: \$240.



Visit AES Booth 850 or www.heilsound.com

Heil Sound Spotlight Series Gold Pearl

Large diaphragm dynamic microphone similar to the successful PR 20. Gold pearl finish, also available pink (benefits the Koman Cancer Center), red, and white. List Price: \$210.



Visit AES Booth 850 or www.heilsound.com

Heil Sound Handi Mic Pro Plus

Compact, low profile dynamic microphone designed for applications that require small (4.5-inch) mics capable of handling very high SPL. Perfect for drum kits. List Price: \$110.



Visit AES Booth 850 or www.heilsound.com

Holophone H2-Pro

The simple and elegant Holophone H2-Pro Surround Sound Microphone with modified LFE is the only patented device specifically designed for capturing discrete 5.1, 6.1 and 7.1-channels of surround sound. The system's flexibility, ease of use, and performance makes the H2-Pro ideal for capturing live events and is perfect for studio use.



Visit AES Booth 1236 or www.holophone.com

Holophone H3-D

The Holophone H3-D Surround Sound Microphone is designed to deliver crystal clear 5.1 surround sound capture for pro audio, live music production, educational applications and faith-based facilities, as well as professional broadcast studios. The H3-D is a powerful tool for engineers, content producers and artists on all levels to craft realistic and exciting surround sound tracks.



Visit AES Booth 1236 or www.holophone.com

Holophone H4 SuperMINI

The breakthrough Holophone H4 SuperMINI Surround Sound Microphone system delivers expansive 5.1 channel audio field capture in a compact package, mountable on any professional quality video camera. The system offers an integrated multi-channel pre-amplifier, virtual surround headphone monitor and Dolby Pro Logic II surround encoder.



Visit AES Booth 1236 or www.holophone.com

Lavry Engineering Black AD10

New AD with exclusive *Digital Alias-Free Emulation* modes: clear, tube, transformer, both, +4 XLR/10 1/4-inch inputs, ultra-stable internal/external clock.



Visit AES Booth 249 or www.lavryengineering.com

Lavry Engineering Blue 2A2D2MicPre

Modular system: AD, DA, and MicPre modules can be mixed in 1U rack for up to eight channels conversion/4channels MicPre.



Visit AES Booth 249 or www.lavryengineering.com

Lavry Engineering Gold AD122-96 MkIII

Gold Standard Mastering AD with soft saturation for maximum clean level. True 24-bit/-127dB noise floor and unrivaled detail.



Visit AES Booth 249 or www.lavryengineering.com

Millennia Media HV-3R eight-channel remote controllable microphone preamp

The Millennia HV-3R employs the same HV-3 circuit found in their HV-3C and HV-3D units. Remote control is achieved via Millennia's ultra-high performance ÆLogic software. MIDI and Ethernet are its primary remote protocols. The MIDI interface allows plug-and-play compatibility with Pro Tools systems.



Visit AES Booth 611 or www.mil-media.com

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Gabriel**, **Status Quo**... Denmark

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T21sub

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Millennia Media STT-1 Origin Twin Topology Music Recording System

The Millennia STT-1 is a single channel music recording system offering a selection of Millennia's core vacuum tube or solid state circuits at every function, including transformer or transformerless mic preamps, line input with gain, opto-compressor/limiters, parametric equalizers, DI instrument input with gain, and opto-de-essers.



Visit AES Booth 611 or www.mil-media.com

Millennia Media TD-1 Twin Topology Half-Rack Recording Channel

The Millennia TD-1 features selectable vacuum tube or all discrete solid-state DI amplifiers, dual Reamp outputs, Line input with 20 dB switchable pad, HV-3 microphone preamplifier with 65 dB gain, dual band "NSEQ" style fully parametric EQ (20 Hz - 25 kHz), audiophile-grade headphone output.



Visit AES Booth 611 or www.mil-media.com

Mojave Audio MA-200

The MA-200 is Mojave Audio's large diaphragm tube condenser microphone, featuring a 3-micron capsule, Jensen transformer, NOS JAN 5840 tube and a fixed cardioid pattern. Includes heavy-duty carrying case, shock mount and cables. List Price: \$995.



Visit Booth 866 or www.mojaveaudio.com

Mojave Audio MA-100

The MA-100 is Mojave Audio's latest – a small diaphragm tube condenser microphone that features an interchangeable cardioid and omnidirectional capsules, a Jensen transformer and NOS JAN 5840 tube. Includes heavy-duty carrying case, mic clip and cables. List Price: \$795.



Visit Booth 866 or www.mojaveaudio.com

MXL USB Mic Mate Universal USB Microphone Interface

The Mic MateT enables any dynamic or condenser microphone to be connected to a Windows or Mac OS personal computer. With a 20 Hz - 20 kHz frequency response, this is a USB 2.0 Plug-and-Play interface that provides 48V Phantom Power for studio condenser mics and facilitates easy laptop recording.



Visit AES Booth 432 or www.mxlms.com

Novation ReMOTE 25 SL Compact

This USB MIDI controller's 25 semi-weighted keys with aftertouch, knobs, switches, programmable trigger pads, assignable pitch bend/mod wheels, LCD screen and exclusive Automap Universal technology allow intelligent control of software/plugin-ins. Novation's Xcite+ bundle and Ableton Live Lite 6 accompany. \$399.



Visit AES Booth 136 or www.novationmusic.com

NTI DR2 Digirator

NTI's handheld Digirator delivers a wide variety of stereo test signals at up to 192 kHz/24 bit, plus Dolby D, D+, E, DTS, DTS-HR surround & powerful tests for measurement of transparency and signal latency.



Visit AES Booth 260 or www.nti-audio.com

NTI MR-PRO Analog Minirator

The battery operated MR-PRO is optimized for maximum features & functionality with THD+N < -96 dB and +18 dBu output, together with monitoring of impedance, signal balance and phantom power.



Visit AES Booth 260 or www.nti-audio.com

NTI AL1 Acoustlyzer

NTI's handheld AL1 Acoustlyzer, for professionals requiring one audio test tool with both comprehensive electrical and acoustical measurement capability & PC data transfer.



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Pass Labs XVR1 Electronic Crossover

Each high- and low-pass filter is user configurable as 6, 12, 18 or 24 dB per octave slope, and each "pole" of the four slope settings is independently adjustable from 22 Hz to 18,000 Hz. In addition, there are two independent Q (sharpness) controls for each filter.



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This is a select sampling of products to be displayed at the 2007 AES show, October 5-8, Javits Center, NYC

Pass Labs INT-150 Integrated Amplifier

First integrated unit from Pass Labs, 150 watts per channel into 8 ohms, 300 watts into 4 ohms. Two balanced/single ended inputs and two single ended only inputs. Balanced and single ended preamp outputs. Remote control. Introductory retail price \$6500.



Visit www.passlabs.com

Peluso P12

Inspired by the legendary AKG C12, the P12 has nine polar patterns controlled from the power supply and faithfully reproduces the airy sonic qualities and robust low end of the original C12. It comes as a complete system with the flight case, shock mount, power supply, and more.



Visit AES Booth 126 or www.pelusomicrophonelab.com.

Prism Sound Orpheus FireWire Interface.

With no compromise and Prism Sound quality, this is the DAW interface you'll need. Eight analog i/o including four mic pre's and two instrument. 10 digital i/o comprised of ADAT, AES and SPDIF. Plus OverKillers, MIDI, wordclock i/o and Prism's legendary clocking complete the perfect box.



Visit AES Booth 712 or www.prismsound.com

Royer Labs R-122

World's first Active Ribbon microphone. Phantom powered ribbon mic has 13 dB more output than our R-121. Compatible with almost any mic preamplifier.



Visit AES Booth 922 or [royerlabs.com](http://www.royerlabs.com)

Royer Labs R-122V

World's first vacuum tube ribbon microphone. Unprecedented richness, depth and detail in the midrange frequencies. Dedicated power supply with shock and road case.



Visit AES Booth 922 or [royerlabs.com](http://www.royerlabs.com)

Royer Labs SF-24

Stereo Active Ribbon microphone. Condenser level output and impedance matching circuitry. Works with phantom power and eliminates preamplifier mismatches and gain-related noise.



Visit AES Booth 922 or [royerlabs.com](http://www.royerlabs.com)

Rane G4 Quad Gate

Combining familiar front panel controls with the accuracy of digital signal processing, Rane's full-featured G4 Quad Gate is suitable for a variety of demanding dynamics processing applications. Gating, ducking and downward expansion modes are offered, together with internal high- and low-cut filters and external side-chain inputs.



Visit AES Booth 551 or www.rane.com

Rane PEQ 55 Parametric Equalizer

Rane's dual-channel PEQ 55 features 5 bands of fully parametric equalization, adjustable High-Cut and Low-Cut Filters, and three bands of Accelerated Slope Tone controls per channel. Front panel controls select between dual mono 5-band, stereo linked 5-band or stereo linked 10-band operating modes.



Visit AES Booth 551 or www.rane.com

Rane C4 Processor

The C4 is a four-channel broadband compressor, dynamic EQ and peak limiter in a 2RU chassis. All signal processing is done with advanced DSP algorithms and uses Rane's proprietary Relative Threshold method.



Visit AES Booth 551 or www.rane.com

Rupert Neve Portico 5015 Mic Pre/Compressor

The Portico 5015 is single-channel Mic Pre and Compressor with independent input and output transformers for both mic pre and compressor circuits. The mic pre channel is identical to the 5012 and 5032 modules, Custom transformers, "Silk" and a swept HPF. The compressor-limiter has fully variable controls for threshold, ratio, attack, release, and make up gain. The 5015 also includes an external link for stereo and multi-channel configuration and Feed-forward/Feed-back selection to change the compressor's response from a slower, more musical style to a faster, more transparent sound.



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- ♦ Speech Intelligibility STI-PA
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- ♦ **Real Time Analyzer:** Fast RTA with 1/3 and full octave resolution. Calculates SPL, LEQ and Max/Min for each band. Relative display against a stored reference. Fast logging via PC interface.
- ♦ **Zoom FFT:** Extremely fast, real-time Zoom FFT with resolution to 0.7Hz over the entire frequency range. The ideal tool for visualization of comb filter and narrow band effects.
- ♦ **Reverberation Time RT60:** Octave band RT60 measurements according to ISO3382 with auto trigger, ranging and averaging.
- ♦ **Delay Time:** Delay time between reference signal and input from built-in microphone, using special chirp.
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A vintage Neumann U87 microphone, a large-diaphragm condenser mic, is shown mounted on a silver-colored metal stand. The microphone has a distinctive cylindrical shape with a silver grille and a black body. The stand features a complex, adjustable boom arm and a base with four legs. The background is a plain, light-colored surface.



A professional condenser microphone with a silver grille and black body, mounted in a black shock mount. The microphone has the 'Neumann' logo on the grille.

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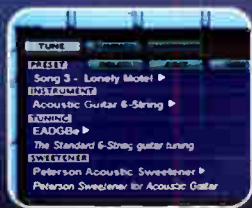
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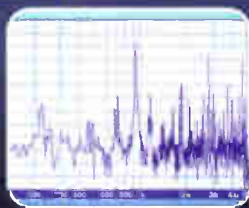
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This is a select sampling of products to be displayed at the 2007 AES show, October 5-8, Javits Center, NYC

Sound Devices 702T Two-Channel Audio Recorder with Time Code

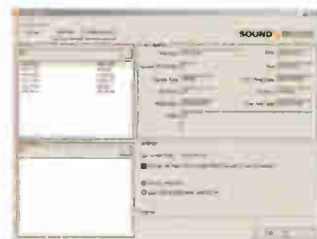
The two-channel 702T is a powerful two-track file-based audio recorder with time code. The compact device records and plays back to convenient, removable CompactFlash cards and external FireWire storage devices, making field recording and dual-system production simple and fast.



Visit AES Booth 918 or www.sounddevices.com

Sound Devices Wave Agent

Wave Agent is a Windows-based multi-format file conversion utility. The application converts data-compressed FLAC audio files generated by Sound Devices 7-Series recorders into Broadcast Wave files while maintaining all metadata. Additionally, Wave Agent converts sound files between monophonic and polyphonic WAV file formats. Available as a no-charge download.



Visit AES Booth 918 or www.sounddevices.com

Steinberg WaveLab Essential 6

Audio Editing, CD/DVD Burning and Podcast Publishing. Offering a high-value suite of audio editing tools tailored to the needs of musicians, smaller recording environments and podcast authors, WaveLab Essential 6 integrates podcast tools with professional editing, restoration and mastering capabilities.



Visit AES Booth 826 or steinberg.net

Steinberg Nuendo 4

"Created for Engineers!" Delivering a next-generation audio production environment for audio post, and production, Nuendo 4 elevates creativity and productivity to new heights. Extraordinarily intuitive and fully customizable, Nuendo 4 offers streamlined, precision tools that save time and boost creativity in today's media, recording and film industries.



Visit AES Booth 826 or steinberg.net

THAT Corp 1280 Series Dual Balanced Line Receiver IC

The THAT 1280 Series of precision differential amplifiers was designed primarily for use as balanced line receiver for audio applications and are available in gains of 0 dB, +/- 3 dB and +/- 6 dB. All three versions of the part typically exhibit 90 dB of common-mode rejection and are available in a RoHS-compliant 14-pin SOIC package.



Visit AES Booth 809 or www.thatcorp.com

THAT Corp 2162 Dual Pre-Trimmed Blackmer VCA IC

The THAT 2162 contains two high-performance Blackmer voltage-controlled amplifiers (VCAs), and operates from a split power supply up to +/- 16 Vdc. Available in a RoHS-compliant 16-pin QSOP package, the 2162 is extremely flexible and capable of being configured for a wide range of stereo or multichannel applications.



Visit AES Booth 809 or www.thatcorp.com

THAT OutSmarts Balanced Line Driver

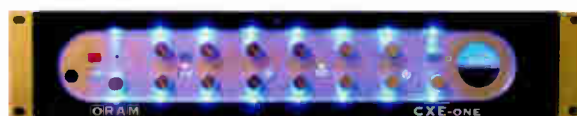
THAT is proud to introduce the latest in their series of analog I/O ICs with a difference: OutSmarts, the new Balanced Line Driver. With patented dual feedback-loop design, it performs better when clipping into single-ended loads, without compromising differential performance. At -101 dBu (unweighted), OutSmarts delivers at least 3 dB lower noise than the competition.



Visit AES Booth 809 or www.thatcorp.com

Trident Audio ORAM CXE-ONE Channel Strip

Ultra-high quality channel strip featuring mic preamp, instrument input, multi-band EQ with High- and Low-pass filters and Dynamic Limiter/Compressor.



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Trident Audio The Light Console

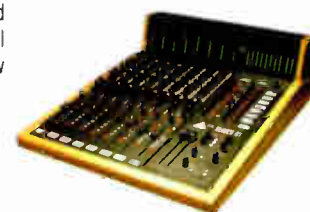
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This is a select sampling of products to be displayed at the 2007 AES show, October 5-8, Javits Center, NYC

Yamaha Commercial Audio Systems MY16-MD64, MY16-ES64, MY16-EX

The Yamaha MY16-MD64 card adds 16 channels of MADI input and output connectivity to Yamaha audio devices with mini-YGDAI slots; up to three MY16-EX expansion cards increase capacity up to 64 channels. The MY16-ES64 card adds 16 channels of bi-directional EtherSound connectivity; up to three MY16-EX cards increase capacity up to 64 channels.



Visit AES Booth 318 or www.yamahaca.com

Yamaha Commercial Audio Systems TXn Amplifier Series

The Yamaha TX4n at 2,200 watts, TX5n at 2,500 watts, and TX6n at 3,000 watts per channel into 2 ohms offer legendary Yamaha on-board DSP power and incorporate a sophisticated 24-bit/96-kHz DSP engine that enables an extraordinary range of control and processing capabilities.



Visit AES Booth 318 or www.yamahaca.com

Yamaha Commercial Audio Systems ACD1 Amplifier Controller

This new device enables complete remote control and monitoring of power amplifiers and provides a convenient, reliable means to connect a computer running Yamaha's advanced NetworkAmp Manager II amplifier control and monitoring software to Yamaha Tn, PC1N, XP, XM and XH series amplifiers.



Visit AES Booth 318 or www.yamahaca.com

Zaxcom Deva 5.8 Hard Disk Audio Recorder

The Deva 5.8 is a 10-track hard disk recorder for film and television production. With eight hardware faders and built-in hard drive, DVD-RAM and Flash memory slot the Deva 5.8 is a complete solution for recording mixing and effects in one package.



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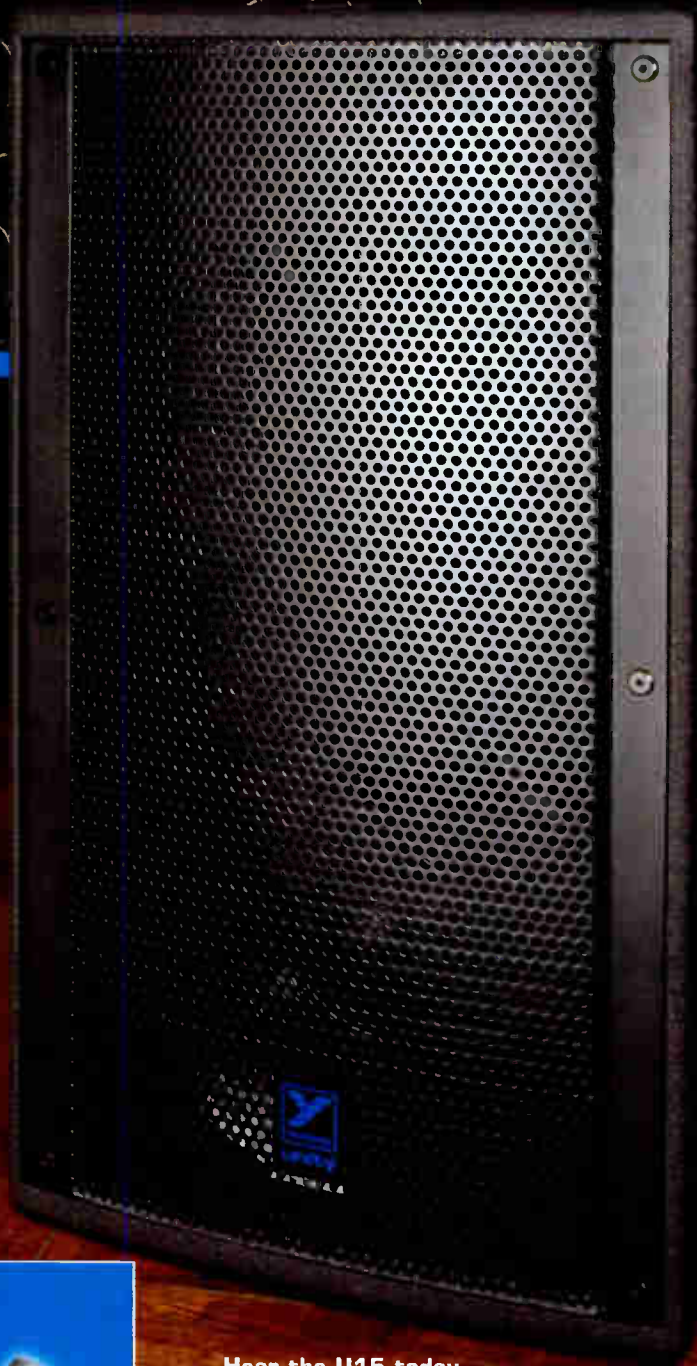
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NEW PRODUCTS

DACS Clarity HeadLite 2 Headphone Amplifier



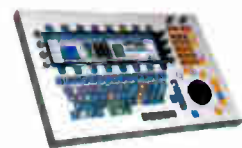
Clear your mind and cover your ears (with headphones), because your mind is about to be blown by this four-input, four-output amplifier. The

HeadLite 2 features new ICs for improved performance, especially with low impedance headphones. Two self-cleaning push buttons are on the rear panel for adjusting input sensitivity. You'll also love conductive plastic pots, upgraded selector switch and a balanced input option using the THAT InGenius IC. Standard channels easily drive 400 ohm headphones at maximum output levels, and with excellent signal to noise ratios, low cross talk, minimal distortion and a bass response that stays flat down to under 20 Hz, you've got a rugged and reliable package that will last for years.

PRICE: \$860

CONTACT: DACS/Independent Audio | ☎ 207-773-2424 ➔ www.dacs-audio.com

FAIRLIGHT Xynergi Unified Media Production Center



Like shampoo and conditioner, you need your programs and equipment to work together cleanly, synergistically. You need this unique desktop interface that offers intuitive tactile control of a media center, powered by the Fairlight CC-1 platform and compatible with the most widely used media processing tools. The Xynergi controller features self-labeling key switches that display full-color images, icons or text, supporting any type

of language or icon-driven menu with an "on-demand" QWERTY keyboard for easy labeling/document programs/email. Eight touch-sensitive rotary controllers and multiple soft keys offer precise editing/mixing control over multi-band EQs, three stage dynamics, multi-dimensional panning, aux sends and more. And, given an operator's specific task, relevant functions are displayed, while those that aren't are hidden. The result is less keystrokes, more intelligent control and greater productivity.

PRICE: \$23,000

CONTACT: Fairlight US | ☎ 626-793-3940 ➔ www.fairlightus.com

SONODORE Small Diaphragm Omni Microphones



When it comes to the perfect microphone for applications such as an orchestral score recording, "going Dutch" won't mean only picking up half. This ultra-high quality small diaphragm omni microphones line made by Dutch manufacturer Rens Heijnis Audio Electronics is built to satisfy the most scrupulous recording aficionado. Especially

appropriate for SACD or DVD formats, Sonodore mics bring a robust musicality without sacrificing extraordinary dynamic range, low noise or accuracy. While a "flat" acoustic grid is included with the Sonodore RCM-402 omni microphone (pictured with 2-channel 60-volt outboard power), interchangeable acoustic grids can be used as accessories to customize the response for different applications. The sound, design, price, time is right.

PRICE: \$3,300 (RCM-402 microphone), \$600 (PS-402)

CONTACT: Sonodore/TransAudio Group | ☎ 702-307-2700 ➔ www.lasvegasproaudio.com

CROOKWOOD M Class Mastering Consoles



Whether working with analogue, digital or analogue plus digital machines across stereo or surround environments, this series' six small footprint controllers (the 19-inch M3 is shown) monitor sources, select sources for a record path and insert processing equipment into the path in any order. Monitoring is equipped with Bob Katz's K scale calibration, while all audio work is performed remotely in 1RU units. Acoustic reflection is minimal,

while audio is passively controlled by relays and buffered where required by true balanced amplifiers for neutral monitoring. Options include MS processing, internal converters and metering.

PRICE: \$4,000 - \$24,000

CONTACT: Crookwood | ☎ 011 44 1672-811-649 ➔ www.crookwood.com

Telarc International continues its upward trajectory, bringing Direct Stream Digital (for Super Audio) recordings to the masses, by adding a **Digital Audio Denmark** AX24 A/D converter to its arsenal. The immaculately clean-sounding DAD AX24 - which is available with two, four, six or eight I/Os and supports up to 24-bit, 385 kHz - immediately won over a mastering suite full of Grammy Award-winning ears when played through a set of carefully aligned ATC monitors.

Speaking of ATC monitors, it's been ratted out by Grammy Award-winning mastering engineer Gavin Lurssen that the **ATC SCM150ASL** was the crucial set of monitors used while he delved way into the audio when finalizing the sound track for Disney's *Ratatouille*. These unflinchingly accurate drivers have also driven the mastering of Sony Picture's *Across the Universe*, plus releases for Alison Krauss, Robert Plant and Matchbox 20.

Here at **PAR** we got no problem with Michael Jordan. But when we say, "Be like mic," we mean microphones. In this case, a multiple-capsule **SoundField SPS422B** surround sound microphone, which was used by veteran television and movie composer Gary Change and collaborator Neil Leonard as they played and recorded ambient electronic music through a state-of-the-art sound system in five of Italy's medieval "acoustics-as-alchemy" sanctuaries in June 2007 (an example of which is pictured). The SPS422B allows for monitoring in mono while recording in surround, capturing a lossless stream that SoundField software coupled with **Pro Tools** could manipulate to present the exact spatial imaging desired when allowing the walls to express their centuries-old design. **Tara Labs** interconnects, **Blue Sky** loudspeakers and a **Marantz** PMD570 were also used for the material, which was encoded to **DTS**.



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World Radio History

by Bob Katz

Lipinski Sound L-3601 PowerStand

This unique speaker stand doubles as one of the best-sounding amps this reviewer has ever heard.

I was the first U.S. mastering engineer to own a pair of uncompromising, hair-raising Lipinski Sound L-707 loudspeakers (which I declared "tight, clean, dynamic and open" in the June 2004 *PAR*), which have since received numerous accolades. So it's no surprise that I'm looking at Lipinski amps and speakers for my Studio B (now under construction).

In Studio A, we have a Pass X250 Class A stereo power amplifier, which is heavy and hot. Looking for something conversely compact, I was very interested in auditioning Lipinski's new L-3601 PowerStand, a handsome integration of their Class D (PWM) monoblock L-301 amplifier and a 36-inch speaker stand, designed to put the tweeters at ear height for the most neutral frequency response. One of the attractions of Class D amps is that they run cool

and draw no significant power line current at idle, which would make a considerable saving on my Florida electric and air conditioning bill compared with the Pass Labs' constant 270 watts idle and 1,000 watts maximum. Though Class D amps have improved since 2004, when I rejected one in favor of the Pass, I decided to be skeptical but hopeful about the Lipinski.

FEATURES

Each Lipinski L-301 monoblock has a single line input, either unbalanced on RCA or balanced on XLR. There is also a balanced and unbalanced line output suitable for feeding a subwoofer amp. Two independent 300-watt power amplifiers inside the monoblock can be run in biamp or bridged mode. In biamp mode, connect one amp to the tweeter and the other to the woofer for a total of 2 x 300 watts per channel; that is more than twice the rated power per channel of the Pass X250, though to my ears Class A power amps tend to deliver like AB and D amps of twice the rating. Lipinski does not recommend using bridged mode, as it reduces the damping factor; it should be reserved for high power PA applications, a single 600-watt amplifier per monoblock.

When the amps arrived, I was impressed by their light weight; I can lift one in each hand, but it takes two men or a forklift to move a single Pass X250. Each Lipinski amp has a clear Lucite plate covering an embossed logo. When powered, the front panel and logo can be illuminated in a variety of user-settable color schemes (or not if you prefer). It is a gimmick, but a very nice looking one. I set them to a copper-gold color, which looks, I must say, sexy; it gives the illusion of a gold plate reflecting the ambient light of the room. I give the look a big thumbs up. I wish I could say the same about the metalwork, which is "semi-industrial," and the back panel silkscreen labels are light gray on grainy gray metal.

The object of powered stands is to combine the advantages of powered loudspeakers (short loudspeaker cables for lower resistance) with

the advantages of separates (reduction in distortion due to microphonics). Andrew Lipinski says he has measured up to 10-percent vibration-induced distortion in some powered loudspeakers due to microphonics. Well, that won't happen with these, at least after I spent 15 minutes assembling each stand and an hour per stand filling the pipes with white sand from Home Depot. The sand reduced the resonance of the pipes considerably, but not completely. But in use I could not detect any sonic effect. Nevertheless, I recommend that Lipinski revisit the design of the tubes. I used four Dr. Scholl's



round callus cushions on top of each stand to ensure good contact with the speakers (don't laugh). For bi-wiring, I built some three-foot lengths of custom speaker cable using Mogami 8-core 2.5 mm OFC speaker cable, using up all eight wires for an equivalent of better than 10 AWG and terminating in gold Atlona locking banana plugs.

IN USE

Finally, after a two-day warm up period
LIPINSKI continues on page 78 ►

FAST FACTS

APPLICATIONS

Mastering studio, mixing and recording studio

KEY FEATURES

Class D (PWM) monoblock L-301 amplifier; 36-inch speaker stand; a single line input, either unbalanced on RCA or balanced on XLR and a balanced and unbalanced line output suitable for sub input

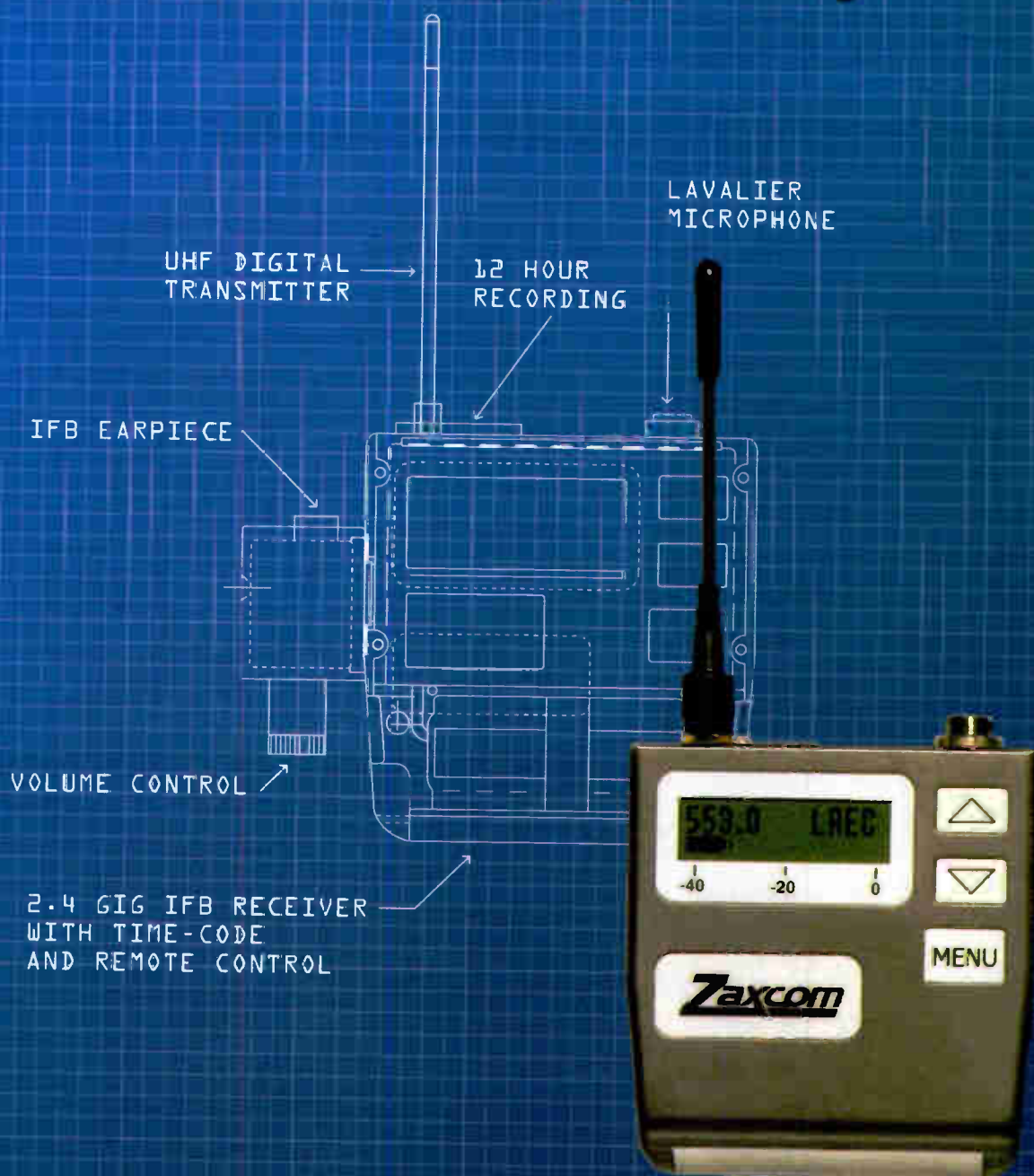
PRICE

\$2,995 (Monoblock L-301 amplifier, 2x300 watts), two for \$5,990; \$595 (L-360 stand without amplifier), two for \$1,190; \$3,590 (L-3601 powered stand), \$7,180 for two channels

CONTACT

Lipinski Sound | ☎ 916-273-9726
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World Radio History

came the listening. I used the balanced XLR inputs to mate with my Crane Song Avocet (reviewed the December 2004 issue of *PAR*) and matched levels against the Pass to 0.1 dB using a Crane Song input attenuator. I recommend low gain mode, since high gain produces

a slight but not bothersome hiss – audible only in a totally quiet room like mine but still to be avoided. At low gain with the L-707s, the hiss is only audible within a few feet.

Using my collection of 24-bit masters and playing through the Avocet's excellent DAC, I was immediately impressed with the sound. It's very musical, with a great midrange: pure and sweet but with an open top and a holographic, dimensional soundstage. The bottom end is extremely deep, tight and (I am astonished to say) more extended and solid than that of my Pass X250, which costs about three times as much! Soundstage width also slightly exceeds the Pass because dual mono amps mean better channel separation. You can play these puppies as loud as you want without any distortion or harshness.

| SUMMARY

These are the best amplifiers I have heard in this room, beating (by far) the Bryston, Lexicon and Hafler, and now proving even marginally better than the physically more imposing Pass

X250 (original model). Five of these will make an excellent set of uncompromised surround amps; unlike the Pass they'd probably perform from one 20-amp wall outlet, though I recommend two dedicated circuits when separate powered subwoofers are considered.

This amp uses a Second Generation Digital module, which Lipinski has integrated to get the best sonic performance. Andrew says he is not interested in building gear unless he can make it sound better than anyone else's: a bit of an ego statement, but I think the amp results speak for themselves. If this had been available two years ago, I would not have bought the Pass, regardless of impressive appearance. Luckily, they are now obtainable for my Studio B. These Lipinski PowerStands are definitely worthy of integration into the most exacting of studios.

Founder of Digital Domain mastering studio in Orlando, Bob Katz has been a recording and mastering engineer for over 30 years, mastered three Grammy-winning and other Grammy-nominated albums, and is the inventor of the K-System of metering and monitoring, and the patented K-Stereo and K-Surround processes.

PRODUCTPOINTS

- Pure, dimensional, tight, solid, grainless, extended sound
- Deep, tight, extended bass and soundstage
- Amplifier can be used separate from the stands

- Expensive
- Semi-industrial metalwork

SCORE

One of the best-sounding power amplifiers this reviewer has ever heard.

hear this *the amethyst*

"Detailed and exciting, the Amethyst is at home in front of anything from horns to vocals, rock drums to screaming guitar amps. The Amethyst is a powerful tool, and its artful craftsmanship inspires creativity. Its consistently open and natural sounding—a must have for any mic collection."

— Reviewed by Beau Sorenson

Studio Engineer, Smart Studios, (Death Cab for Cutie, Garbage, Sparklehorse...)

You have a real hit with this mic!...Our engineers now grab for it like inmates at the asylum try to grab people walking down the hall. So forgiving on vocals and loud guitar amps...

— Reviewed by Bill Tullis

Studio Engineer, SoundsAtlanta Studios



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DTS Master Audio Suite V1.5

The DTS Master Audio Suite (MAS) software is a one-stop application for HD audio encoding, decoding, quality control and bit-stream management, and is targeted at facilities and individuals that need to prepare Blu-ray Disc and HD DVD optical discs in addition to DVD-Video, DVD-Audio and DTS Surround Music CDs. The recently announced V1.5 update adds a host of new features, including full Mac-Intel compatibility, the powerful stand-alone DTS-HD StreamPlayer, plus useful support for Blu-ray's interactive Secondary Audio layer: DTS Express audio streams in mono (48 - 96 kbps), stereo (64 - 192 kbps) and 5.1-channel, including downmix (192 - 256 kbps) and DTS 5.1 Music CD Encoding. In addition, CD mastering users can encode a 5.1 or DTS-ES 6.1 bit-stream at 44.1 kHz, and also produce a standard-format DTS .wav file that can be imported into conventional CD-audio burning software. Real-time DTS-HD and DTS file playback also is provided with synchronization to picture. For enhanced compatibility, the MAS application uses standard ASIO hooks via Windows XP and Core Audio under Mac OS X.

Usefully, all file loading is via an easy-to-initiate drag and drop GUI, with all settings capable of being saved to create custom tem-

plates for encoded schemes using common parameters. Also, downmix settings can be saved as part of an encode scheme or separately. To save time, a single encode can serve for both HD DVD and Blu-ray projects, with Auto Verify and Run PBR for each session; batch encoding also is now available for Mac OS X and Windows XP users.

AN ENHANCED WORKFLOW PRODUCT

In essence, Master Audio Suite is designed as an enhanced workflow product intended to create, evaluate and prepare DTS-HD digital audio streams that are com-

patible with both HD DVD and Blu-ray optical disc formats, but which can also be used to create legacy streams for use on other optical formats. A companion product from DTS, Surround

Join Tool, SAS supports the creation of audio streams compatible with DVD-Video, DVD-Audio, and DTS Surround Music CD in formats up to 6.1 channels. (SAS can be upgraded to a full version of Master Audio Suite via an electronic license key.)

MAS runs under Windows XP and Mac OS X operating systems using a conventional Sun Java 2SE Runtime Environment; recommended RAM is at least 2 GB. The application ships on a pair of CD-ROMs — one to load the Master Audio Suite Encoder and StreamTools components and the other for the DTS-HD StreamPlayer. Simple-to-follow setup screens let the user select session parameters relevant to the bitstreams being prepared, including DTS-HD Master Audio and DTS-HD High Resolution Audio (HD DVD and Blu-ray), DTS 96/24 and DTS-ES, plus DTS Digital Surround (for DVD, HD DVD and Blu-ray) and — uniquely — DTS Express (HD DVD and Blu-ray interactive audio). StreamTools consists of editing tools for streams created with Master Audio Encoder, and usefully includes Join, Append, Trim, Split, Restripe and Verify operations, plus the ability to retrieve high-level information about the bitstream. While most of the heavy lifting will have already been achieved in a DAW or multichannel editor, there are often occasions while preparing DTS-encoded bitstreams when it can be very handy to remain inside the application and adjust some timing issues within the production workflow, and avoid a quality-compromising trip back to PCM audio format to edit and re-encode. For added flexibility, it goes without saying that MAS supports both lossless (DTS-HD Master Audio) as well as "lossy" (DTS-HD HR and DTS) encoding formats.

MASTER WAY TO DTS-COMPLIANT BITSTREAM CONVERSION

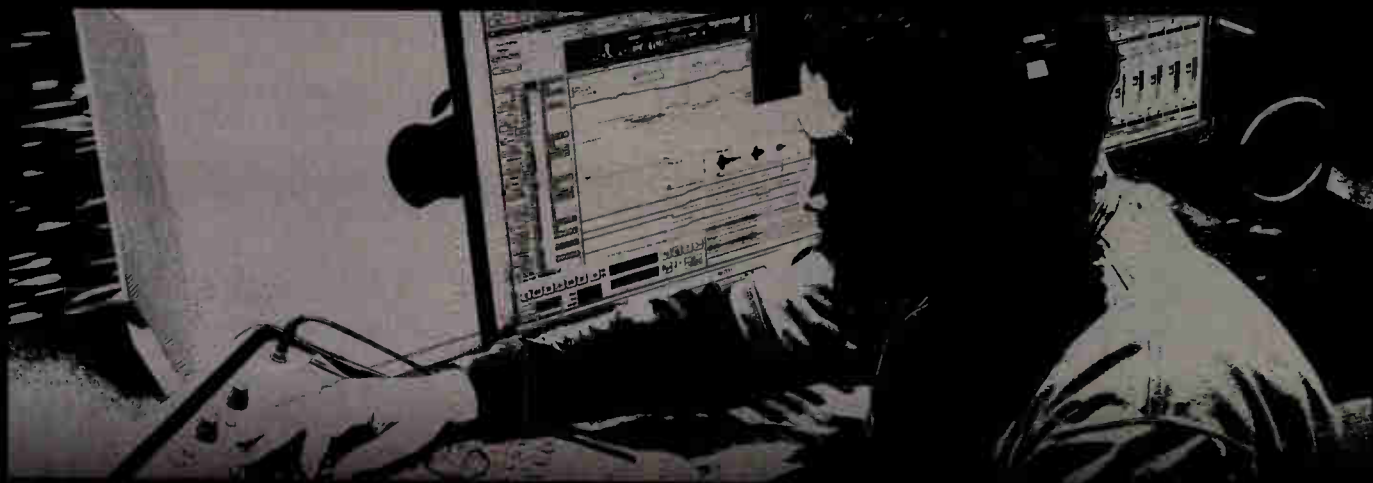
With DTS now a mandatory encoding for latest-generation HD DVD and Blu-ray media (as well as being implemented in virtually all current DVD players) there is a growing amount of multichannel audio material that needs to be converted from master WAV or similar media into DTS-compliant bitstreams, with all of the companion menus and ancillary data. And this is where MAS comes into its own.

The application's DTS-HD Encoder is particularly versatile. It handles a

DTS continues on page 82 ►



Audio Suite (SAS), is a limited version of MAS. Consisting of the DTS-HD Encoder, SoundCode DTS StreamPlayer plug-in and



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myriad of formats, including DTS-HD Master Audio, a lossless "bit-for-bit" variable bit rate with an embedded core of DTS Digital Surround constant bit rate up to 1,509 kbps, plus DTS-HD High Resolution Audio, constant bit rate from 2,046 kbps up to 5,760 kbps (Blu-ray Disc) or 3,018 kbps (HD DVD) kbps, plus DTS Digital Surround constant bit rate up to 1,509 kbps for DVD-V and DVD-A,

standard time code frame rates from 23.976 to 29.97/30, drop and non-drop, can be sourced.

Version 1.1 of DTS-HD Master Audio Suite ships with the SoundCode DTS-HD StreamPlayer, a decoder that includes both RTAS (real-time) and Audio Suite plug-ins compatible with Pro Tools|HD and LE systems on both Mac and Windows XP and other

PC-based DAWs.

The SoundCode StreamPlayer decodes and plays back standard .dtshd and .cpt files locked to the Pro Tools transport, with time code display; usefully, when synchronized to QuickTime or external video players, it serves as a lip-sync quality control platform prior to disc authoring.

Other features include integral level meters, downmix monitoring of 5.1 and stereo down mixes, DTS Core monitoring and 7.1-channel loudspeaker remapping that supports multiple speaker configurations. With the launch of Version 1.5 of Master Audio Suite and Surround Audio Suite, both applications will ship with a newly developed, stand-alone version of the DTS-HD StreamPlayer that uses Core Audio for Mac Intel and ASIO on the Windows XP platform (see screenshot p. 80). The SoundCode version will also still be available.

ENHANCED WORKFLOW FOR BATCH ENCODING AND SOUNDTRACK ASSEMBLY

As will be appreciated, Master Audio Suite offers a great deal of flexibility. Its three key components — DTS-HD Encoder, SoundCode DTS-HD StreamPlayer and DTS-HD StreamTools — can be used individually or collectively, dependent upon your needs, to prepare a range of multichannel products. The screens and menus are well laid-out and make it extremely easy to navigate through what, for most people, can be a somewhat bewildering experience, as .wav and similar files are assembled for encoding into com-

plex multiplexed bitstreams that then get delivered to the mastering application and burnt onto a master CD/DVD or transferred to a manufacturing plant. The SoundCode DTS-HD StreamPlayer decoder plug-in for ProTools provides intuitive DTS stream playback plus confidence monitoring, while StreamTools comprises a well-balanced tool set for encoded stream editing, verification and bitstream management.

It soon becomes obvious as the user accesses component files and selects the encoded-file format(s) to be prepared with MAS that a great deal of thought has gone into the interface. The application literally bristles with neat features, such as audio file drag-and-drop, dynamic parameter settings based on optical disc- and stream-type selections, plus an intuitive fader-based downmix panel.

The list of bitstream formats and combinations that MAS can handle is truly prodigious. I might be tempted to offer that there are few, if any, channel-count and data-rate configurations that the application cannot handle. Master Audio Suite is an ideal choice for all DTS-HD encoding formats, ranging from audio streams for Blu-ray Disc and HD DVD,

| DTS-HD ADVANTAGES:

1. Automatically Backward Compatible:

A backward-compatible DTS Digital Surround core that plays on any DTS decoder is always present inside every Blu-ray Disc Primary Audio or HD DVD Main Audio bitstream. That means a receiver you purchased five years ago with just regular DTS decoding can still play back the 1,509 kbps audio.

2. More Efficient Use of Disc Space:

The DTS-HD Encoder provides for high data-rate compression efficiency and eliminates the need to encode redundant backward-compatible components as required by other audio coding schemes. The result is greater efficiency and less disc space needed with a single DTS-HD Master Audio sound file.

3. Flexibility to Fit Any Bit-Budget:

The DTS-HD Master Audio Suite provides more encoding options to maximize audio quality within the allocated audio space on any type of disc project. DTS-HD supports all mandatory and optional audio formats for Blu-ray Disc Primary Audio, HD DVD Main Audio, HD DVD Sub Audio and Blu-ray Disc Secondary Audio stream types.



1,234 kbps for Music CDs and Sub/Secondary Audio low bit rate streams.

And there are dramatic advantages to be gained from DTS-HD, since the technique encodes bit-for-bit and carries legacy core and down mixes — all in one data stream that requires a single encode pass. The bitstream is also highly efficient, since the non-redundant encoding scheme produces smaller file sizes and lower bit rates. As is to be expected, MAS supports up to 6.1-channel data streams for DTS Surround Music Discs, up to 6.1-channel for DVD-V/DVD-A and up to 7.1-channel for primary/main stream for Blu-ray Disc and HD DVD. A range of user-selectable down mixes is also provided, including 7.1-to-5.1 and 5.1-to-2.0 channels. A range of sample rates are available, including 44.1/88.2 kHz for DTS Surround Music CD (noting that DTS Music CD Encoding is only available in the pending V1.5 release, and that stereo-only is supported at the higher sample rate), 48/96 kHz for DVD-V and DVD-A, 48/96/192 kHz for HD DVD and Blu-ray Disc, plus 12/24/48 kHz for Secondary/Sub-audio delivered on HD DVD and Blu-ray Disc (48 kHz only). Finally,

including constant bit rate (CBR) that can include up to 7.1 channels at sample rates from 48 to 96 kHz and a variable bit rate (VBR), bit-for-bit stream that can include up to 7.1 channels at sample rates from 48 to 192 kHz, with backward-compatible Digital Surround 5.1/6.1-channel core at a bitrate between 768 and 1,509 kbps. Not overlooking, of course, DTS Digital Surround encoding from mono at 192 kbps (BD only) to 6.1 channels (DTS-ES) at 1,509 kbps at a 48 kHz sample rate, plus 96/24 encoding of 5.1 channels at a fixed bitrate of 1,509 kbps.

Let's not forget additional content for Blu-ray Disc Java and HD DVD HDi discs. Usefully, MAS seamlessly integrates sub-audio and secondary audio that enable new features, such as Internet downloading and playback of interactive audio. (Sub/secondary audio can also be overlaid on a title without interrupting playback to provide such features as picture-in-picture, multiple-menu audio, pop-up windows and so on.)

ENCODE QUEUE: CHUGGING AWAY UNATTENDED

In the practical world, it makes sense to gather our digital file elements, check and double-check the encoding parameters,

and then let the software batch process the required bitstreams. Master Audio Suite sets this up very quickly and then chugs away unattended. The Encode Queue section acts, in DTS jargon, as "an information and priority management hub for multiple encode jobs" while displaying pending, in-progress and completed sessions ... couldn't be easier. You can also dynamically adjust the priority of each encoding session by moving it up or down in the queue using a couple of nudge buttons. An encode completion log file is also generated for client billing and verification. And, given that we live in the real world, MAS features a comprehensive

| PRICES:

Master Audio Suite: \$1,495 list

Surround Audio Suite: \$795 list

With the imminent release of Master Audio Suite V1.5, DTS unveiled a price reduction on Surround Audio Suite to \$249, and has announced a special pricing program for educational facilities.

diagnostics function. A status window keeps the user informed of operations, menu selections, successful encodes, as

well as any errors. An encoding session can be cancelled at any time, and the software set to generate a log of key parameters. A particularly neat feature is the program's downmix panel, which dramatically streamlines the production of dedicated 7.1-to-5.1, and 5.1-to-stereo down mixes with the use of faders. MAS incorporates a set of common default downmix settings, with phase-reverse switches.

CONCLUSION

All in all, the DTS-HD Master Audio Suite comprise a set of affordable, well thought-out applications, plug-ins and utilities for streamlining the encoding of multichannel DTS-encoded bitstreams for a wide range of current and emergent optical media.

DTS | 818-706-3525 | www.dts.com

Acknowledgements: My sincere thanks to Ronny Katz, Senior Manager, Professional Audio, for providing access to DTS-HD Master Audio Suite and Surround Audio Suite at the firm's Agoura Hills, CA, corporate headquarters.

Mel Lambert is principal of Media&Marketing, a Los Angeles-based consulting service for the professional audio industry.



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by Alan Silverman

Craftsmen of the Craft



Professional mastering has become much more important to musicians producing their own records, as sessions are increasingly recorded in residential space and engineered by those whose first calling isn't engineering. The good news is that excellent mastering is more affordable than ever. The bad news is, with so many mastering shingles hung out today, terrible mastering is more affordable than ever, too.

In my experience, the number of new clients coming in with examples of their music badly "mastered" has been eyebrow-raising as of late. Much of the damage seems to be due to what I call BBSS Syndrome: boom it, bright it, smash it and ship it. Mastering doesn't always work magic, but when it does it's usually executed with balance and finesse, not with extremes. Good mastering will bring out the energy, impact and emotion of a track without resorting to overbearing processing.

Mastering is only a craft in support of an art, but there's an art to doing it well. Hand the same mix to a group of experienced, well-equipped, full-time mastering engineers, and

able to nearly everyone, you can practically have your pick of the best mastering houses from all across the world.

| FIND THE RIGHT MASTERING ENGINEER

How do you select and work with a mastering pro when you're just starting out? Word of mouth still seems to be the most effective way of finding the right person. Someone with an extensive major-label track record can be impressive — it's undeniably a sign of skill and accomplishment — but it may also come with a higher price tag and the downside of less available time for your project.

It's not always a good idea to put too much stock in a genre-based selection, either; a talented engineer can work wonders with any music. Good sound is good sound whether it's a band, orchestra or Balinese temple bell.

Mastering samples are a guide, but unless you've heard the unmastered sources, you can't really know what was done. No matter how you choose, there will always be an element of uncertainty.

If you're new to record production, how do you get the best results in your first mastering situation when you don't have a history with the engineer or with the craft? If you're attending the session and have not worked in the room before, keep in mind that no matter how great the monitoring system is, you will not be familiar with it. You won't know what you've got until you've brought it back to your own listening environment(s). To cover yourself, establish a clear, out-front agreement with the mastering house as to exactly what their policy is for revisions. This is especially important for unattended and online mastering. Most houses will do reasonable revisions as a courtesy or for a nominal fee. But what is reasonable? Let's just say that remixing half the tracks and expecting revisions at no extra charge probably isn't. Mutual trust, good communication and common sense will rule here.

| MAKE THE MASTER WORK FOR YOU

One trend cropping up lately is that clients new to mastering actually seem to be a little too deferential to their mastering engineers.

Maybe it's because of all the hype that surrounds mastering and mastering gear, or maybe it's the connotation of the word "master." That word should always refer to the result, not the person running the gear!

Trust your instincts. If your ref CD from the mastering house somehow feels wrong — even if you're not sure why — it probably is wrong. One test that's important to have for every mastered project — even those from the best pros — is a level-matched comparison to the original unmastered mix. By "level-matched," I mean perceived level, not the setting on a knob. Poor EQ will show up right way once the before and after levels are matched. If the mix sounds better than the master, it's time for a do-over. If nothing else, this test can teach a valuable lesson of the compromises the record industry's "loudness wars" have forced upon musicians.

| GREAT EXPECTATIONS

The flip side of the coin is managing your expectations. With so many great-sounding recordings around, there is the hope that the right mastering engineer will make your record sound just as good as the best, but every mix has its own DNA that determines just what kind of creature it can be. Good mastering will make it sound like an ideal version of itself ... not something else. In my early years as a mixer I remember actually being disappointed during my first-time experience with a high-profile mastering engineer when he told me he didn't think the mix needed anything. It didn't sound like my favorite records of the time; it was what it was, and the engineer made the right call in not trying to make it something it was not. At my request, he added some presence EQ anyway; but, after listening at home, the artist and I agreed it was better without EQ. Knowing what not to do is one of the most valuable skills of a good engineer.

Some of the best job satisfaction in mastering comes with that last twist or click when a mix opens up to a new intensity, emotion or beauty that lifts you out of your chair. That's when the door to the safe opens, revealing the goods. It doesn't happen with every track, but if it happens enough, you're in the right job. Of course, that "lift" could be self-delusional. You need to hear from the project's principals to know if you've really nailed it. The prize is to come up with a version that lets them kick back and enjoy their project like never before.

Alan Silverman is the owner of Arf! Mastering in New York City and a multiple Grammy Award nominee, including for Album of the Year.

Lavry Engineering Gold AD122-96 MKIII A/D converter, a Silverman mastering mainstay.

the differences in mastered versions will range from subtle to substantial, yet one version may feel so right as to make the others seem off the mark. The mastering engineers who get the latter result are most often the ones we hear and read about.

Mastering can sometimes be the only opportunity for a musician-recorded project to benefit from the touch of an experienced professional engineer. Many mastering engineers have done and heard so much in their careers — often with having backgrounds in tracking, mixing, production and performing — that they are fantastic resources for people starting out in record production. Now that high-speed Internet connections are readily avail-

| SUMMARY

I felt it was important to document some of the more common troubles associated with the Duende, but I should note that there are many users who haven't experienced any complications. There are also those who could never resolve their problems and ended up parting ways with the Duende. My experience with SSL via the web support portal was favorable; the company showed its desire to work with users to get problems resolved, though some of the problems seem to be across the board, regardless of platform or host applications.

Here's the bottom line: You won't find a better DAW EQ and dynamic processing investment than the Duende, providing that it plays well with the rest of your setup. Thirty-two channels of vintage SSL in a single-space rack unit makes my ears smile, and, if it plays well with your system, you will be smiling, too. However, I strongly recommend a careful look at the latest online Duende support documentation and, if possible, a test drive with the

| EARLY REPORTS ON DRIVER 2.1

Since I completed the review of Duende using driver 2.0, SSL has released 2.1 and the reports have been positive. New features include linking two units together (at 44.1/48 kHz it is recommended that each unit is on its own FireWire buss; for 96 kHz, this is required).

I still must use stereo plug-ins when inserting on a channel in SONAR 6.x, as mono is not compatible and produces a ghosting effect. In this instance, this essentially cuts the capability in half. In the original Nuendo test, Duende worked well yet did require periodic reboots throughout a week-long mix session. Unfortunately, I have not had the opportunity to validate if this is better with the new drivers.

It should also be noted that LaCie drives do not play well with Duende, regardless of the FireWire buss, due to the FireWire chipsets used. As for audio interface compatibility, there have not been any major changes. However, SSL is working with many different manufacturers to work out compatibility issues, and the company updates the FAQ section of its site with any changes.

I still recommend trying the Duende on your system to verify compatibility before committing to purchase, although the new drivers do give reason to be cautiously optimistic.

—Dan Wothke


Duende on your own system.

Dan Wothke currently runs the gauntlet of all things media in his role as Media Director at Belmont Church in Nashville, TN. He is also intimately familiar with all things SSL from his years as chief audio technician at Masterfonics' the Tracking Room (featuring a SSL 9064J). Dan invites you to write him at dwothke@yahoo.com.

The manufacturer responds:

"We have put a lot of work into the new V2.1 software, both internally and working with third parties such as Apple to streamline the efficiency of new software. Reports from users are very favourable. We learned a great deal about FireWire, Macs and PCI busses in this development process. This information has been added to our web FAQs to help better support our user base for the future."


—Niall Feldman, SSL director of product management



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Masterfonics, Nashville

This legendary mastering facility moves forward with experienced leadership and renewed energy.

by **Strother Bullins**



From left: Vocalist Inga Brittingham, Cinderella Bassist Eric Brittingham and Tommy Dorsey at Masterfonics

Masterfonics has long been a name associated with prolific excellence in the Nashville recording scene. Masterfonics began as a mastering facility, expanded to the multi-building tracking, mixing and mastering empire of Music City, and — as it currently stands — has gone back to simply being a great place for mastering by consummate professionals with great ears and renowned mastering suites.

"I've been at Masterfonics for 13 years," says Mastering Engineer Tommy Dorsey, one of the two principals at Masterfonics. "[Mastering Engineer] Jonathan [Russell] has been here for six years, and seven at Georgetown [Masters] before he came here. We've really applied ourselves to a craft and business we love. Our clients like the environment we work in and the results are proven. When [acoustician] Tom Hidley designed these rooms, he really hit the nail on the head. This place is a good balance of many things: productivity, attitude, technology and comfort. It has been a great environment for music for many, many years."

Masterfonics and Georgetown Masters were top mastering facilities during the early 1990s "peak years" in the Nashville music industry. Dorsey and Russell started as apprentices during that period and steadily built industry respect as star players in highly regarded positions. Dorsey was the valued apprentice to Mastering Engineer Glenn Meadows and fellow Mastering Engineer Benny Quinn at Masterfonics; a block away, Russell was Mastering Engineer Denny Purcell's right hand man at Georgetown Masters. As Dorsey and Russell began gradually building their own independent clientele, their personal inclinations toward newer

tastes in music positioned them to be the sought-after talent for the increasingly modern styles emerging in the Nashville.

The opportunity to work as staff mastering engineers at Masterfonics has helped to fuel their own personal high standards of excellence and raised the bar for Masterfonics in the process. Dorsey and Russell have experienced, first hand, the successes and the hard times of these two industry powerhouses, and they bring an amazingly high level of expertise to the table as individuals and as a team. As times have changed in the mastering business, Dorsey and Russell are in the right place as the Masterfonics Mastering team.

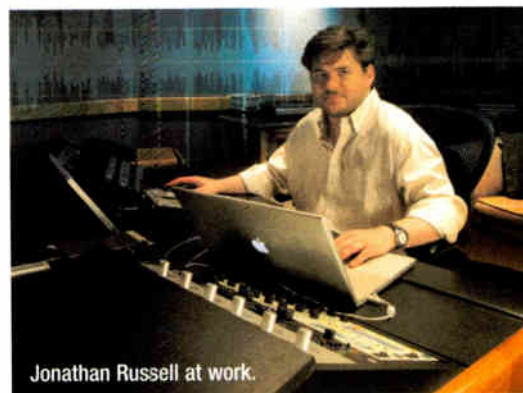
In the following interview, Dorsey and Russell share their viewpoints from a unique perspective at a legendary address on Music Row.

PAR: PLEASE EXPLAIN THE RECENT HAPPENINGS AT MASTERFONICS MASTERING.

Dorsey: "As you may be aware, Masterfonics Mastering, Masterfonics Recording Studios and the Masterfonics building have been in a major state of change over the past six months. With the exception of the mastering equipment, all gear from the studios at both 28 Music Square East and 1033 16th Avenue South building has been sold. Masterfonics Mastering will continue to call the Masterfonics building home; the rest of the building will eventually house a separate commercial recording studio. The mastering department is its own company operating under the name Masterfonics, now owned by the two Masterfonics mastering engineers: Jonathan Russell and myself. We purchased the mastering equipment and the Masterfonics name from the previous owners of the company, Voss Development, earlier this year.

"Masterfonics Mastering has continued to

operate throughout the liquidation process of the recording studios and the sale of the building, and we purchased their gear and the name at the outset of this process. The mastering department at Masterfonics had a very strong year in 2006 and continues to thrive during 2007."



Jonathan Russell at work.

PAR: PLEASE DESCRIBE SOME OF YOUR MOST INTERESTING CLIENTS AND MASTERING WORK AS OF LATE.

Dorsey: "Nashville is home to a thriving number of hip-hop production companies. Over the years I've had the opportunity to work on numerous hip-hop albums, singles and mixtapes. Since I have extensive experience mixing inside the box with DAWs, many of my clients choose to bring their mixes in multitrack format to the mastering session. On some sessions, the engineer will provide me with simplified stems, and on other sessions I'll get the whole expanded multitrack with active plug-ins, pre-mastering on the stereo bus and everything. Sometimes they'll bring their rig; other times we'll use mine. I really enjoy getting to help refine the mixes as part of the mastering process."

Russell: "While recently mastering Leon Russell's *Angel in Disguise*, producers Mike Lawler and Russ Ragsdale wanted to retain the depth of their original mixes by keeping each stage of the process digital. It took a lot of critical listening to arrive at the ideal minimal signal chain and in the end, the results were great.

"On a recent release for Starbucks, country artist Rissi Palmer's debut release was mixed by Trina Shoemaker and Billy Whittington. The mixes were fantastic on their own merits, but were completely different from each other. Therefore, the batches of mixes required different signal paths to get the desired sound while maintaining continuity.

"Also, I'm currently working on a project by artist Bonnie Bramlett for producer Johnny Sandlin, who mixes to 15 IPS 1/4-inch Dolby. The project is kept in the analog domain for all processing until the final stages of mastering. Sandlin is not caught up in 'the level wars' — it's so nice to work with someone of Johnny's caliber who really knows what they want and how to get it."

Dorsey: "I've recently transferred more than a dozen analog quarter-inch album masters featuring several classic artists from the '60s and '70s for Capitol Records. I've done a lot of this type of work over the years and it is really a pleasure. It's exciting to capture these marvelous vintage sounds using popular technology of today, such as 88.2 kHz sampling rates and 24-bit Prism converters. The next step in the process was to create high-quality 16-bit masters with as little sonic compromise as possible, so I was asked to master some of the transferred albums. It's amazing how incredible these old analog masters sound when transferred into high-resolution digital. With no EQ or limiting, the sound coming off the digital is wonderful and actually quite competitive compared to the sometimes brick-wall-limited modern mixes we hear. The vintage mixes have a sense of sonic balance and continuity that is refreshing to go back and revisit."

PAR: DESCRIBE YOUR VIEWS ON CLIENT DEVELOPMENT AND REACHING INTO THE ESSENTIALLY GLOBAL AUDIO PRODUCTION BUSINESS OF TODAY.

Dorsey: "It probably goes without saying, but the Internet has drastically changed and improved our business. When we first started using the Internet to transfer song files, we were immediately taken by how convenient the process can be. It's easy to see how this flexibility has expanded our business. But the more this uploading and downloading becomes a part of our daily process, the more we realize how clients are seeing this as a way to facilitate production creativity.

"A musically exciting aspect of working on projects featuring various dance remixers is how they will be converging from different parts of the world within a short period of time. With the flexibility of the Internet, I can be getting mixes from various cities or even countries all in the same day, literally minutes after the mixes have been approved. All of the mixes may get mastered onto a single project/playlist.

"Jonathan has been working on numerous live album releases for the Dave Matthews Band over the past year. The live concerts are recorded in venues throughout the world. The multitracks are sent to Charlottesville, VA to be mixed, and then the mixes arrive in Nashville for mastering via the Internet using our secure FTP server. This is an ongoing process that occurs throughout the artist's tour. The Internet facilitates instant referencing and the approval process is convenient for all involved parties."

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Strother Bullins is the Features and Reviews Editor for Pro Audio Review.



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in minutes by plugging in a Cat5e cable from a Digital Snake stage unit to the 48-channel M-400, transporting 24-bit digital streams for none of the bulk or noise of analog snakes. Other impressive features include rapid recall of setups, 100 mm motorized and touch sensitive faders, an 800 x 480 color screen, digital patchbay, 16 aux/mon sends, 8 DCA's, eight mute groups and an intuitive interface. Link the M-400 to your PC to send/receive data and prepare channel setups/configurations before arriving at the venue.

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There may be no "I" in "TEAM," but there is one in DJ: the iDJ2, to be specific. And all eyes will be on you when this revolutionary DJ mixing console turns your iPod into two turntables and a crate of records — and the dance floor into a frenzy. Execute any number of crucial maneuvers from the Universal Dock: scratch, control pitch, create loops, cue songs,

key-lock, play two songs simultaneously and more, all through a full-color, high-resolution interface and balanced XLR/stereo RCA output. Additionally, iDJ2 allows you to hook up more iPods, thumb drives and external USB hard drives through external rear ports, analog equipment through line inputs, and the exclusive Crate feature keeps anticipated (non-DRM) music in a waiting list for spontaneous sets.

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No matter how much amplification you have, you can't actually move walls. But thanks to Fishman you can play with the acoustics in a room — meaning the acoustic instruments — and bowl folks over in the process with the way these pedals shape new tones without adversely affecting the signature sound of the instruments.

Available in Chorus, Delay (pictured) and Reverb models that marry the resonance of instrument and pedal in parallel, the AFX pedals are rugged all-metal construction

with eight effect presents, a level knob, two effect-adjustment controls and an all-analog, buffered bypass foot switch. Effects are of the 24-bit A/D/A, 32-bit processing variety for highest audio quality, and 1/4-inch stereo ins and outs. Input Gain switch and Clip light are standard. The pedals run on 9 V battery or wall-mount adaptor.

PRICE: \$289.95 each

CONTACT: Fishman | ☎ 978-988-9199 ➔ www.fishman.com

A-Designs is a-okay with FOH engineer Jason Raboin, who has used a pair of conveniently portable Audio Pacifica dual-channel solid-state mic preamps to add guarantee solid upper mid cut, presence and air without sibilance on tours for Joan Baez, the Cowboy Junkies and Modest Mouse. A REDDI tube direct-inject box has also proven positive for bass guitar.

Sennheiser has been making sure the Vans Warped Tour didn't sound misshapen, with the Sennheiser 935 being used by Paramore, the evolution 945 accompanying Coheed and Cambria, Red Jumpsuit Apparatus utilizing e 835 and e 604s, and Boys Like Girls trusting Sennheiser for reference-quality presentation.

Yamaha Commercial Audio

Systems recently supplied **NEXO** GEO T, D, S, Alphas and PS Series line arrays/fills (pictured) to allow the first Festival International de Jazz de Montreal to take place with lightweight rigs that provided amazing powerful clarity.



Yamaha also saw its new DSP5D Expander for the PM5D console embark on its first live use, accompanying progressive metal band Dream Theater. The DSP5D took the PM5D to 96 mono plus 16 stereo input channels, also offering high-quality internal effects all in an extremely small footprint.

Thanks to **Aviom**, the 40th anniversary of the "Summer of Love" was a summer of loving connectivity. The company's personal monitoring systems assisted the Trevor Lawrence, Jr. (drumming with Alicia Keys), Hall & Oates, Felix "D-Kat" Pollard (drumming with Taylor Hicks), drummer Danny Seraphine, the brass section touring with Michael Buble, and George Michael's band.



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QSC™

by Mel Lambert

Allen & Heath iLive Digital Mixing Console

The iLive live production console represents a dramatic departure for Allen & Heath. Until now the UK-based manufacturer has enjoyed a reputation as a purveyor of affordable analog designs;

iLive's all-digital topology not only offers the familiar system recall and reset plus remapping of on-surface faders, it also enables the system to connect to various units from the company's iDR Series of audio management tools. Combine that with EtherSound connectivity to such devices as Fostex NetCIRA audio distribution products via Cat-5, and audio professionals begin

routing to 32 mix outputs assignable as mono/stereo groups, mono/stereo internal effects, mono/stereo aux busses, mono/stereo monitor matrix and a user-programmable choice of primary mono, stereo/LR, LCR, LCR+, LR+Sub, LCR+Sub, LR+M and LCR+M. Patchable channel-direct outs also are available.

In addition to the user surface, MixRack can be controlled via Ethernet from a PC or a local controller using PL-Anet, Allen & Heath's proprietary control protocol for its iDR units. Maximum Cat5 cable length between the surface and engine is 330 feet,

double duty. The unique combination of Ethernet serial control between the control surface and processing rack, plus optional multi-channel EtherSound over Cat5 means that hardware can be located pretty much anywhere in the signal topology and then easily accessed and controlled from the surface. Once experienced, it's a way of working that definitely offers a host of operational advantages.

Also, it's possible to display the signal flow between the control surface, iDR64 DSP engine and other equipment located on the control network while using a laptop computer running Allen & Heath's proprietary iLive system manager application. A low-cost router switch also opens up interesting Wi-Fi potential.

The iLive control surface is available in four sizes and prices (listed as manufacturer's suggested retail): iLive-80 with 20 on-surface faders and four layers offering 80 assignable channel strips (\$30,000); iLive-112 with 28 faders/four layers/112 channel strips (\$50,000); iLive-144 with 36 faders/four layers/144 channel strips (\$55,000); and iLive-176 with 44 faders/four layers/176 channel strips (\$60,000).

By way of example, iLive-176's surface measures just 55 by 14 by 28 inches (W x H x D), and weighs 119 pounds (273 pounds in flight case). Each mixer configuration includes a rear-panel I/O rack that can hold up to four input and/or output modules in analog/digital formats. These

ports are very useful for connecting a variety of sources at the mix position, such as CD players and other devices

that might be used for music playback, as well as signal processing

units and recorder feeds and/or PA sends that need to reside at FOH or monitor-mix. An EtherSound link is required between the surface and rack to handle bidirectional digital audio transport.

While the smaller iLive-80 features a 32-in/16-out configuration (expandable to 64 x 32) and ships without a flight case and lacks EtherSound, the other three support up to 64 input sources routing to 32 assignable mix outputs via 16 DCA-format groups. A variety of I/O modules are available to accommodate analog and digital sources and outputs, including an eight-channel analog mic/line input module with XLR connectors; a eight-channel/dual-input analog module with Dual-Phoenix connectors; eight-channel digital input plus digital output modules offering three simultaneous formats (AES-EBU via XLRs, S/PDIF via RCAs and optical ports); an

iLIVE continues on page 92 ►



to see that iLive is more than a powerful live-production mixer; it can serve as a multidimensional system for controlling audio in a wide range of pro-audio applications.

THE "I" = iDR64 PROCESSING

The heart of the iLive's modular components is the iDR10 MixRack that houses an iDR64 digital audio processing engine. As the name suggests, MixRack provides 10 card slots for loading a user-choice of I/O modules, linking via standard Ethernet to the control surface. The iDR10 chassis can be located close to the stage or performance area to reduce mic/line cable runs. Via separate input and output virtual patch bays, the iDR64 engine currently provides a maximum of 64 full-function signal paths - with stereo-linked inputs -

dependent upon the cable used, according to Allen & Heath; fiber-optic Ethernet also is available. Other modules enable EtherSound digital audio distribution. The rack is shock mounted in a custom, touring-grade flight case; it can also be supplied with a back-up PSU for critical installations.

SYSTEM SETUP: ACCESSING MIX BUS CONFIGURATIONS AND SURFACE STRIP ASSIGNMENT MENUS

The touch-sensitive TFT screen streamlines the system set-up via a series of menus for assigning mix bus and surface strips. The supplied hardware can be arrayed across any combination of groups, aux sends, main outputs and matrix busses to develop the bus architecture needed for a dedicated FOH or stage-monitor console, for example, or one that does



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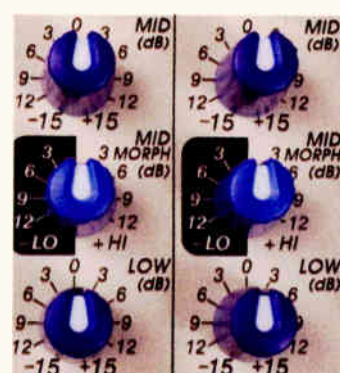
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eight-channel analog line-level output module with XLRs; and a double-width, 16-channel digital output module offering a pair of simultaneous ADAT connections plus a single iDR-format port.

Sample rate is fixed at 48 kHz (which means that the digital output for a CD player, for example, will need to be sample-rate converted). All analog-to-digital conversion is via 24-bit Delta Sigma circuitry; internal DSP uses 24-bit, 48-bit data highways. (System latency is said to be less than 2 mS from rack-in to rack-out, and under 2.25 mS from control surface inputs to outputs via EtherSound.)

Control Surface: Analog-style layout with full-color LCDs and touchscreen TFT display

Loaded with a maximum of 10 eight-channel I/O cards, iLive provides a total of 80-ins/80-outs; four eight-channel I/O cards within the control surface add 32 in/outs. During system setup, the assignable surface strips can be mapped as channels, mix masters, DCA masters, PAFL masters and even MIDI controllers. The control section is live at all times, and offers a very elegant one knob/function layout that is selectable to the target signal path using a single tap of a button adjacent to each channel strip.

The control section houses all the important processing adjustment knobs and indicators; from left to right, these comprise a Preamp, Insert, Gate, High-Pass Filter, Parametric EQ, Compressor and Limiter/De-

esser. Pressing any SEL key causes the corresponding page to be mapped to the companion touch-sensitive TFT display, which shows the DSP parameters in familiar frequency/level plots and input/output gain dynamics, etc.; it allows the displayed curves to be redrawn with the touch of a finger. Because of its reduced size, iLive-80 omits the on-surface Dynamics controls and maps these to the TFT.

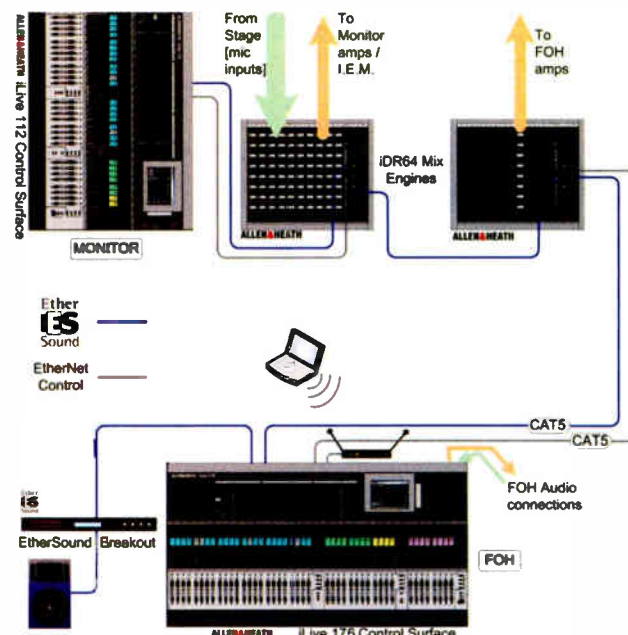
The Preamp controls the remote iLive mic/line amps with a peak-level indicator. Digital inputs also feature +/-24 dB gain trim. Insert allows send/receive ports to be assigned to pretty much anywhere in the signal path. The High Pass filter offers a center frequency from 20 - 400 Hz in order to eliminate LF rumble and pops, while the Gate — one per input channel — offers full parameter control with side-chain EQ. The signal level is displayed behind the threshold setting to help the user quickly find the gate point. The Parametric Equalizer comprises a four-band section with HF and LF bands selectable to shelf or bell response.

While it's totally feasible to adjust system settings from the control strip, the companion TFT display can be set to display conventional cut/boost across all four bands or as a 25-band third-octave graphic section on every mix channel, with full touchscreen adjustment of each band from 63 Hz to 16 kHz (useful for setting up group output and stage-monitor monitor feeds, for example). Cut/boost settings can be adjusted on the TFT screen or, for more accurate control, flipped to on-surface faders. To prevent inevitable confusion, the Fader view shows all bands, as well as Mix master levels. Frequencies displayed on the individual channel-strip LCDs change color to indicate EQ mode — a neat touch.

A Compressor is provided on each input and mix master channel; again, signal level is displayed behind the threshold setting control, with a graph and histogram of gain reduction plotted into the TFT screen. The

Limiter/De-esser adds overall gain reduction to each input/mix master; the de-esser mode can be used in conjunction with the compressor on troublesome vocal channels. Delay, accessed from a channel thumbnail on the TFT screen, is provided on all input/mix

Typical Front of House and Monitors application



ILIVE FIRMWARE VERSION 1.1 ADDS ENHANCED FUNCTIONALITY

It must be conceded: early manifestations of the Allen & Heath iLive console lacked a number of basic features. Responding to user feedback, the Brits spent the past year refining the newly-released V1.1 operating software, which adds a programmable I/O router and other features to provide the following enhanced functionality:

Input/Output Patchbays; Input Channel Inserts; Channel direct outputs; EtherSound routing; Copy/Paste/Reset editing; FX tap tempo function.

V1.2 software is scheduled for availability by end of the year. Additional features include linking of two DSP Engines to provide 128 inputs by 32 busses, bus inserts, FOH/stage-monitor input splitting, and additional effects processors. More added features are to be determined.

channels for microphone and speaker alignment, etc., with parameters displayed as time or distance.

But built-in dynamics and EQ DSP is just the beginning, since iLive also features two FX engines that can be assigned to channel strips without using up input channels, thereby providing the equivalent of 68 sources feeding a mix. A total of four more FX engines can be configured via a DSP processing option.

The iLive's PFL/AFL signal-monitoring system can best be described as comprehensive. Users can select input and mix PFL or AFL in-place, PFL override AFL (particularly useful for setting up stage monitors), auto-cancel or additive mode, clear all and delay, plus dual AFL masters for personal monitoring and wedges. A user preference links the PAFL function with channel SEL and/or MIX keys. Headphone amps are also featured at both the MixRack and control surface. A neat feature: operating a PAFL key causes an LED on the corresponding MixRack input/output module to light, so that a stage tech can see instantly which signal source is being checked.

In addition to 16 main channel/mix LED

meters and dedicated PAFL metering (dedicated meters within each channel strip, above the highly legible digital scribble strip/ID, pan control and control buttons), the TFT screen offers a range of factory and/or user-defined custom metering displays, including gain reduction for dynamics. A peak indication across fader layers draws attention to level overload, even if that layer is currently buried. DCA Groups are available with contributions from input channels and mix masters. Surface strips may be configured as DCA masters anywhere on the surface; each one can be named and color-coded for instant access.

As well as user-assignable, five-character channel names, users can set each channel to one of seven colors for enhanced ID at a glance; target channels can also be held back or frozen on the current setup rather than toggling with layers — useful if you prefer to have lead vocals under your fingers, for example, as the other assignable source are accessed via layer switching.

Systems parameters can be stored in a number of memories arranged through a built-in Scene Manager as Libraries, Scenes of selectable settings and Shows — the latter a configuration of current settings, all scenes and libraries. USB ports enables easy off-site storage and recall.

For installations that might be used 24/7 by a number of operators for different productions, assignable password-protected permissions are available for up to eight operators. Scenes can be named, descriptions added using an on-screen keypad, and their contents viewed and edited using a series of touchscreen menus. It's possible to save a selection of input channels — including names, color coding and parameters — that might be appropriate for a band's settings during a music festival, for example. Dedicated keys select scene recall, with the present and next scene being displayed at all times on the TFT screen.

INTUITIVE, WELL-PRICED TECHNOLOGY

All in all, Allen & Heath's new iLive Digital Live-Production Console is a remarkable piece of intuitive technology. The combination of Ethernet control of the remotely located DSP Engine from the well laid-out user surface or PC and EtherSound-based audio transfer is extremely powerful. The console system has application in conventional live environments — front-of-house with stage monitor and/or as a dedicated monitor desk with shared processing resources — as well as audio-for-video and corporate presentation setups. For its price, I don't think you'd find a well-made, great sounding alternative with such a fine pedigree.

The console's DSP Engine ensures minimum latency with processing available on all channels all the time — no calling up resources and effectively rebuilding the topology each time the user changes a system topology. The abundance of on-board DSP and special effects processing eliminates the need for expensive and bulky outboard racks, and ensures consistent sound from show to show. The iLive control surface retains the familiarity of a traditional analog console with its one-knob-per-function dedicated rotary controls and switches, together with a display that provides a quick overview of all settings.

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My sincere thanks to Tim Schaeffer, Director of Sales-Pro Audio with American Music & Sound, A&H's North American distributor, and the ever-affable Dave Lewty, Allen & Heath Product Manager.

Mel Lambert is principal of Media&Marketing, a Los Angeles-based consulting service for the professional audio industry.

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Noise - En (nV/√Hz)

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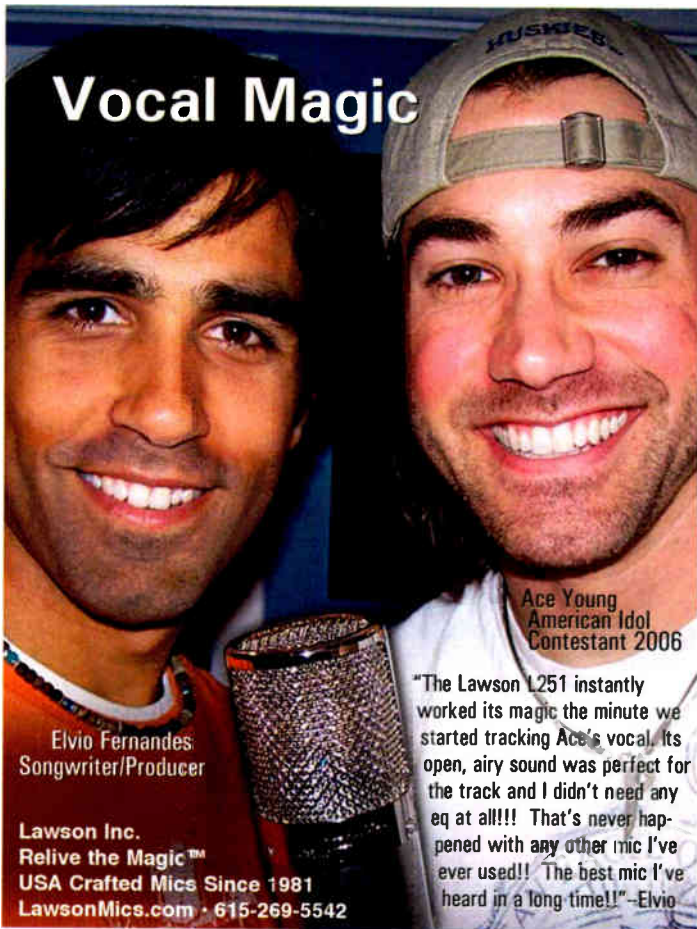
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by Strother Bullins

Sound of Black, Wet of Sweat, Smell of Jäger



stage from scratch. Then they left it to me to do it. The first few ideas were done in Microsoft Paint. From that came a builder in Detroit, who, from the drawings in Paint, built the stage you see out there."

For the five-person Jäger crew, a typical day on Ozzfest started with the hydraulic leveling of the stage with a generator. From there, proper

On one incredibly hot morning this late August, I stood at my closet door, debating what shirt to wear for my day at the Verizon Wireless Ampitheatre just outside of Charlotte, NC.

Should I dress to be "cool," or dress to actually *be cool*? In this case, dressing literally cool would mean some light-colored shirt, as I would spend an entire afternoon in the blazing heat at the Jägermeister Mobile Stage for the Ozzfest 2007 US tour.

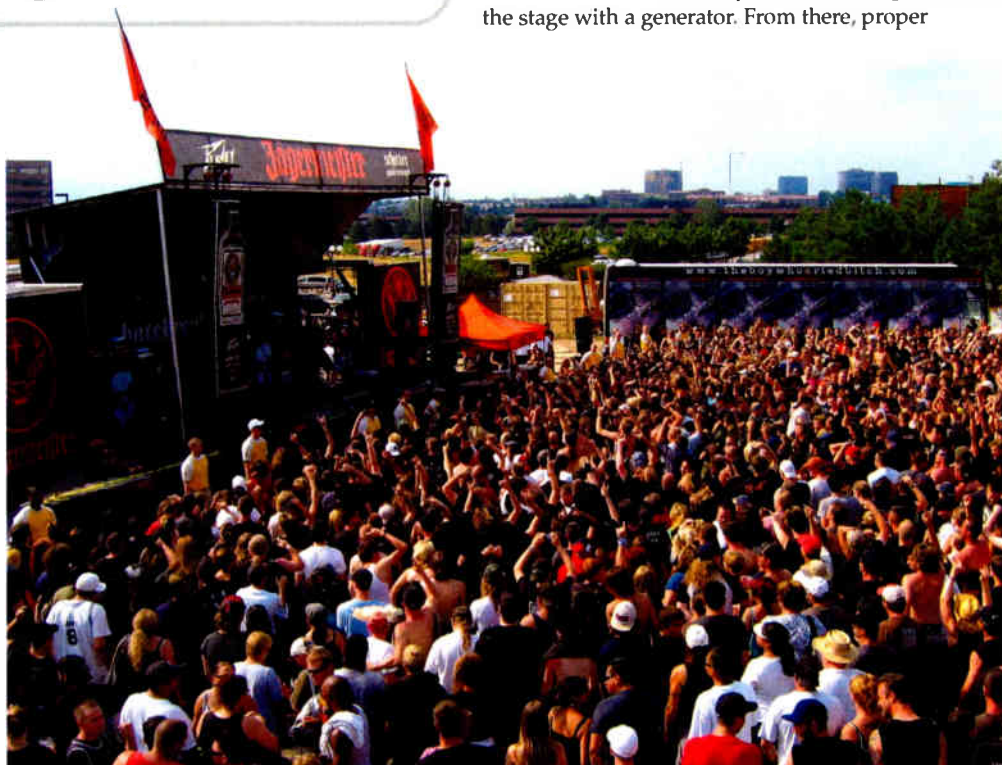
After further consideration of the afternoon's intense musical lineup — Hatebreed, Behemoth, DevilDriver, Nile, Ankla, The Showdown, 3 Inches Of Blood, Daath, and In This Moment, Chthonic and Black Tide — dressing "cool" would naturally mean a black shirt of some sort (or maybe no shirt, though I do like wearing shirts). Black shorts, black socks (if any), and a pair of dirty black Chucks would probably be a nice touch. But, boy was it going to be hot. Which way did I go? Was it even really a contest? I chose "cool." After all, in the world of extreme metal — with its rhythmic, intricate and cacophonous music, the physical anarchy of circle pits, and constant messages of individuality — an official uniform still exists. And it's black ... not sensible off-white linen.

All joking aside, this was an extremely pleasant day in that sweaty, thirsty, tired, rock & roll kind of way. All musicians played their black little hearts out and gave the main stage of Ozzfest — later to feature Lordi, Static X, Lamb of God and Mr. Osbourne, himself — both a suitable introduction and a real challenge.

However, in my opinion, most impressive was the afternoon's behind-the-scenes support, thanks to the talented, unique crew for the Jägermeister Mobile Stage. For the touring, staging and live audio savvy, this all-enclosed, rolling live venue is a show unto itself.

ALL THE WORLD'S A STAGE

Jägermeister's unique touring concept



The Jägermeister Mobile Stage unfurled at Ozzfest (and on the road, right)

started as a simple idea, and — with the help of Mobile Stage Road Manager and Jäger company man Jack Carson — it developed into the true metal monster of Ozzfest 2007.

"I started on this project about two years ago," explains Carson. "[The company] had a concept of a fold-out stage that would be towed behind a Suburban. The first problem with that was the people with the idea weren't production savvy; they just wanted to go with it. I explained that they needed a straight truck with production, lights, scrims, etc.

"We drew a few designs to make the stage design look bigger. Then I was knocking around the idea of towing the stage with a Freightliner truck with a 16-foot garage in the back — basically the length of a tour bus. I literally said, 'Why don't we put the stage in this other trailer, then tow it and all the gear with one rig that has bunks on it.' From there, I suggested a design and the idea to build a

things are put in the proper places as it opens, sound and audio equipment is downloaded, all gear is placed, and "in three to four hours," offers Mobile Stage Manager/FOH Engineer Dave Summers, the stage is set and ready to rock. "It's a completely self-enclosed unit. It doesn't need anything else to run. It's all there: crews, generator, and absolutely everything else."

HEAVY SOUND WITH A LIGHTER PA

Ozzfest 2007 Second Stage sponsor Jägermeister spec'ed one heck of a PA — one that effectively kept this day's 17,000-plus metalheads content, energetic and kicking up dust. A Peavey Versarray line array sound system with 207,000 watts of Crest Audio power delivered the goods. System highlights include 24 flown Versarray 112 enclosures, 12 Versarray 218 subwoofers, eight Peavey VSX



26 loudspeaker management systems, and 48 Crest Audio power amplifiers (18 CD 3000 amps to power the 12-inch speakers in the Versarray 112 enclosures and QW Series monitors; 12 CD 2000 amps for the Versarray's dual-ribbon drivers; and 18 Pro 9200 amps for the Versarray 218 subs).

THE METAL MIX

"For Ozzfest, we decided what would be the best cabinets for packing, displacement sound, coverage, and the easiest way for doing all of that in the least amount of space," offers Summers. "We're covering this large area and large audience very well, running at 107dB A-weight/118dB C-weight for six hours straight. We ended up using a lot of high-powered amps in the Crest Audio Pro series to have the least amount of racks. For sure, the Crest Audio Pro Series 9200/8200 amplifiers — mono bridged 4 ohms, putting 6000 watts to each subwoofer — have been great. Running the whole system at 4 ohms is a comfortable place to have it."

There was some retooling of the standard Jäger Mobile Stage for Ozzfest. "There were specific needs, due to the Ozzfest rider, that we had to beef up," recalls Summers.

However, that's not to say that the normal rig is anything to sneeze at. "Our standard setup was used for a large show by Vince Neil of Mötley Crüe," offers Carson. "He was very vocal about its top-notch sound and lights, and offered that he'd be happy to play on it at

any time ... and that's our regulatory rig that was *not* picked up for Ozzfest."

Elsewhere, Crest Audio mixing desks resided at monitors, and at FOH a Yamaha PM5D was deemed necessary to handle rapid-fire set changes.

The fact that we had five-minute changeovers necessitated that," explains Summers, a firm, yet flexible analog mixer fan. "There was no way that any engineer could do this without a digital console. We'd need four consoles, a matrix mixer, and guys that had 10 minutes to redial the consoles every day. Nothing would've ever been the same. The decision was finally made to go digital because you could push a button and recall from the last show. It's that simple. I love analog console sound, but digital is the way the world will be. The PM5D is great. Its functionality is fast. Sure, I'd still use a Crest LMX or a Midas Heritage 3000, but do I understand that we'll all be using digital in a year from now? Yes."

COOL IN THE HOT CAROLINA SUN

Regarding other beloved gear, Summers can't stop gushing about Rane compressors and gates. "I totally love the Rane stuff," he declares. "When we run our analog stuff, I use the Rane C4 compressors and G4 gates. I love their digital EQs, their functionality and the ability to make up gain, and how precise the filters are — because it is a digital filter, not an analog filter. They're the best."

Summer temperatures were obviously a major consideration in equipping the Jäger stage for such an intense trek, and the Peavey line array was well suited for the gig. Versarray 112 line array enclosures utilize dual ribbon drivers with a patent-pending Ram Air Cooling design, which, according to the manufacturer, is "a dissipation process that focuses the airflow through the voice coil, resulting in minimal heat buildup and power compression." Further, Peavey's UniVent technology is built to cool the Versarray 218 subwoofers by pumping air through the enclosure while in action.

PULLING IT OFF

The Verizon Wireless Ampitheatre show was an unmistakable success, and so were the dates of Ozzfest 2007 in the rearview mirror, happily offers Carson.

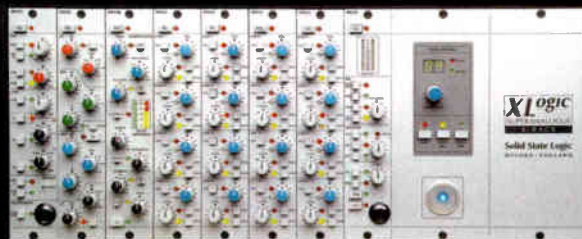
"A lot of people wanted to say that the stage wouldn't be big enough. But you can see that it is; just look on the faces in the audience. Jägermeister and Peavey really pulled off something special to handle Ozzfest. Sending out a brand new mobile stage — now with only 25,000 miles on it — on a tour like this is a task. The stage has held up very well, the PA has held up well, and we couldn't be any happier."

I agreed, and added that I couldn't have been any sweatier. No one laughed. But at least I was wearing black.

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by Mike Rivers

Core Sound Mic2496 Portable Mic Preamp

This PDA-sized two-channel mic preamp and A/D converter provides convenient location sound recording via S/PDIF input.

Location recording media has changed over the years, but front-end preferences for portable recording are still as follows: compact size, battery operation and very good built-in microphones and/or a respectable mic preamp. Most modern flash memory recorders make excellent recordings when fed high quality source signal, but their built-in mics or mic inputs often leave much to be desired. To make a good microphone recording, it's often necessary to carry along an outboard preamp.

Before flash memory recorders were commonplace, Core Sound — a company dedicated to the needs of the location recording enthusiast — developed the PDAudio card, which turned a PDA (Personal Digital Assistant) into an audio recorder. The Mic2496, developed shortly thereafter, provided an analog front end to the PDAudio system.

FEATURES

The Mic2496 is a two-channel mic preamp and A/D converter. Output is S/PDIF TOSLink optical and coaxial (the "coaxial" output is

actually an 1/8-inch phone jack). There's no analog output, limiting its utility to recorders with a digital input. Inputs are via a single 5-pin Mini-XLR type connector (Switchcraft TA5 Series). For this review, Core Sound provided a mic input breakout cable to a pair of XLRs and a digital output cable with a conventional male RCA plug. Power is either from a 9-volt battery or 7 - 14 V DC external supply.

The A/D converter is 24 bit, providing sample rates of 44.1, 48, 88.2 and 96 kHz, selectable by two toggle switches on the outside of the case (one selecting between 44.1 and 48 kHz, the other doubling the rate). Alternate sample rates of 32/64 kHz and 176.4/192 kHz can be selected by internal jumpers. Other internal jumpers change the microphone power to 9 V for use with Core Sound's binaural microphone system, employing DPA 4060 or 4061 mics, and to raise the coaxial S/PDIF output level for compatibility with Sony Walkman-style TC-D3/7/8 recorders.

At maximum gain, 43 dBu (6 mV) at the input produces 0 dBFS at the digital output. At this gain setting and with the input terminated with a 150 ohm resistor, quiescent noise is around -73 dBFS: a respectably low level. At minimum gain, -2 dBu in yields full-scale output. Input impedance is a practical 1.6 kilohms. A toggle switch controls the phantom power.

Level control is by dual concentric pots. With no clutch tying them together, tweaking the record level can be a bit fiddly. Signal level indication is by a pair of LEDs, -12 dBFS "activity" (green) and -1.5 dBFS "clip" (red). Other LEDs indicate phantom power and low battery. The preamp is packaged in a solid 3- x 5- x 1 3/8-inch black anodized aluminum case, and it weighs a mere 10 ounces sans battery.

Core Sound estimates five hours operation on a standard 9-volt alkaline battery with phantom power off, and four or more hours when using phantom power (depending on the microphone's current requirement). The review unit had no power switch, but current models do include a recessed power switch.

IN USE

I used the Mic2496 with a variety of new and old dynamic and condenser mics, and concluded that it sounds much like other non-exotic contemporary transformerless preamps. It doesn't add any particular character, but it's clean and reasonably quiet. If you're familiar with Mackie XDR preamps, you know what the Mic2496 sounds like.

With only two level indicators displaying essentially a 12 dB working dynamic range, it's best to use the Mic2496 with a recorder that has decent meter resolution. The on-screen meters of the companion Core Sound PDA are fine, but a Nomad Jukebox 3's meters leave a lot to the imagination. I'd prefer the lower level LED indicating between -16 and -20 dBFS: typical "analog 0 VU" level on a digital recorder.



I didn't really hear the effect of truncation on casual recordings when feeding the Mic2496's 24-bit optical output to the 16-bit input of the Jukebox 3. More serious recordists will likely use it with a 24-bit recorder. While comparing the Mic2496 with various preamps using assorted mics in the studio, the Mic2496 was a little crisper than a Soundcraft 600 console, not as warm and full as a Great River MP2, but very similar sounding to a Mackie 1402 VLZ-Pro mixer.

The digital output is IEC-958 Type II (consumer format) with no copy protection. The sample rate flag embedded in the data stream is fixed at 48 kHz regardless of the actual operating sample rate. Most modern digital recorders don't care about this flag, synchronizing Word Clock to the incoming S/PDIF data. While happy at 48 kHz, an aging Sony

FAST FACTS

APPLICATIONS

Location recording when you can use a digital output

KEY FEATURES

Compact size; wide choice of sample rates; clean sound; battery power

PRICE

\$549; mic breakout cable, \$35; coax adapter cable, \$35; TOSLink cables, \$15-30 depending on length and configuration; AC adapter, \$12

CONTACT

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PRODUCTPOINTS



- Good sonic performance
- 48 V phantom power
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- Compact and rugged construction
- Supports sample rates to 192 kHz



- Cables emerge from too many directions
- Experienced occasional crackles when recording from the "coax" S/PDIF output

SCORE

The Mic2496 — while pricey compared to modern portable preamps — sounds good and takes up little space, but gets bristly when connected in a useful manner. It needs a facelift and it's getting one.

PCM-2300 DAT didn't like the Mic2496's 44.1 kHz output, refusing to record until I forced the sample rate flag to 44.1 kHz using a Digital Domain FCN 1 format converter. If you plan to use this preamp with older equipment, be aware of this issue.

I greatly dislike mini phone jacks, and this one was no exception. I experienced occasion-

al crackles when recording from the "coax" S/PDIF output. An NTI Digilizer confirmed that, indeed, it was not a very robust connection. The TOSLink optical output, however, was completely solid.

My principal requirement for a portable recording system (beyond basic sound quality, of course) is robustness — ideally a straightforward recorder requiring only microphone cables and possibly AC power. The Mic2496 is sized to fit back-to-back with an HP iPAQ PDA, making a neat handful, but cables emerging from three sides of the preamp made combining it with the Jukebox 3 a bit awkward. Careful cable dress is important to avoid a messy spaghetti pile. I would have preferred the preamp outputs on the rear, leaving the front free of cables, but seeing how crowded things are inside, I doubt this was possible. (The manufacturer responds: In part thanks to these comments, a \$599 Mic2496 V2 is debuting at AES, and on it we've moved all cabling to the rear and all controls to the front; also, it adds zero-latency monitoring, raises the 48-volt phantom power current available to a true 10 mA per channel, and now has six level-monitoring LEDs per channel that indicate -40, -30, -20, -10, -3 and clip. — Ed.)

SUMMARY

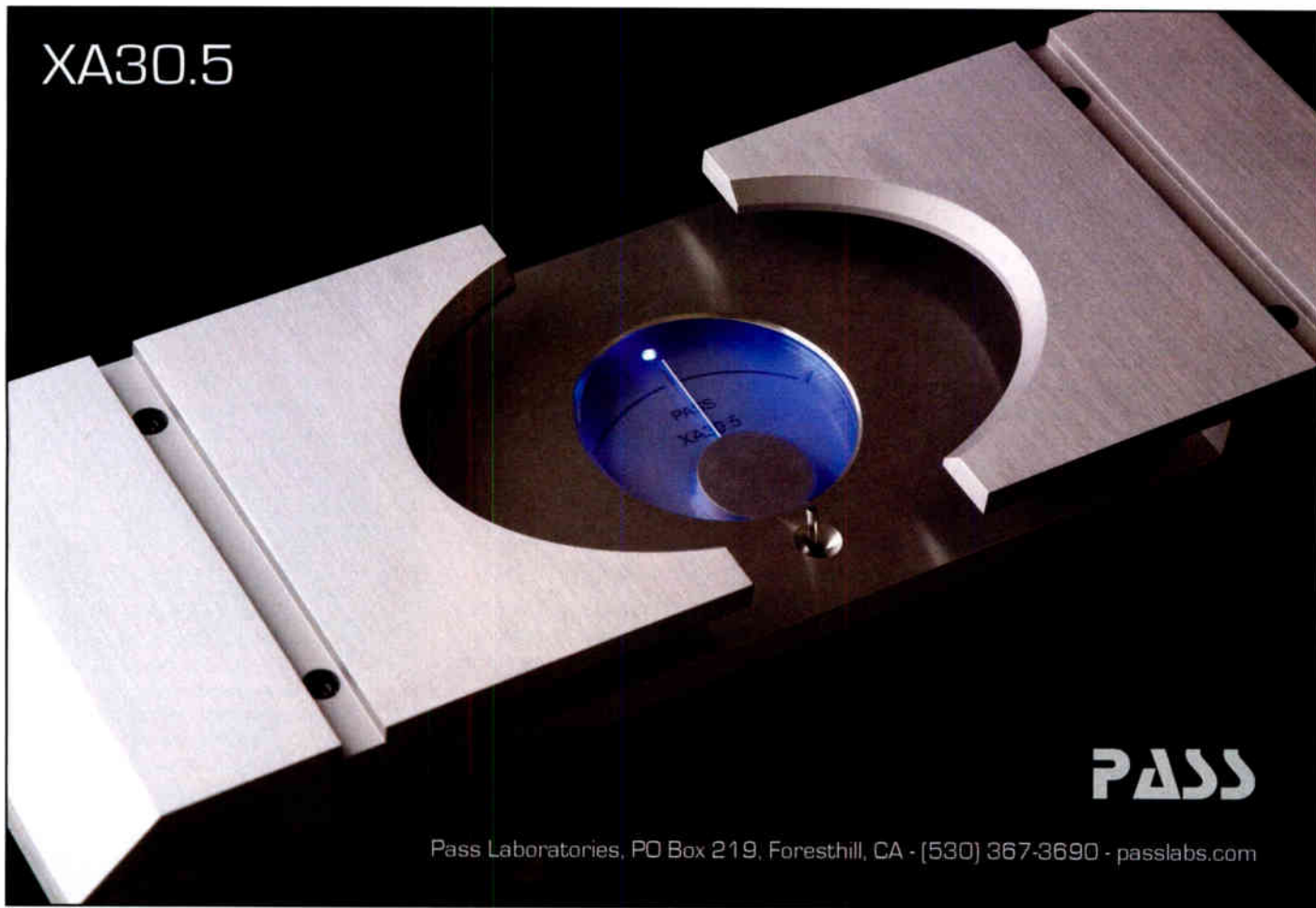
The Mic2496 is a clean and quiet full-featured preamp that covers the bases for portable recording. Use it with a recorder that takes an S/PDIF input, either coax or optical, and watch those record level meters!

Mike Rivers has a long list of credits with Rounder, Folkways Legacy and the Smithsonian. He has a degree in Electronic Engineering and is the author of the last Mackie Hard Disk Recorder manual.

REVIEW SETUP

Mic24/96, Creative Labs Nomad Jukebox 3, Sony PCM-2300 DAT recorder, TASCAM CDRW-5000 CD recorder, Computer with Cool Edit Pro and Lynx L22 sound card, Great River MP2H preamp, Mackie 1402 VLZ Pro mixer, Soundcraft 600 console, NTI Digilizer, Minilizer, and Minirator. Microphones used for test were AKG C-451 and C-414, Neumann U-87 and KM-84, Studio Projects LSD2, Beyer M88, M260 and M160. Monitoring with Sony MDR-7506 headphones and KEF 103.2 speakers.

XA30.5

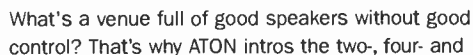


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The latest news and products

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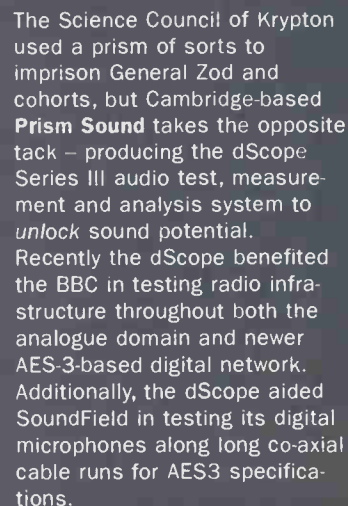
You want to work with power, you want to play with power, and the Panamax MFP-400 gives you the opportunity for both. The MFP-400 supplies clean, noise-free power to high-definition (HD) flat-panel displays for enhanced image and sound quality, while protecting the system from damaging over/undervoltages. The MFP-400 features a slimline design that makes it ideal for mounting out of view behind flat-panel displays, and offers power cleaning and premium filtration to eliminate common symptoms of AC line noise, such as loss of detail, pops, hisses, hums and visual artifacts. This allows equipment to perform at its full capability for an optimal production suite or public venue performance.

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World Radio History

by Dan Wothke

Aviom Pro 16 Series

Since Aviom's introduction into the market in 2002, the company has made quite an impact specifically in the area of personal monitor mixing. Add in the company's proprietary A-Net protocol for networking audio and Aviom is quickly becoming a staple in the audio market.

It has taken some time for A-Net to catch on, but it has proven a solid protocol and Aviom continues to develop more products while maintaining a practical and cost-effective approach. It seems every month or two another manufacturer is getting on board by providing expansion cards and interfaces to work with the A-Net protocol, which helps validate Aviom's staying power.

I was shipped three units from Aviom's

To get the analog signal out of the box, the unit is equipped with two DB25 connectors on the back and can be fanned out to XLR, 1/4-inch or any desired connector. (The pin out diagram is on the Aviom website.) As an option, Aviom also sells handy patch bays destined to fill this need.

When used in conjunction with the AN-16/i input module or AN-16/i-M Mic Input Module, the units easily become a multi-channel digital snake. This removes the trouble

and mute. Pan and volume controls also resemble the A-16II, all with comparable LED indicators for current selection, pan settings and volume. The master section is also spot on, including bass, treble and volume. The A-16R only offers one headphone output jack but can handle multiple headphones with an external 1/4-inch TRS split.

So, other than the 1RU mountable enclosure, what is added to the functionality? For starters, there is a Mix In knob on the front that controls the level to the corresponding 1/4-inch TRS jack



Aviom A-16II and AN-16/i

on the rear of the unit to add in an analog stereo input directly into the box. This would facilitate

the need to get a click or reference track directly into the box without having to use up one of the 16 inputs on the AN-16/i. Also included is a pair of stereo insert points, which can affect the entire mix before the split to the main and headphone output. Main L/R female XLR and Balanced TRS outputs are provided on the rear of the unit. With these outputs fed to an amp, the A-16R acts as a mixer not only for headphones, but also for a set of mains in a small setup, even for a powered wedge. Rounding out the rear is a set of MIDI In/Out/Thru jacks allowing the user to control presets or save/load settings. In Aviom's Pro16 line, MIDI is not converted and sent down the A-Net cabling. It can only be used with a direct local MIDI connection.

Many applications would benefit greatly from an A-16R. The unit can be remotely controlled from the A-16CS, which looks identical to the A-16II but acts solely as a remote control for the A-16R. This enables almost full use of the A-16R, analog inputs and inserts, but keeps the area where the performer is setting the mix clean with only a Cat5 cable required to connect the two units. However, the auxiliary analog Mix input cannot be controlled via the remote control. That must be controlled from the front of the A-16R.

Another application would be for a conference room or gymnasium, or anywhere an installed product with a flexible mixer is desired. For such an environment, a mixer can be moved anywhere within an Ethernet-networked, A/V-room room if the A-16R and A-16CS are used together. In a gymnasium or arena, all of the A/D and levels are set in the



Aviom AN-16/o



Aviom A-16R

Pro16 line – the AV-M8 eight-channel mic input module with AN-16/o output module and A16R personal 16x2 mixer – and enjoyed putting them through the paces, chronicled here.

AN-16/O

The AN-16/o is a 1RU (19- x 6.7-inch) 16-channel output module taking 24-bit/48-kHz uncompressed digital audio and converting it to either mic or line levels. The front of the unit contains eight level selector switches for either mic (-17 dBu) or line (+4 dBu) level. Each switch sets the pair of output levels. Green signal and red clip lights are provided, along with indicator lights for power and A-Net connectivity. The rear of the unit contains a jack for the provide wall wart-type power supply.

A-Net connections are available via locking Neutrik EtherCon connectors and include input, expansion and output. Two units can be linked together with the A-Net connection, allowing for up to 32 channels to be transmitted down a standard Cat5 cable. If the user needed more inputs, Aviom's AN-16SBR bridging unit can be installed and will transmit up to 64 total channels. However, the install must include one AN-16/o for each 16 channels to be converted.



Aviom A-16CS

and expense associated with long analog cable runs by utilizing A-Net connectivity. In most cases it is much easier to run one Cat5 cable up to 500 feet as opposed to multi-channel audio cable. Neither the AN-16/i nor AN-16/o produce any gain themselves; the units only convert the raw audio into digital for transmission over A-Net. Both the AN-16/i-M and the AV-M8 do, however.

A-16R

If you are familiar with the A-16II (Aviom's flagship personal monitor mixer), then imagine that with some additional functionality mounted in a 1RU (19- x 6.7-inch) enclosure and you basically have the A-16R. The front of the panel looks very similar to the A-16II with the channel select buttons, recall, group, solo

A/V room, with only the remote control required for the engineer or talent to control the entire mix.

This removes the need for a large rack of equipment to be moved around and is easy enough for someone without a large degree of technical expertise to use. The only additional equipment that is required would be a device to set gain for mic and line inputs. And this is where the next piece of equipment comes in handy.

AV-M8

In an environment where inputs don't change often, the AV-M8 is available. The AV-M8 is a 1RU (19- x 9.25-inch) eight-channel mic input module designed as a preamp and converter from analog to digital for A-Net distribution. Level control is via six dipswitches per channel, where different combinations of on/off settings control the amount of level for the associated input. I have had my battles with dipswitches in the past, trying to get things set to the correct position while getting a headache, but Aviom has this unit laid out for easy setup. The combination of position-to-level settings is screen printed right on the unit.

Each input has a 5 mm Euroblock connector to screw in bare wire ends. With the flip of a dipswitch, both low-cut filtering (at 85 Hz) and 48 V phantom power are available. Inputs 1 and 2 have a separate switch for a 20 dB pad to accommodate line inputs. The rear of the unit has a standard IEC power connector, power light and dipswitches for stereo linking any two channels. ADAT optical connectors are used to cascade two units together, and a switch selects clock source for either the ADAT clock or internal. A-NET expansion port and output ports live on the rear of the unit. My only wish for this unit is some type of LED indicators for level, stereo linking and even phantom power. Basically, this unit is a preamp and converter ideal for simple audio conferencing without the need for drilling or trenching. Any clipping that would occur would happen on the pre, so an available LED clip light could prove to be helpful. The AV-M8 mounting options are standard rack ears or mounting flange for table mounting.

CONCLUSION

I can't help but refer to the Aviom products as the Lego blocks of audio. By themselves they are simple, durable pieces, but properly assembled they can create a distribution system that's very effective yet simple enough for anyone to operate. The design is consistent from one product to another, and digital routing could not get

Aviom AV-M8



any simpler as there is no configuration required by the end user. The A-Net protocol automatically routes 1:1. It is feasible to have an entire facility – whether theatre, house of worship, studio/post house or even movie/TV set – to utilize the A-Net systems with the only cable

required for distribution being standard Cat5. This makes everything neat and, relative to multi-channel analog cable, inexpensive. Each one of these units is designed for a particular need, as Aviom continues to expand the company's usefulness in multiple facets of audio networking and distribution (and for greater sample rate and audio capabilities, Aviom offers the Pro64 line of products).

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by Mike Rivers

Unearthing Ground Isolation Tools

Electrical and electronic pollution from everyday devices has made attention to shielding and grounding more important than ever. Plain ol' AC power hum, as well as switching power supplies, computers and wireless telephones scattered throughout the studio contribute their own flavors of buzzes, hisses and crackles.

It's commonly believed that balanced connections are less noisy than unbalanced, but the common mode rejection that balanced connections offer isn't a universal cure for hum and noise. One can easily lose faith in the science, as well as in the guru who advised you to spend the extra bucks for balanced equipment. Contemporary manufacturers often take design shortcuts to provide feature-packed equipment at low cost. Balanced connections are quite common in today's studio gear, but not all shortcuts offer effective ground isolation or high common mode rejection.

With a balanced connection, noise that sneaks through the cable shield appears equally on both wires. Since the input stage looks at the difference between the voltage on each wire and ground, the noise common to both wires is effectively cancelled. The cable shield can also carry noise into a circuit, and this needs its own solution.

In this feature *PAR* will outline some of these things that go "hummmmmmmmm" in the night, and things that make them stop...

GROUND - DEFINITION AND CONVENTION

Electrical power wiring always has one foot firmly planted in the earth to reduce damage from lightning and other power surges. In electronics, however, the term "ground" has several meanings, only one of which is a connection to earth. In the practical sense, ground is a very low impedance path for electrical current to return to its source. Ground doesn't have to be referenced to the earth for noise-free performance — battery powered gear works just fine.

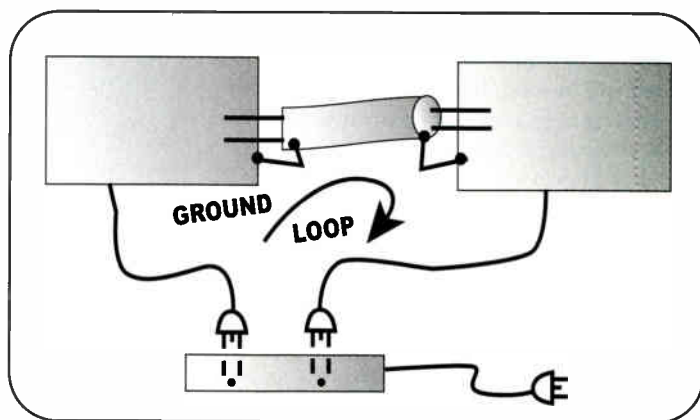
In audio equipment, "ground" is the common point to which signal and circuit power are referenced. It's usually the negative power supply terminal (or zero voltage in a bipolar-powered unit), but it's not necessarily the chassis. This can lead to ground loops and the "Pin 1 Problem."

Pin 1 is the terminal of any equipment's signal connector to which the cable shield is (or is supposed to be) connected. By convention, it's Pin 1 of an XLR connector, the sleeve on a phone plug, or the shell of an RCA connector. On a terminal strip or multi-pin con-

necter Pin 1 is wherever the manufacturer puts it.

THE PIN 1 PROBLEM

For effective shielding, the cable shield (Pin 1) must be connected solidly to the chassis. The shield becomes an extension of the box, leaving no place for noise to enter. In "vintage" days, when everything had a metal chassis, it was easy to build like this (though often "vintage" equipment is ground-faulty). Today's use of plastic cases and PC board-mounted connectors that aren't physically attached to the box make this more difficult. A manufacturer's attempt to eliminate ground loops by isolating the shield from the chassis can create a



roundabout path between the cable shield and ground, mitigating one problem but causing another.

With this type of construction, noise current flowing through the cable shield doesn't stop short at the equipment cabinet. Instead, it pulls the equipment's ground point off zero potential, causing the input ground (Pin 1) to act as a signal input. Pin 1 Problems can exist either at the input or output, or both. Manufacturers are now becoming conscious



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of Pin 1 problems and are cleaning up their act, using properly grounded connectors and solidly bonding the signal and chassis ground together. The Audio Engineering Society has addressed these issues in their recent AES48-2005 standard.

It's best to simply not buy any equipment with Pin 1 problems. But how do you know what's "Pin-1 clean?" Perhaps this is something that we reviewers should begin to report. It's fairly easy to test for a Pin 1 problem, but there's no established standard for what's acceptable.

ONE-END-ONLY SHIELDING

To prevent noise from coming in via the cable shield, why not simply disconnect it? Indeed breaking the current path by connecting the shield at one end only (OEO), sometimes called "telescoping shields," is a time-honored method of de-noising a system. Nearly all first generation project studios used RCA jacks, and were good candidates for OEO wiring. Today, personal studios are generally wired with store bought cables that have the shield connected at both ends, making OEO wiring impossible without modifying the cables.

This is not a bad thing, though. A disconnected shield is a hole for EMI to creep in. Without tape hiss and internal hum to cover up the problems introduced by OEO shielding, we need a different approach to make our systems "quiet enough for digital."

GROUND LOOPS

Having more than a single path between the shield and the chassis creates a loop: the infamous "ground loop." Since a ground loop is usually a low resistance path, current induced in the loop can be fairly substantial. The most common ground loop occurs when two pieces of equipment have their grounds connected through both the cable shield and the AC power safety ground.

PROBLEM SOLVERS

There's no better balanced output or differential input than from a good transformer. In the tube days, input and output transformers were necessary to convert the low impedance of the outside world to the high impedance of the tube circuitry. Transformers also offered a bonus: isolation between ground and signal currents and excellent common mode noise rejection. The

inherent low impedance of solid state designs reduced the primary need for costly transformers, resulting in significantly less expensive audio devices.

Transformers can introduce distortion, they have a limited usable bandwidth, and can have irregular frequency response within that bandwidth. Transformers make excellent problem solvers when interconnecting equipment that has the wrong combination of design shortcuts, but it's important not to compromise system behavior by adding a transformer.

With those problems in mind, let's take a look at a few commercial solutions to noise problems in audio installations.

JENSEN ISOMAX

Jensen IsoMax dual channel Ground Isolators employ transformers optimized for specific applications. The PI series is designed to solve most "transformerless input" hum and common mode rejection problems; the DM series is optimized outputs; while the PB series is universal, working equally well on both inputs and outputs. A good measure of transformer quality is weight, and these weigh between 1.5 and 2.5 pounds.

DIP switches configure Pin 1 connections. In one configuration, input and output Pin 1's are connected directly to the steel case



Electrical power wiring always has one foot firmly planted in the earth to reduce damage from lightning and other power surges. In electronics, however, the term "ground" has several meanings, only one of which is a connection to earth.

(and hence to each other), providing complete shielding integrity. This provides excellent common mode rejection when neither the source nor destination has a Pin 1 problem. Alternately, Pin 1 can be completely disconnected from case ground, or it can be grounded through a series resistor and capacitor. This R-C network effectively **GROUND ISOLATION** continues on page 104 ►

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lifts the input and/or output ground at audio frequencies, but provides an effective high frequency ground for EMI.

On the bench and in use, these isolators are so good that they're boring (which is a *good* thing). Of the products I examined for this article, only Jensen provided full specifications. THD was well below 0.05-percent over the full audio bandwidth at input levels exceeding +20 dBu. The specified -3 dB frequency response points are different for each model, but all extend well beyond the audio range at both ends. The DM (output) isolator, for example, extends from 0.1 Hz to 15 MHz!

At the recent no-budget Washington Celtic Festival, our sound system was assembled from gear gathered from several sources, with system engineering strictly on-site based on what was available the day of the show.

The monitors hummed. The console's transformerless balanced monitor outputs fed a pair of parametric equalizers with unbalanced inputs and outputs, which in turn fed a power amplifier with transformerless balanced inputs. Bypassing the equalizer reduced the hum substantially. But patching a Jensen PB between the console output and equalizer input didn't eliminate the hum, but reduced it to a tolerable level, so on with the show!

Jensen transformers, however, are strictly top of the line and are expensive (in the \$200-\$270 range).

EBTECH HUM ELIMINATOR

The Ebtech HE-2 is a two-channel isolation transformer in an all steel case with 1/4-inch TRS jacks in and out. It's also available as an eight-channel rack mount configuration, and XLR connectors are now offered. The input and output sleeve (ground) terminals are isolated, effectively breaking the ground loop resulting from chassis-to-chassis connection through the cable shield. Frequency response is fine at nominal line levels, but I found it to be shy on headroom at +4 dBu, THD at 30 Hz was



below 0.5-percent but climbed rapidly as the input level increased. At +14 dBu (a mere 10 dB of headroom over +4 dBu), THD at 30 Hz was above 50-percent (that ugly waveform started out as a sine wave!). Interestingly, the low frequency distortion is largely even order harmonics. When listening to a hot kick drum track through the HE-2, the harmonics added some extra oomph that wasn't at all unpleasant. Above 100 Hz, there's

little to complain about, with THD remaining comfortably below 0.05-percent up to +16 dBu. Ebtech's only published distortion specification claims less than 0.005-percent at 1 kHz, but at an unspecified level.

A disconnected shield is a hole for EMI to creep in. Without tape hiss and internal hum to cover up the problems introduced by OEO shielding, we need a different approach to make our systems "quiet enough for digital."

Being smaller, lighter and less expensive (\$60 on the street) than the Jensen Iso-Max series, I didn't expect Jensen quality. But it's a good field tool kit tool.

HOW LOW CAN WE GO? RADIOSHACK GROUND-LOOP ISOLATOR

I picked one up to see what a \$17 two-channel isolation transformer compared with the "pro" units. The RadioShack isolator turned out to be better than I expected. It's strictly from Consumerville, appearing as a small potted lump with captive cables terminated in RCA plugs. Input and output shields are isolated. THD at +10 dBu at 30Hz was a tolerable 2.5-percent, less than the



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PRODUCT INFORMATION:

Ebtech Hum Eliminator HE-2
two-channel isolation transformer

Price: \$90 (HE-8 eight-channel
rack: \$290)

Ebtech HumX

Price: \$90

Ebtech Audio | 847-639-4646 |

www.ebtechaudio.com

Jensen IsoMax dual-channel
Ground Isolators

Price: \$200-\$270

Jensen Transformers | 866-ISO-MAX1 |

www.jensen-transformers.com

RadioShack Ground-Loop
Isolator (#270-050)

Price: \$17

RadioShack | 800-843-7422 |

www.radioshack.com

THAT 1200 Series InGenius
Balanced Line Receiver ICs

Price: N/A

THAT Corporation | 508-478-9200 |

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Ebtech, though above 100 Hz the Ebtech has it beat as long the operating level stays below +10 dBu. Considering the near throwaway cost, however, it can do for an emergency fix, and you can send someone out to pick one up on a Sunday afternoon.

AN ISOLATOR OF A DIFFERENT COLOR: EBTECH HUM-X

Unlike transformer isolators which break the ground path through the cable shield, the Hum-X deals with a different ground loop path. When two pieces of equipment are plugged into grounded electrical outlets

their chassis are connected through the safety ground of the AC power cord. This parallels the ground path through the shield, forming a loop through which current can flow.

Most of us have at some time fixed a hum problem by lifting the safety ground with an adapter and managed to live through the experience. Ground-lifting is a useful diagnostic technique to guide you to a better solution, but it's not a permanent fix.

The \$60 Hum-X is a safe way to break the

power ground. It's a completely sealed 3- x 1.25- x 1.25-inch block with a 3-pin US-style AC plug on one side and matching receptacle on the other. The plug blades are oriented so that, when plugged into a conventional power strip, it doesn't eat up the adjacent outlet.

Although I didn't take a hacksaw to the Hum-X, probing it with a VOM suggests that it involves of a pair of back-to-back diodes in series with the safety ground pin. For small ground loop currents, the diodes don't conduct, effectively breaking the ground path. But an equipment ground fault will cause sufficient current to flow, turning on the diodes and provide the safety ground protection. Pretty clever – ground when you need it, no ground when you don't. It seems to fix any hum problem where a ground-lift adapter works.

LIFE IN THE COMMON MODE

In addition to providing ground isolation, a good transformer offers a high degree of common mode rejection, probably the major benefit of balanced interconnections.

Not all transformerless inputs are evil, however. Circuit topologies exist that provide common mode rejection on par with a decent transformer. One such circuit, developed by Bill Whitlock of Jensen Transformers, has recently been implemented in IC form by THAT Corporation, and the InGenius 1200-series balanced line receiver has begun to make its way into professional audio gear. Like the "Intel Inside" identity program, THAT has begun an InGenius identity program to identify products that utilize this component; look for the logo on an input near you.

SUMMARY

If you have a ground-induced hum or common mode noise problem, any of these transformer isolators will be effective. The Jensen isolators offer sufficient headroom so as not to be of concern in most audio installations. The Ebtech and RadioShack devices get unhappy near the maximum output level of most professional units, so they may require some gain re-staging for best performance. The Hum-X, while a bit pricey compared to the cheap-and-dirty ground lift adapter, adds a degree of safety that could save a life. Priceless.

Mike Rivers has a long list of credits with Rounder, Folkways Legacy and the Smithsonian. He has a degree in Electronic Engineering and is the author of the last Mackie Hard Disk Recorder manual.

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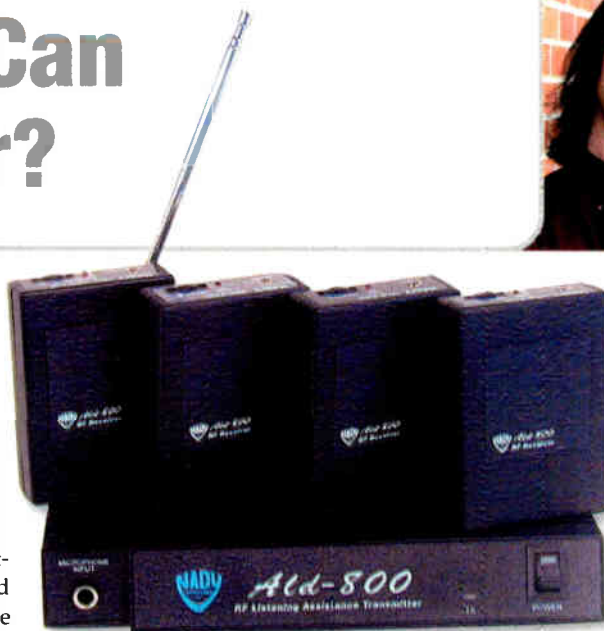
Oh Say, Can You Hear?

For my last column, I wrote about the ABCs of audio distribution throughout the House-of-Worship facility. I mainly focused on areas outside of the HOW that would most benefit staff, volunteers and people transitioning from one room to another. I received an email from one of the faithful *PAR* readers pointing out a possible solution that I failed to mention for distribution around the HOW: the use of an FM transmitter to distribute sound. Depending on the frequency, standard FM receivers located around the campus can pick up the signal. FM is also one of the most common methods for audio distribution inside the sanctuary.

WHAT DID HE SAY?

One of the challenges of HOW audio — especially in a smaller facility where arrays of properly aligned delay speakers are not in the immediate budget and coverage is less than optimal — is for everyone to hear the spoken word at a consistent and comfortable level. This especially rings true for those who are hard of hearing. Generally we envision the elderly when referring to anyone with hearing difficulties, but the situation is not limited to just that demographic. Ask anyone young or old with a hearing problem and they will immediately share with you the frustration of struggling to make out the spoken word above the rustling of paper, restless children and air conditioning — all types of extraneous noises that many listeners can tune out. Luckily, there are a few cost-effective methods to remedy this and not leave out anyone wishing they could hear better during the service.

FM transmitters are probably the most common solution. Several manufacturers have developed and honed equipment specifically for this purpose, including transmitters and receivers available in one package deal. This is very similar to using a walkman to listen to your favorite radio station, except the available range is far less depending on the rating (usually rated in feet) of the



Nady ALD800 listening assistance system

transmitter. Product categories of price point and sanctuary size are covered, ranging from 50 feet and up, while some packages offer extension antennas to greatly expand the range. Belt pack receivers simply equipped with a mini-jack for ear buds and a volume control can be set on a fixed frequency to match that of the transmitter. This makes the product very user friendly for the listener, and offers the most flexibility as they can sit anywhere within the sanctuary. A nice feature of the belt pack is a battery indicator, so the person does not have to get up in the middle of the service to visit the welcome desk and pick up a new battery.

The downside to an FM solution is hygiene and batteries. Covers for ear buds must be changed out for each new user, which can get tedious and the cost does add up. Batteries should also be considered as part of the reoccurring cost. These are minor compared to the ease of use, flexibility and overall low cost of ownership since these are generally — to quote Mr. Ron Popeil — “set it and forget it” systems.

An alternative to FM transmission is IR (Infrared), which relies on sight lines to transmit the audio. The transmitter is usually a large box mounted within the sanctuary. They work well in an open area, are not susceptible to any frequency interference and are touted as the best-sounding systems. The coverage areas, depending on the system, can be comparable to FM and repeaters can be added to the system to expand the range. However, if walls or other obstacles are an

issue, they may not be the right fit. But in an open area IR is a viable option. Movie theatres commonly use the obstacles as a boundary, so the movie in one theatre cannot be heard in the theatre next door.

BRING YOUR OWN RECEIVER

You can customize your own transmitting option for the choir by purchasing a FM transmitter that works on the standard FM 88 - 108 MHz frequency range, then require each choir member to bring in their own “walkman” and ear buds. Transmit on a vacant frequency, have the choir tune to that frequency, and you now have a full choir with ultra-low latency monitoring, not to mention no loud wedges or rear fill monitors to cover the area. The FM medium is very useful for delivery throughout the church.

TRANSLATION PLEASE

A growing trend in HOW is the ability to translate sermons in real time to foreign language attendees. I had the pleasure of seeing this take place at a conference in England. There were over 10 different countries represented and about that many languages. The setup just for translation itself was quite impressive, with each language having a portable vocal booth setup with an English feed right into the booth. To avoid bleed from the English making its way down the line with the translation, most of the translators wore headphones. Then they had a microphone setup with a portable mixer and a feed to a FM transmitter. Each language was assigned a different frequency and the people that attended the conference would stop by the table to check out the receiver that was used for their language. Everyone sat together in that auditorium and was not hindered by any language barriers thanks to the organized setup and use of this technology.

Whether a HOW has one or many secondary languages, translation can be accomplished through FM transmission. The only obstacle is the person who is translating, because he or she can be a distraction. Granted, not every HOW can install specific isolation booths for this purpose, but it is possible to work around that problem by feeding the primary language message to the translator in another room via a wholly separate FM

frequency (or other distribution method).

INDUCTION LOOP SYSTEMS

In permanent installations, an induction loop system could be a good fit. The induction system works with an amplifier and wire that is installed around a fixed seating area. The wire itself can be installed underneath the carpet to keep it out of sight, or in a lower ceiling. Within the loop, an electromagnetic signal is created and picked up by a "T" coil. Most hearing aids have the ability to receive the signal from T coils with the minor switch of a setting, and users can control level right from their hearing aids; listeners can then hear the spoken word with no additional equipment necessary.

If a listener does not have the ability to receive the signal, additional belt pack receivers can be purchased. The benefit is very low cost, as there are no batteries to replace, no covers for the ear buds, and less receivers. The downside is the coverage area is fixed within the room, which limits where that person and their family member can sit. These systems can be tricky to install in some facilities, but would be a good option to consider when installing in a new facility.

The three listening assistance options explained here are generally the most cost effective, but can also be limited in their own ways. As I stated earlier, it is not always the elderly who need the assistive listening devices (though they are probably the primary demographic), so different solutions need to equip different generations. With that said, one of the highlights of sitting atop the sound booth is seeing a grandma worship and listen to the sermon right next to a teen or 20something with piercings and weird-colored hair. The generational differences seem to melt away when in a HOW, and isolating people based on their hearing needs would sometimes eliminate this important and encouraging aspect in a House of Worship.

I am not overlooking the obvious here, which is having correctly designed rooms and proper audio coverage can help minimize some of these needs when referring to the hard-of-hearing. Many, if not most, of the

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wireless assistive
listening system



HOW engineers I have had the pleasure of interacting with have found the needs and the budgets aren't always equal, so we must work with the confines of our situation. I will be addressing those issues in upcoming columns, and will be sure to call upon some well-qualified designers in an effort to continually search for effective ways to address sound for HOW rooms. Regardless, as long as we have HOW, we will have different groups of people with different audio needs. Aurally

clarifying the message for those who cannot hear well or do not speak our language fluently is an important need that we must always strive to meet.

As always, I welcome your thoughts and input on this and any HOW column.

Dan Wothke currently runs the gauntlet of all things media in his role as Media Director at Belmont Church in Nashville. He invites you to contact him at dwothke@yahoo.com.

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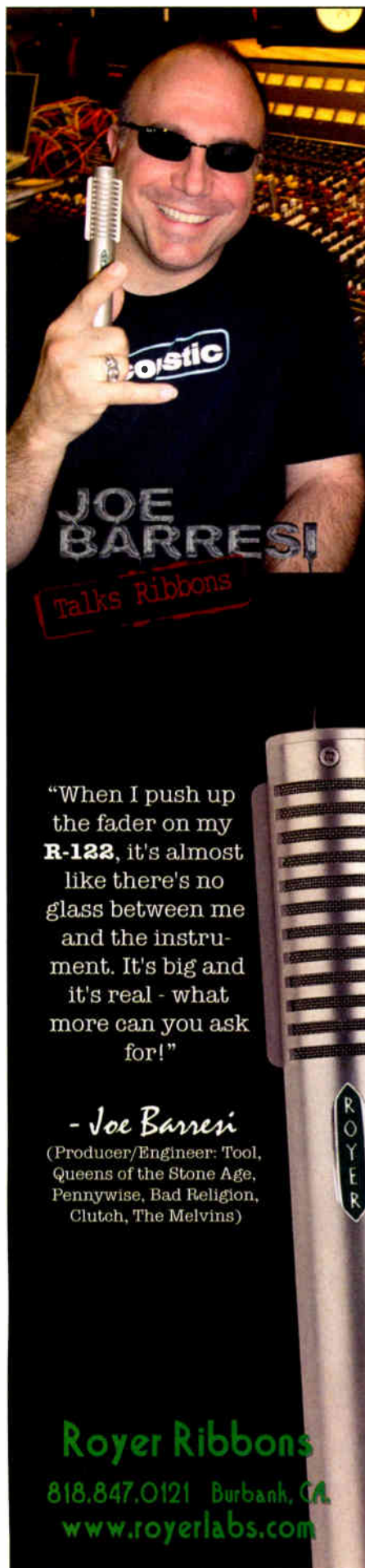
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JOE BARRESI
Talks Ribbons

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- Joe Barresi
(Producer/Engineer: Tool, Queens of the Stone Age, Pennywise, Bad Religion, Clutch, The Melvins)

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FEATURES: Plug-ins; Windows Vista/Mac OS X/Linux; 24 bit/192 kHz; DSP effects; VST, AU, RTAS, TDM; 30 Input Stages, 60 Compressors, 5 EQ models per band; Side Chain-able Filters.



PRICE: \$1,499 (TDM-RTAS-AU VST), \$750 (Native-RTAS-AU VST)

CONTACT: URS at www.ursplugins.com.

PETERSON StroboSoft Suite

FEATURES: Vista/2000/XP/Mac OS X DAW software; strobe tuner; 0.1 cent accuracy; 59 tuning presets; 38 exclusive tuning Sweeteners, unlimited user-presets; 20 Hz - 5 kHz; 96 kHz support; spectrum analyzer; oscilloscope; VST/AU (available 11/07).

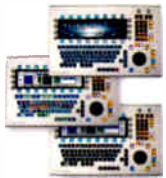


PRICE: \$149.99

CONTACT: Peterson at 708-388-3311, www.petersontuners.com/stroboSoft.

XYNERGI Media Production Center

FEATURES: DAW interface; software, tactile control combination; self-labeling key switch technology; CC-1 engine powered.



PRICE: \$23,000

CONTACT: Fairlight US at 626-793-3940, www.fairlightus.com.

AUDIO IMPRESSIONS INC. Realtime Instruments Strings Sample Library

FEATURES: DAW software; 70 Violins, Violas, Cello, Basses recorded at 192 kHz; supports all sequencers via AU or VST plug-ins; Native Instruments' Kontakt 2.2, AudioMulch 1.0, FX Teleport and Windows XP Pro-SP 2.



PRICE: \$3,499

CONTACT: Audio Impressions Inc. at 818-456-7103, audioimpressions.com.

SONNOX Oxford Limiter

FEATURES: Professional stereo limiter plug-ins; reconstruction meter, enhance section, comprehensive dither and noise shaping, attack and release controls; 192 kHz capable;



many presets; Windows/Mac compatible; for Pro Tools, TC Powercore and all Native formats (AU, VST, RTAS).

Pricing: \$531 (TDM), \$450 (Powercore), \$351 (RTAS)

CONTACT: Sonnox at www.sonnoxplugins.com.

DIGIDESIGN Pro Tools

FEATURES: DAW software; Mac OS X, Windows; 24 bit/96 kHz with Pro Tools LE, 24 bit/192 kHz with Pro Tools|HD; advanced audio and built-in MIDI sequencing; comprehensive mixing and automation environment; large collection of bundled effects, virtual instruments, compatible applications; support for RTAS, AudioSuite, TDM (HD-only) plug-ins; VST supported via optional third-party software.



PRICE: Beginning at \$349.99

CONTACT: Digidesign at 650-731-6300, www.digidesign.com.

FRONTIER DESIGN AlphaTrack

FEATURES: Windows XP/Vista, Mac OS X controller compatible with most major DAWs; transport controls; 100 mm touch-sensitive fader, three touch-sensitive knobs, touch-sensitive jog/shuttle strip; user-definable buttons; footswitch input.



PRICE: \$249

CONTACT: Frontier Design at 800-928-3236, www.frontierdesign.com.

LYNX AES16e PCI Express Interface

FEATURES: PCI Express card for 16 channels of 192 kHz AES/EBU I/O; ASIO, WDM for Windows, CoreAudio for Mac OS X; 24 bit, 192 kHz; 32 x 16 onboard digital mixer.



PRICE: \$795

CONTACT: Lynx at 949-515-8265 x204, www.lynxstudio.com.

ABLETON Live 6

FEATURES: Mac OS/ Windows sampler/sequence DAW software; unlimited 32/192 recording; improvisational workflow with Session View; integrated MIDI, audio effects;



and Peripherals

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PRICE: \$599 (boxed), \$499 (download), upgrades from \$119.

CONTACT: Ableton at 646-723-4550, www.ableton.com.

ADOBE Audition 3

FEATURES: DAW software; VSTi virtual instrument support with piano roll editor; spectral editing; multitrack editing; Convolution Reverb; Analog Delay; Mastering tool; Tube-modeled Compressor; radius time, pitch stretching from iZotope; Guitar Suite effects; Top/Tail Views; noise reduction, phase correction tools; waveform editing tools; ASIO, VST, DirectX, ReWire support; Windows Vista.

PRICE: \$349 (upgrade from \$99).

CONTACT: Adobe at 800-585-0774, www.adobe.com.



RNDIGITAL SPLI-ZER

FEATURES: Mac OS X, Windows XP/Vista plug-ins; adjustable frequency split point and audio levels; low- to mid-split point from 80 Hz to 1475 Hz; mid- to high-split point from 80 Hz to 16 kHz; Hanning or Blackman windowing; Mute, Sum, Low Band, Mid Band and High Band for instant monitoring; level meter and adjustable gain for each band.

PRICE: \$129

CONTACT: RNDigital at 727-230-1603, www.rogernicholsdigital.com.



ADK Lyve Tracker Studio System

FEATURES: 4RU rack DAW interface/software; based around a Steinberg Nuendo 3 or Cubase 4 audio engine, providing the Yamaha PM5D digital console with recording of up to 192 simultaneous tracks; features Intel Core 2 Duo, RME MADI technologies; removable hard drive bays; DVD-RW for backup.

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CONTACT: ADK at 859-635-5762, www.adkproaudio.com.



PRESONUS FireStudio Project

FEATURES: Windows/OS X DAW interface/software; eight Class A XMAX mic preamps; 24-bit/96-kHz, +114 dBu dynamic range rate conversion; zero-latency matrix

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time plug-ins; over 2 GB drum loops, samples.

PRICE: \$699.

CONTACT: PreSonus Audio Electronics, Inc. at 225-215-0636, www.PreSonus.com.

CEDAR Cambridge v4

FEATURES: DAW restoration, speech enhancement software; Windows XP; 24 bit/96 kHz; timecode, automation; file processor; real-time process manager; spectrum analyzer; dither; noise shaping; two-channel tools; metering; gain; DC filter; manual declick, dethump; broadband noise reduction modules: auto dehiss, dehiss-3; declick-2; vintage; decrackle; debuzz; declip phase correction; dynamics processor; limiter; dialogue noise suppression; EQ; adaptive filters.

PRICE: Dependent Upon Configuration.

CONTACT: CEDAR at 207-828-0024, www.cedaraudio.com.



CUBE-TEC SpectralDeHiss

FEATURES:

Fingerprint-style denoising plug-ins for Pro Tools; Mac only; algorithms for "colored" noise problems; hand-editable noise floor, psycho-acoustic scale; Gain for attenuation, smoothing; noise-free equalizing mode; RTAS, AudioSuite.

PRICE: \$1,950.

CONTACT: Cube_Tec at 905-469-8080, www.cube-tec.com.



STEINBERG Nuendo 4

FEATURES: DAW software; Universal Binary/Vista 64; automation; MediaBay database; post editing; 38 New VST3 Effects; routing and summing, side chaining, track presets, MPEX3.

PRICE: \$1799.99.

CONTACT: Steinberg at 877-253-3900, www.steinberg.net.



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Ryan Hewitt

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Ryan Hewitt

(Engineer/Mixer: Red Hot Chili Peppers, blink-182, Alkaline Trio)



See us at AES Booth #866

Continued From Page 109

SOUNDTOYS Native Effects

FEATURES: Six Mac/Windows plug-ins: EchoBoy, Phase Mistress, FilterFreak, Crystallizer, Tremolator, Speed; analog emulations; Pro Tools LE & M-Powered; Audio Units support for Logic, Live, Digital Performer.

PRICE: \$495.

CONTACT: Sound Toys at 802-951-9700, www.soundtoys.com.



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PRICE: RX: \$349, RX Advanced: \$1199

CONTACT: iZotope at www.izotope.com.



APOGEE Duet

FEATURES: Mac FireWire 24-bit/96-kHz I/O; two balanced XLR inputs; two unbalanced instrument inputs; headphone, unbalanced output for monitors; control built into Apple Logic Pro, Soundtrack Pro, GarageBand.

PRICE: \$495.

CONTACT: Apogee Digital at 310-584-9394, apogeedigital.com.



UAD Nevana X2

FEATURES: DAW hardware/software; two PCIe UAD-1e DSP Cards; plug-ins create virtual Neve 1073/SE, 1081/SE, 33609/SE, 88RS consoles; UA Mix Essentials (VST/AU/RTAS); card expandable.

PRICE: \$200.

CONTACT: Universal Audio at uaudio.com.



BIAS Peak Pro XT 6

FEATURES: Mac OS X DAW software; 32 bit/10 MHz; destructive/non-destructive

sample editor; Playlist views, crossfades, envelopes; 100-percent Red Book CD mastering; BIAS dither cloning, sample rate conversion; Podcasting; DSP support; AU/VST effects/instruments; sampler support; edit UI/metadata; SoundSoap Pro analysis, restoration.

PRICE: \$1,199.

CONTACT: BIAS at 707-782-1866, www.bias-inc.com.



JLCOOPER MCS-3800 Media Command Station

FEATURES: DAW peripheral for Pro Tools, Nuendo, Logic Pro, Digital Performer, Samplitude, Sequoia, Final Cut Pro and others; eight touch-sensitive faders; 60 function keys; full-size numeric keypad; transport controls; LCD/LED displays; MIDI I/O, two card slots.

PRICE: \$3,299.95.

CONTACT: JLCopper at 310-322-9990, www.jlcooper.com.



IZOTOPE RX

FEATURES: DAW restoration software; iZotope Advanced Audio Algorithms: Denoiser,

SSL X-EQ

FEATURES: DSP-powered plug-in for Duende; 10-band parametric EQ, each band assignable to a different type of EQ; high and low filters, each can employ five different topologies; shelving with a Q range to 10.3, plus undershoot/overshoot; Bell EQ's with nine response curves; 20 Hz – 20 kHz, +/- 20 dB; switchable between series, parallel flow; real-time FFT signal analysis display; stereo input/output metering with peak hold, RMS meter, dynamic range display; A/B facility; MIDI capability.

PRICE: \$599.

CONTACT: Solid State Logic at 212-315-1111, www.solid-state-logic.com.



MCDSP Emerald Pack

FEATURES: Mac OS X/Windows plug-ins; emulations of vintage equalizers, compressors, tape machines, channel amplifiers; custom console modeling; convolution reverb, mastering limiters; Pro Tools HD, LE and M-Powered.

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CONTACT: McDSP at 650-318-0005, www.mcdsp.com.



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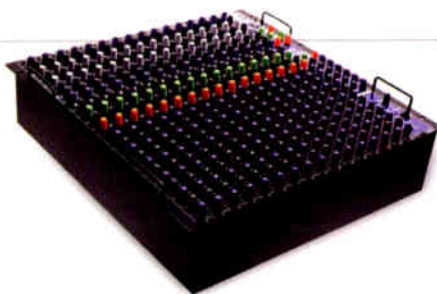
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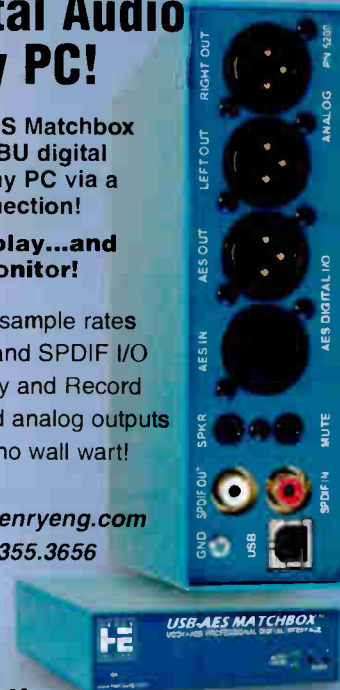
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“Kingdom Come” | Jay-Z



SINGLE: “Kingdom Come”

ALBUM: Kingdom Come (Roc-A-Fella)

DATE MIXED: Summer 2006

SINGLE PRODUCER: Just Blaze for F.O.B. Entertainment/N.Q.C. Management, LLC and Hip Hop Since 1978

SINGLE VOCAL ENGINEER: Gimel “Young Guru” Keaton for Loreal Inc. at Sony Music Studios, New York City

SINGLE TRACKING ENGINEERS: Andrew Wright and David Brown for N.Q.C. Management, LLC at Baseline Studios II, New York City

SINGLE MIX ENGINEER: Dr. Dre at Ocean Way Recording, Hollywood

SINGLE CO-MIX ENGINEER: Gimel “Young Guru” Keaton for Loreal Inc. at Baseline Studios II, New York City

MASTERING: Tony Dawsey at Masterdisk, New York City

OTHER PROJECTS: Keaton has worked with such artists as Kanye West, Ludacris, T.I., Linkin Park, Beyonce, Talib Kweli, The Neptunes and Memphis Bleek.

SINGLE SONGWRITERS: S. Carter, J. Smith, R. James, A. Miller, L. Parker

CONSOLE: Solid State Logic 4000G+ (Baseline Studios II)

RECORDER: Pro Tools|HD

VOCAL PRE-AMPLIFIER: Avalon VT-737SP

VOCAL MICROPHONE: Neumann U87

SELECT OUTBOARD PROCESSING: Solid State Logic
SL 4000 mix buss compressor, Empirical
Labs Distressor and API 550B

ENGINEER’S DIARY

Gimel “Young Guru” Keaton — engineer, vocal engineer and co-mix engineer for “Kingdom Come” — insists that the catchy track came down to, as usual, the unique genius and immense talent that is Shawn Carter (a.k.a. Jay-Z).

“Jay’s a phenomenon,” begins Keaton. “He’s not a normal rapper. You can present him with a new track and within a half-hour he’ll have a verse. By the time he steps into the booth, he’s said the verse about 10 times to himself. Then he gets the performance — the swagger — down. About two or three takes, and he gets it ... and he never punches. Then the process goes again for a second verse, then a third verse. In an hour or two, you have a song done.”

Keaton added his own expertise to the “Kingdom Come” track, which used “the most pop sample ever,” he explains; Rick James’ “Super Freak,” which begat MC Hammer’s “U Can’t Touch This,” was transformed “into something for the most urban of artists, Jay-Z. I had to break the loop into three or four different frequency ranges, then combine them back to control them a lot better. That [sample] is a great sound; it reminds me of those mid-90s records that defined our sound in hip-hop. It’s about making a new arrangement with the sounds that were originally presented in a way that wasn’t thought of by the original artist.”

Jay-Z, a “very controlled” rapper, says Keaton, is most often recorded via a Neumann U87 and Avalon VT-737SP to Pro Tools|HD. “If it’s my session, I have a U87 set up for him going into an Avalon VT-737SP with extremely light compression with a semi-quick release time,” he explains. “If you look at the wave on the oscilloscope, those higher-than-average bumps in the hills are what I’m trying to tame. My compression ratio is normally 3 or 4:1 and the threshold is really light, maybe going -2 or -3 dB on the majority of the words. But Jay’s a calm rapper, so he doesn’t have the sharp transients that someone else — like DMX, for example — would have.”

*Strother Bullins is the Reviews
and Features Editor for
Pro Audio Review.*



Gimel “Young Guru” Keaton

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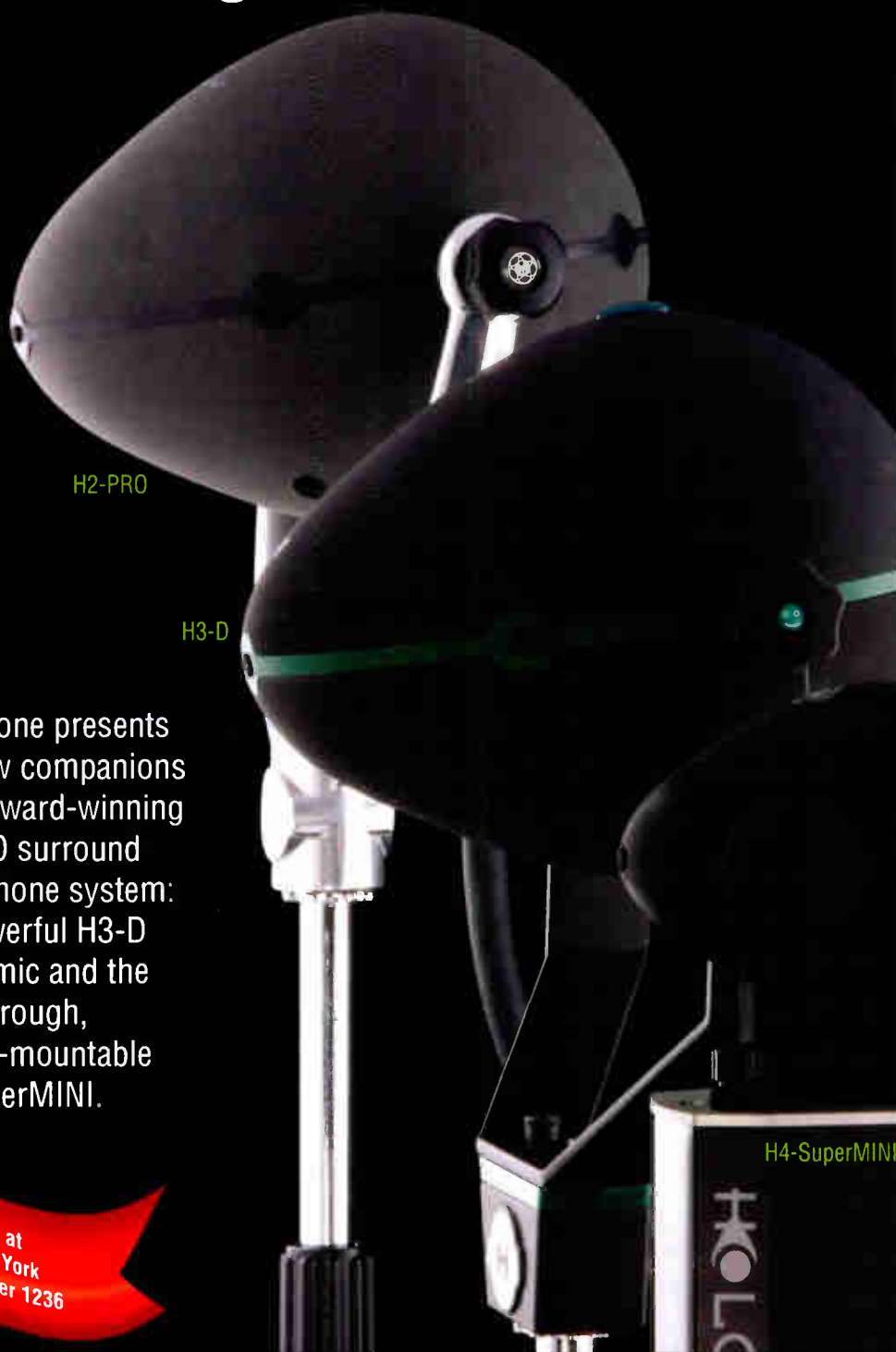


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World Radio History

User List

- ABC HDTV
- Any Given Sunday (film)
- Apple Inc.
- Asbury College, KY
- ATK Audiotek
- BellExpressview-Canada
- Bose Corporation
- Branford Marsalls live in Amsterdam
- BTU Beijing TV - China
- Bonnaroo Festival
- CBC HDTV
- CBS HDTV
- Celine Dion
- Central Michigan University
- City TV-Canada
- Cowboy Junkies
- Cristiano Pinheiro, Brazil
- CTV, Canada
- Dave Matthews Band
- Dome Productions
- Dolby Labs
- Drew Carey Show
- EA Sports
- Eddie Mapp
- Elton John
- ESPN
- Evanesence
- Extreme Makeover, Home Edition TV
- Foggy Labs
- FOX HDTV
- Frank Serafine
- Genie Awards
- George Jonsen - Mortal Kombat III,
- London Philharmonic
- Hank Neuberger - Third Wave Productions
- Hiroshima TV - Japan
- Hong Kong TV (ATV)
- Iron Maiden
- John Cale
- Joss Stone
- Kansai TV (Osaka, Japan)
- Kanye West
- King Bolden, film
- KitanShima TV - Japan
- Klaus Landsberg
- Korn
- LDS Motion Picture Studio
- Lionel Richie
- Mammoth Sound and Vision
- MBC TV - Korea
- Mexico Soccer
- Mormon Tabernacle Choir
- Moscow International House of Music
- MTV Canada
- MuchMusic Video Awards
- NBA Basketball
- NBC HDTV
- NFL Football
- NHL Hockey
- NHK Japan
- North by NorthEast Music Festival
- One Love - Bob Marley Tribute
- Paul McCartney
- PBS HDTV- USA
- Pellissippi State University
- RAI, Italy
- Real Madrid (Soccer)
- Resul Pookutty, India
- Robert Margouff
- Sam Roberts
- Sandia Labs
- Sapporo (TV) - Japan
- Shakira
- Teatro Municipal do Rio de Janeiro
- Televisione Svizzera DI Lingua Italiana
- The Kentucky Derby
- The Church of Scientology
- Three Superbowls
- TNT HDTV
- TSN HD
- TV Asahi - Japan
- TV Azteca (Mexico)
- Universidad Francisco Marroquin
- University of Central Florida
- Van Morrison, Grand Ol' Opry
- Warner Music Group, Flaming Lips
- Waterman, Andrew
- Whitesnake
- 47th and 48th Grammy Awards
- And the list goes on...