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APRIL 2008

ProAudio Review

The Review Resource for Sound Professionals

World Premiere Review!!

API 1608 Analog Console

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The Incredible Shrinking Recorder P56

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Evaluating audio products for professionals in commercial recording, broadcast production, audio for video/film, project studios, live sound, contracting and multimedia.

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Pro Audio Review
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April 2008 | Vol. **14** Issue **04**

PRO AUDIO REVIEW (ISSN 1083-6241) is published monthly by NewBay Media LLC, 810 Seventh Avenue, 27th Floor, New York, NY 10019. Subscription information can be found at www.proaudioreview.com, by calling 703-852-4600 or writing to the above address. Letters to the editor are welcomed at the above address or par@nbmedia.com.

Phone: 703-852-4600 | **Fax:** 703-852-4582

Periodicals postage paid at New York, NY 10199 and additional mailing offices. POSTMASTER: Send address changes to Pro Audio Review, P.O. Box 282, Lowell, MA 01853.

REPRINTS: Reprints of all articles in this issue are available.
Contact Caroline Freeland at cfreeland@nbmedia.com.

Printed in the U.S.A.

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FROM THE PUBLISHER

John Gatski

A Little More Recognition Please

Every since we began our *PAR* Excellence Awards back in 1996 at the annual U.S. AES convention, I have had periodic requests from companies to extend our innovative product award to other trade shows. Since *PAR* covers a number of niches within Pro Audio, there have always been plenty of shows that we covered including NAB (broadcast production/post), NSCA (live) Infocom (contracting/presentation audio and NAMM (with its divergent set of music dealer products).

So for the first time, we are branching out our brand of product recognition with our first ever *Pro Audio Review Hot Gear* list. Our first list of *Hot Gear* product, published in this issue, comes from the NAB 2008 convention. Our editors and contributors who helped out with the coverage from NAB made their recommendations after the show and the results are published on page 56-57.

The list is a pretty good one and is quite varied — from hand-held FLASH recorders, to production consoles, to test equipment and many others in between. Congratulations to all the winners. We even made a cool little logo so they can wear the badge with pride.

ProAudio Review HOT Gear 2008 NAB SHOW

We also will be making Hot Gear lists from InfoComm/NSCA and the Summer and Winter NAMM conventions. We will continue with the formal *PAR* Excellence Awards at AES in San Francisco.

CONSOLE-ATION

I wanted to pat our reviewers on their backs for the great crop of console reviews written up for this annual studio console issue. We have the world exclusive review on api's killer 1608 analog board (front cover shot, too) and first looks on the Euphonix S5 Fusion and SSL Matrix. After nearly three years since its introduction, we finally get a look at Malcolm Toft's low-cost and feature-filled ATB analog console.

Despite the computer mixing lure, pro audio folks still like their consoles and many of them are still them analog. I snagged one of John Oram's 8T boards in 06, and I love it. I also have an old American-made Mackie 1402 and that puppy still gets a lot of use as well. As long as they keep making them, we will keep reviewing them.

TAPE NEVER DIES

While I was NAB I stopped by the Otari booth to chat and, of course, the venerable MX-5050 15-ips half-track was on display, the last of the analog two tracks still in production.

But I also picked up a brochure on the Tape Project, an enterprise of fiercely dedicated tape die-hards who claim to be putting remastered classic and current master tape hi-fi recordings on 10-inch reel. Hey, since everyone always says how open reel playback sounds more like the real thing, now you can hear it again for yourself. The titles are most jazz and classical, including the venerable Bill Evans Waltz for Debby.

They even offer a network of repair centers and information where to buy used decks. I guess I will fire up the old Technics 1520 and sample a few of the tapes. Now where is that empty reel … I love the smell of tape in the morning! For more info, go to *www.tapeproject.com*.

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

ProAudio Review

Vol. 14 Issue 04 | April 2008

Evaluating audio products for professionals in commercial recording, broadcast production, audio for video/film, project studios, live sound, contracting and multimedia.

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Subscriptions
Pro Audio Review, P.O.Box 8738,
Lowell, MA 01853
Telephone 888-266-5828
(USA only 8:30 a.m. - 5 p.m. EST)
978-667-0352 (Outside the US)
Fax: 978-671-0460
E-mail: newbay@computerfillment.com
Web: MyPARmag.com

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Letters

MORE ON THE DPA 4017 REVIEW

Reviews/Features Editor's note: In the February 2008 issue of **PAR**, *Matt Hamilton, location sound recordist and owner of Nashville's Gorilla Sound* (www.gorilla-sound.com), *evaluated the new DPA 4017 in the field. This ran alongside Russ Long's main review of this shotgun mic's usefulness in the recording studio.*

In his original draft of the review, Matt stated the following: "Unfortunately, there was a noticeable amount of self-noise hiss in the upper frequencies that was distracting in the headphones. For the price tag, I would have expected a dead quiet microphone."

We later received an update of his evaluation, which was, regrettably, past February's print deadline. After conducting further A/B comparisons, Matt discovered that the comparative Schoeps shotgun "had the same high end 'airiness' and, strangely enough, the same bump at around 16 kHz that the DPA exhibited," he offered. "Why they do this, I don't know; there is nothing useful in that range for dialog and it kind of makes things sound harsh on 'S' sounds. Anyway, to each his own."

We also found Matt's updated draft quite interesting, so it runs in its entirety below.

For this field test I used the DPA 4017 on a documentary and a television production. I was recording straight to camera via a Sound Devices 442 mixer while monitoring with Sony MDR-7506 headphones. While DPA does make a shock mount for this microphone; it was not included. Instead, I used a PSC short shock mount.

I found the 4017 to be ideal for long days of ENG shooting. Its short length made low ceilings easy to navigate. Its light weight cut down on the obligatory arm cramps from long days of booming. The mic's pickup pattern seems to be wider and more forgiving of

LETTERS continues on page 8 ➤

Feedback

We want to hear from you. Send your comments to ***jgatski@aol.com****. Please include name, city, state and job title and firm in the email. For product submissions, contact Strother Bullins at* ***newproductsPAR@earthlink.net****.*

off-axis dialog than the Sennheiser 416. This makes it an excellent choice for documentary work and ENG shoots where it is often unpredictable who will be speaking next. Usually, just a slight shift in angle was enough to make up for any loss in presence.

The onboard low-cut and high-boost are well implemented and easily accessible. For on-camera mounting, the low cut sounds good and is an excellent quick fix. The high-end boost adds a 4 dB shelving boost at 8kHz. According to the manufacturer's literature, it is intended to make up for any high frequency loss caused by a zeppelin and windjammer (or "dead cat"). I did not have an opportunity to try this out, but it is an interesting idea.

> The mic's pickup pattern seems to be wider and more forgiving of off-axis dialog than the Sennheiser 416. This makes it an excellent choice for documentary work and ENG shoots where it is often unpredictable who will be speaking next.

Overall, the sound was brighter than the Sennheiser MKH60. Even with the onboard high frequency boost disengaged, there is a boost from 10kHz on up which translates to a lot of distracting 'air' in the upper frequencies. This is absent on the MKH60. For most location sound work, especially the limited frequency range of dialog recording, there is very little useful information above 12kHz. Other than this minor complaint, which is admittedly a matter of taste, the sound was well balanced with plenty of low end. I found the low end adds nice warmth to dialog without sacrificing presence. In addition, the high SPL rating was helpful for recording some wild gunshots for one of my projects without the slightest hint of distortion.

— Matt Hamilton

STUDIO

The latest news and products

NEW PRODUCTS

HDTRACKS High-Resolution Music Web Store

Created by David and Norman Chesky, HDtracks.com is the world's first high-resolution digital music site offering DRM-free music in multiple formats, as well as cover art and complete liner notes. HDtracks offers the highest quality file formats for any platform or media player. Consumers can choose the file type that best suits their needs: AIFF, FLAC, or 320 kbps MP3. In the near future, select titles will be offered as DVD-Audio quality 96/24 FLAC files. The HDtracks catalog is comprised of many of the finest independent labels in the world, such as the audiophile-oriented Reference Recordings and Chesky Records, David Byrne's eclectic world music label Luaka Bop, and avant-garde imprints such as Mode Records and John Zorn's Tzadik. Additionally, HDtracks offers comprehensive editorial content, with complete liner note scans offered with every album purchase.

PRICE: $1.49 per single

CONTACT: HDTracks | ➲ www.hdtracks.com

JZ MICROPHONES Black Hole PE Condenser Mic

JZ Microphones has launched its third version of the Black Hole dynamic with a pad of -5db and -10db options. The new microphone under the name Black Hole PE (BH-3) has a fixed cardioid polar pattern. As capsule of the Black Hole can handle 135db SPL, this new version can manage also an operatic soprano at close range. Together with the new BH-3, JZ Microphones has also unveiled a new shock mount and pop filter system for the Black Hole. The package also includes a revolutionary pop filter with a curved cone-shaped form. The new Black Hole PE has a five-year warranty.

PRICE: $1,995 ($360 for pop filter)

CONTACT: JZ Microphones | ☎ 371 29994864 ➲ www.jzmic.com

APOGEE ELECTRONICS Duet FireWire DAW Controller

Apogee Electronics' Duet will be sold in select Apple retail stores across the United States. Known for its seamless integration with Apple's Logic Pro, GarageBand and iTunes software, Duet has quickly become a very popular product. It features two channels of Apogee's legendary A/D and D/A converters, two professional microphone pre-amps and a single controller knob that let's users access all of Duet's features with ease. Apple stores to feature Duet in initial distribution are as follows: Glendale, CA; Northridge, CA; San Francisco; South Coast Plaza, Costa Mesa, CA; The Grove, Los Angeles, CA; Third Street, Santa Monica, CA; Topanga, Canoga Park, CA; North Michigan Avenue, Chicago, IL; Fifth Avenue, New York City; Soho, New York City; and West 14th, New York City.

PRICE: $499

CONTACT: Apogee Digital | ☎ 310-584-9394 ➲ www.apogeedigital.com

CASCADE AUDIO Vive Surround Microphone

Cascade Audio has introduced the Vive, a unique, patented surround microphone that produces Dolby Pro Logic and Dolby Digital 5.1-compatible surround sound recordings on any stereo video camera with a mic input jack. Sold as an affordable after-market accessory at B&H Photo/Video, the Vive mounts easily to camcorders to record in true surround sound in a single step. Vive's patented process encodes the signal and records onto any standard media — such as DV, DVD, CD, .WAV or MP3; no external hardware, additional software or "finalizing" steps are required. Vive mounts easily to any camcorder, connects to the auxiliary microphone jack, and is compatible with all Dolby surround sound-enabled receivers in the marketplace. Vive does provide an option for recording a center channel using an additional lapel or handheld microphone.

PRICE: $99

CONTACT: Cascade Audio, Inc. | ☎ 541-505-0055 ➲ www.cascade-audio.com

NEWS

Tony Lindsay, vocalist for Santana and independent artist, isn't packing his **Studio Projects** T3 mic on the current tour with Santana (April 4 – May 4). But whenever he's in the studio, the T3 is with him. "I use it as much as I can," says Lindsay, who first used the T3 while recording his own album, 'Tony Lindsay' (Gruve Records). Since then, he's used

the mic to record his vocals almost exclusively, "unless somebody insists on me using something else," he adds. The Studio Projects T3 is an affordable high end tube mic with a dedicated AC power supply offering completely variable polar patterns.

With design help from David Cherry and George Augspurger, Ward Archer recently rebuilt and rewired a new studio for Memphis, Tennessee's Archer Records — a facility formerly known as Sounds Unreel Studios. The equipment was upgraded to include a 5.1 PMC monitoring system, a 96-channel Pro Tools rig with blackburst synchronization, a 24-track **Sony** analog tape deck, and a 48-channel, all-analog, all-discrete API Vision console with full 5.1 implementation.

Radial Engineering was the recipient of not one, but two "NAMM Best of Show Awards" at January's Winter NAMM show in Anaheim, California. Radial was presented with top honors in two categories: "Gotta Stock It" for the **Primacoustic** Recoil Stabilizer — a nearfield studio monitor platform that both isolates and stabilizes the loudspeaker to produce a more consistent signal — and "Best In Show," given to the entire Radial product line.

by **Steve Murphy**

API 1608 16-Channel Analog Console

A Classic Console is Reborn

The API 1608 console is outwardly modeled on the original API 1604 console, whose production run spanned much of the 1970s and into the early 1980s. These vintage workhorse consoles continue to hold a great deal of value and demand; to wit, many stock (as well as Franken-modified and resurrected) 1604s are still in daily use in tracking rooms, broadcast facilities and mobile recording operations.

Internally, the 1608 can perhaps be described as the child prodigy born of the vintage 1604 console and the more modern, large-format Legacy range. In its stock configuration, the 16x8x2 console ships with 16 input channel strips and faders, 12 550A semi-parametric EQ modules and four 560 graphic EQ modules, a stereo-bus master fader, an eight-input summing-bus sub-master module, four dual echo/send return modules (accommodating the console's eight auxiliary sends and returns), eight unpopulated 500 Series module bays, a complete central facilities and monitoring section, and a full meter bridge with illuminated analog VU metering for all 16 input channels (capable of monitoring preamp output, direct output and sub-master output levels) eight echo return busses (capable of monitoring send output levels) and the L/R main program busses.

API also offers an expansion frame loaded with 16 input channels and the same EQ compliment for $39,900, as well as unloaded (no EQ) versions of the both the console and expander for $35,000 and $25,000 respectively. Unfortunately, I cannot tell you about the automation fader package upgrade for a little over ten grand (rumored to also include a DAW fader-control layer!) but perhaps they will make an announcement soon...

CHANNELING THE FUTURE

Let's start with some generalities: All of the console's generous input and output points are located on the rear panel; all input/return points are actively balanced (excepting instrument inputs) using API's trademark all-discrete 2520 and 2510 op amps, and all output/send points are transformer-balanced; all external I/O connections are on XLR and TRS 1/4-inch jacks with the exception of several DB-25s principally used for multichannel input and monitoring groups;

FAST FACTS

APPLICATIONS:
Studio and project studio

KEY FEATURES:
16 input channels with 212L mic preamps, 12 x 550A and 4 x 560 equalizer modules; 8 aux sends, 8 returns and 8 sub master busses; full stereo and multichannel (5.1) monitoring section; 8 open 500 Series bays, analog VU metering; extensive patch connections on rear panel.

PRICE:
starts at $49,900

CONTACT:
API Audio | ☎ 301-776-7879
➲ www.apiaudio.com

From The Field:
Why I Bought It: The API 1608

By Manny Sanchez, Owner of Chicago's I.V. Lab Studio

I've been actively working in the music business for about 10 years now, working my way up from the lowest of lows as an intern at the Chicago Recording Company. My first experience with professional recording included a Studer A-827 and either a Neve VR72 or an SSL 6000 E, depending on the day. We didn't even get a Pro Tools rig at CRC until it was absolutely necessary in early 2003. Needless to say, I was very lucky to learn a methodology of recording that is not really emphasized in the learning programs of the young engineers that I have met. I never went to an engineering school and I truly believe that I wouldn't be where I am today if I had. Learning from the engineers at CRC gave me a broad perspective of approaches to the and virtually all buttons on the 1608 illuminate when engaged, which proves to look very cool in a dark control room .

At the top of the channel strip – or bottom rather, as the 1608 retains the reverse-orientation of the original 1604 – is a mic/line/instrument input section outfitted with an API 212 preamp and its customary controls: Gain knob, Mic/Line selector, 48V phantom power, Pad (-20 dB mic/-6 dB line) and polarity reverse. The Mic/Line selector button doubles as a peak level indicator by glowing red (regardless of which input type is in use). Mic and line input connectors are XLR, the Hi-Z instrument input is a switching (overrides mic) TS 1/4-inch jack, and the preamp output is available on a TRS 1/4-inch jack.

Continuing up the strip are the send controls that feed the 1608's eight auxiliary busses. Sends 1-4 are configured as mono sends with individual level controls (in concentric pairs) and send on/off buttons (nice!). The remaining four sends are grouped into stereo pairs 5/6 and 7/8, with each pair having a concentric stereo level and pan knob plus send on/off switch. All sends can be placed pre- or post-fader in odd/even pairs. Borrowing a feature from the Vision range, a To Bus switch on 7/8 routes its level to any of the first four sub-masters, where it can then be summed back into PGM (to aid in independent parallel processing tasks).

At the top of the input strip is the routing and output section. Each input channel can be routed to the main stereo program bus (PGM button) and/or directly to any of the eight submaster summing busses (using assignment buttons 1-8). By engaging the PAN button and using the channel's pan pot, also located in this section, the output signal can be placed across a panorama between any selected odd and even summing busses. Note that a channel's post-fader/post-mute XLR direct output is always active.

This section also includes controls for highpass filter engage (FLTR, -3 dB @ 50 Hz, 6 dB/octave) and insert return engage (INS). The half-normaled insert path is placed between the equalizer output and the fader input, and external gear can be connected using the corresponding rear-panel TRS 1/4-inch points. Speaking of equalizers, each channel's preamp output is half-normaled to the input of the EQ module fitted immediately above it; EQ modules can also be patched for alternate use via the rear-panel 1/4-inch TRS EQ I/O points on the rear panel. Since the proportional Q, reciprocal filtering and other features of the excellent API 550As and 560 are well-known and documented, I'll save the space for more 1608 coverage.

An Alps 100mm fader serves as the channel's primary output level control to the direct out, summing busses, stereo program bus and, of course, designated post-fader sends. Immediately above the fader are the channel's Mute and Solo buttons. Two much-appreciated inclusions here are a solo-safe button, and an assignment button to add the channel to the 1608's single master Mute Group (remember no VCAs/no automation). The specific function of channel Solo buttons is determined by the global selection of PFL, AFL or SIP (destructive solo-in-place) on the 845B Central Facilities module Also located on the 845B are the Mute Group master controls, including the thoughtful option to also mute the pre-fader send outputs on channels assigned to the group.

CONTROLLING INTEREST

The 1608's center section provides a wealth of monitoring functions including the selection of three sets of monitor outputs (Main, Small1, Small2), with the Main monitoring outputs supporting up to six channels for 5.1 surround monitoring. Other monitoring features include Mono summing, speaker cut, monitor dim (with dedicated level knob) and a master input selector that offers the choice of

API continues on page 14 ➤

art but little technical training. I've never been taught why an EQ does what it does or how a compressor works. I just had to stay at the studio until the sessions were done, grab a reel of two-inch and begin the experimentation. It's my opinion that learning how to listen is the thing that separates the engineers that spend their time in the studio from the ones who spend their time on audio message boards.

Manny Sanchez,

I feel it's important to share this little bit of my history because I am about to review a product from a vantage point that is fairly removed from the technical. I based my search for a desk on the experiences I have had with them throughout my career. I spent much of my time working with the aforementioned consoles as well as a Neve 8058 and a Trident A-Range. These are all products that have a storied history in the audio world and each have their unique strengths and limitations. I enjoyed the sheen on the top end of the VR, the biting midrange of the SSL, the enhanced bottom end of the 8058, and the overall depth and width of the A-Range. These were all points to consider when settling on a console for my studio in Chicago, The I.V. Lab.

THE PURSUIT OF AUDIO NIRVANA

Since opening about two and a half years ago, I have been searching for a desk that I could afford but that would also allow me to stop spending a majority of a recording budget on a mixing studio. I have worked with Bill Thomas from Mercenary Audio since I got the idea to open the studio, and he has always been well informed and extremely helpful about all my inquiries and purchases. He called me in the summer of 2007 and told me about the API 1608. Bill has an uncanny knack of eliciting enthusiasm, but he didn't have to say much to get me excited about this product; I have worked with API EQs and mic pres, and they have always been very useful in my pursuit of audio Nirvana.

A 16-channel console is obviously not going to be the answer for many engineers making records the way I like to make them, and I was relieved when Bill informed me about the 16-channel expander. This would give me 32 channels as well as eight aux returns. I make records using an Ampex MM1200 16-track tape machine in conjunction with Pro Tools, so essentially having 40 inputs at mix was more than ideal.

FEATURES OF I.V. LAB'S 1608

The console's input module features the API 312 mic pre, of which I already had eight, so I was familiar with (and admired) its sound. The 1608 features eight aux sends (4 mono, 2 stereo), which are useful to me

SANCHEZ continues on page 14 ➤

API Continued From Page 13

three external sources (all six-channel capable), the eight aux busses in stereo pairs and the main PGM out. There is also a dedicated headphone amplifier with on/off switch and under-armrest stereo 1/4-inch jack, as well as a full talkback compliment that includes a built-in mic and T/B To Aux, T/B to All and Slate momentary buttons.

Each of the eight summing bus sub-master sections on the 168B module has separate L and R main program bus assign buttons, a bus on/off switch, a bus Solo button (AFL/PFL), and a Trim knob that provides from 0 to 84 dB of attenuation of its respective balanced (unlike the original 1604) 2520-based active combining amplifier. The 1608 also provides similar control of its main stereo bus summing amplifiers, with individual left and right master on/off switches, Trim attenuators and a stereo insert (PGM INS) engage/bypass.

One of the most impressive and flexible sections on the 1608 can be found in a fairly unlikely place: the E1608 Echo Send/Return modules. The echo VU's read echo sends. The VU Return button allows monitoring of the echo return signal.

Each echo return input provides a full output and routing section similar to that found on the channel strip modules, with eight assignment buttons for routing to the sub-master busses and a PGM button for assignment to the main stereo bus, plus a Pan engage button and knob to enable L/R panning across the main L/R busses and odd/even sub-master bus pairs. A return output level knob is also provided along with Solo, Mute and Safe buttons identical in function to those on the full channel strips. Additional Aux and Mix buttons select as the return's input a corresponding rear-panel auxiliary input (on TRS 1/4-inch and DB-25) or the output of the corresponding send bus, respectively.

If the possibilities for this section (and the console in general) aren't already swimming about in your head and expanding exponentially, let me help you along. Not only are these eight mild-mannered "echo returns" really Super Inputs with multiple switch-selectable input sources and full channel-strip solo/mute and routing control, they are also – Tada! – normaled to the console's eight open 500 Series slots. Add in eight 512C preamps and, well, you can guess the rest...

ONCE AROUND THE BLOCK DIAGRAM

The API 1608 is endowed with refreshingly unfettered internal routing and external patching facilities, leaving the user is free to create, configure and reinvent how the console is best used for any immediate purpose or to adapt it over time as needs change. While the use of the 1608 can be as simple and straightforward as desired, some more creative options are certainly possible. For instance, by patching the console's preamp outputs directly to a DAW's inputs and the returns from the DAW into the EQ inputs, the 1608 becomes a simultaneous high-end API multichannel DAW front end and a full-featured multichannel summing mixer with all EQ, sends and sub-master features available for the mix. Likewise, by putting the board's sub-master outputs and routing capabilities to good creative use, multichannel surround mixing is easily possible, complete with built-in, fully calibrateable surround monitoring.

SUMMARY

When I first heard about the API 1608, I couldn't help but wonder what corners were cut and where the skeletons were buried to bring out an API console at under $50k. As I delved deeper into the product literature

SANCHEZ Continued From Page 13

because I still use a bunch of old school effects and reverbs including a tube version of the EMT 140 plate. My cue system features a 12-channel mixer for each performer, so in a tracking situation I will use the eight sends on the first eight tracks of the mixer and fill the rest with the multi-track return row of my patch bay, if needed. This gives the artist maximum flexibility, and I never have to worry about trying to balance two cues across five people — never an enjoyable experience.

The console has eight busses, which is perfect for me in a mixing or tracking situation. I have learned over the years that spending more time getting the sound shaped and confirmed at the tracking phase will lead to a more efficient mix. I blend multiple kick and snare mics as well as guitar mics at tracking, so I don't have too much to think about at mix. In a mixing situation, I am a big fan of buss compression on drums, and eight of them are more than enough for my needs. This may seem futile to praise such a simple and functional aux/buss system, but anyone working on a vintage console — especially an 80 Series Neve — knows how frustrating it is to exercise switches when you should be working.

There is a 500 Series slot for each channel plus eight additional 500 Series slots in the center section. This is the area where the console really shines. The modular concept is flexible and fresh. I currently have twelve 550a, four 560, two 550b, two 525 and two Speck ASC-V EQ modules. I took the option of short-loading the expander bucket because I am very interested to see the next generation of 500 series modules. The API EQs have been around a long time and are tested and true. I love the idea of being able to add different flavors of EQ and compression into this desk to mix and match the sonic properties of a given channel. I plan on filling the rest of the expander bucket with EQ and the additional 500 Series modules with compression. The character possibilities of the desk aren't limited to API, which was a positive selling point.

NOT INLINE, NOT 'BUDGET'

On the negative side, I could see how people could consider the 16 channel version limiting because it is not an inline console. I don't ever have a problem because, with the extra bucket, it is easy for me to split the console into input and monitor sections. The console doesn't currently have automation, but Bill at Mercenary informs me that it will probably be ready in the fall.

This also isn't a budget console by any means. In addition to the near $75k price tag (short loading the expander of EQ), a patch bay must also be built. This is not to say the 1608 is overpriced, because it isn't. But it is a serious investment.

Overall, since receiving the first console in February, my experiences have been what I expected. It's the beginning of a love affair that has been unrequited for some time in my audio career. I would put any of my recent mixes on the 1608 up against any I have done elsewhere on much more expensive and "vintage" consoles.

The sonic quality of the 1608 is as good (and better in some areas) as other consoles on which I've had the pleasure of mixing. It is a very balanced sounding desk that can kick you in the stomach but can also blow sweetly in your ear. Engineering is art and the API 1608 is a wonderful canvas.

and block diagrams, and as confirmed first-hand on my visit to API HQ, not only did I not find cut corners but on the contrary, I was repeatedly impressed with the 1608's expansive and imaginative features: a full compliment of balanced I/O points along all audio paths, full access to individual channel circuit components, adjustment pots at all the important calibration points, extraordinary internal routing options, and an expansive central configuration and monitoring control section.

These are the hallmarks that make large-format – and large-investment – consoles a joy to use and a relative breeze to maintain. They are essential elements in the hub of a busy, multi-purpose commercial studio and the first things I expected cut to meet such an attractive price point. To say I was pleasantly surprised to find them at all – let alone in such abundance and in such a thoughtful implementation – is a great understatement.

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*


An API 1608 at Overdub Lane Studios

by **Russ Long**

TASCAM X-48 Digital Audio Workstation

Dedicated hard disk recording and DAW capabilities embody SaneWave-designed workstation.

The TASCAM X-48 is a stand-alone 48-track hybrid hard disk workstation. The machine integrates the stability, robustness and ease-of-use of a stand-alone hard disk recorder (like the iZ RADAR) with the GUI, plug-in compatibility and editing features of a DAW.

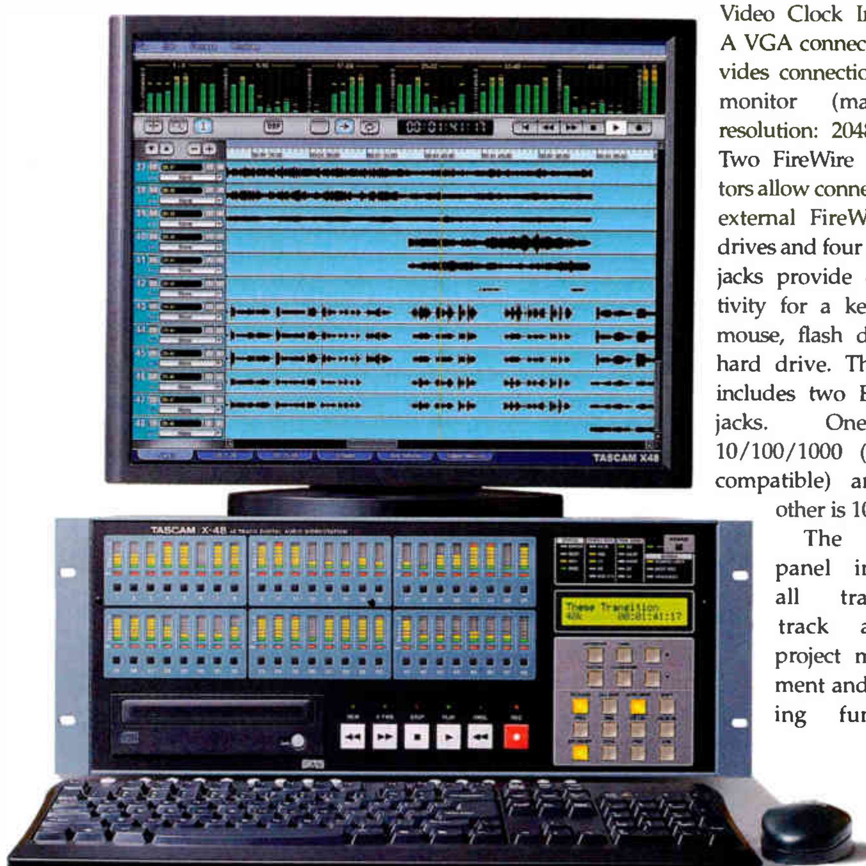
The X-48 features simultaneous recording at up to 96kHz/24-bit across 48 tracks. It supports native Broadcast WAVE audio files and AAF audio file export insuring compatibility with Pro Tools, Logic and other DAWs. Additionally, the X-48 features a built-in, automated digital mixer, VGA monitor output, editing features, an internal 80GB hard drive and a DVD+RW drive for backup. The box supports FireWire hard drives as well as Gigabit Ethernet simplifying transfers between multiple X-48s and/or computer workstations.

| FEATURES

TASCAM developed the X-48 along with SaneWave who has been a primary developer of several pro audio products over the last few years, including the TASCAM US2400 and the Mackie dXb consoles. SaneWave pioneered the development of embedded PC hard disk recorders with built-in GUIs. In the case of the X-48, they set out to build a powerful recording solution that included editing features and plug-in support without the expense incurred with a standalone DAW. The X-48 is built around an Intel processor running the X-48 system on top of Windows XP Embedded OS. The machine's DSP and mixing functions are performed in kernel mode for increased processing efficiency and near zero latency performance.

The 4U X-48 is 19 inches deep and weighs just over 30 lbs. The basic X-48 includes 48 channels of TDIF-1 input and output. Two card slots that can be fit with either the IF-AN24X 24-channel balanced analog I/O card, the IF-AE24 24-channel AES/EBU I/O card, or the IF-AD24 24-channel ADAT optical I/O card allow the box to be configured to specifically meet the needs of the end-user.

The rear panel of the X-48 includes S/PDIF I/O (via two RCA jacks), SMPTE LTC Input and Output (via 2 balanced 1/4" jacks), and MIDI Input and Output. Unfortunately there is no AES/EBU or optical I/O. The machine generates and reads MIDI Timecode and it supports MIDI Machine Control commands. A quarter-inch momentary footswitch jack provides remote punch in and out. The Remote connector is compatible with RS-422/Sony 9-pin edit controllers for machine control. The machine includes PS/2-compatible mouse and keyboard inputs (alternatively, the box will support a USB mouse and/or keyboard). BNC connectors are provided for Word Sync In/Out/Thru and Video Clock In/Thru. A VGA connector provides connection for a monitor (maximum resolution: 2048x1536). Two FireWire connectors allow connection to external FireWire 400 drives and four USB 2.0 jacks provide connectivity for a keyboard, mouse, flash drive or hard drive. The X-48 includes two Ethernet jacks. One is 10/100/1000 (Gigabit compatible) and the other is 10/100.



The front panel includes all transport, track arming, project management and metering functions. There are seven segment track meters for each of the 48 tracks. Six segments display the signal level from -60dBfs to -1dBfs and an additional LED indicates overload. Each track also includes a track record arming button for arming the track. Four status lights (Error, Busy, MIDI, and Disk) flash when the X-48 is accessing the hard drive, busy with a task, has MIDI input or encounters an error. The Sample Rate

TASCAM continues on page 18 ➤

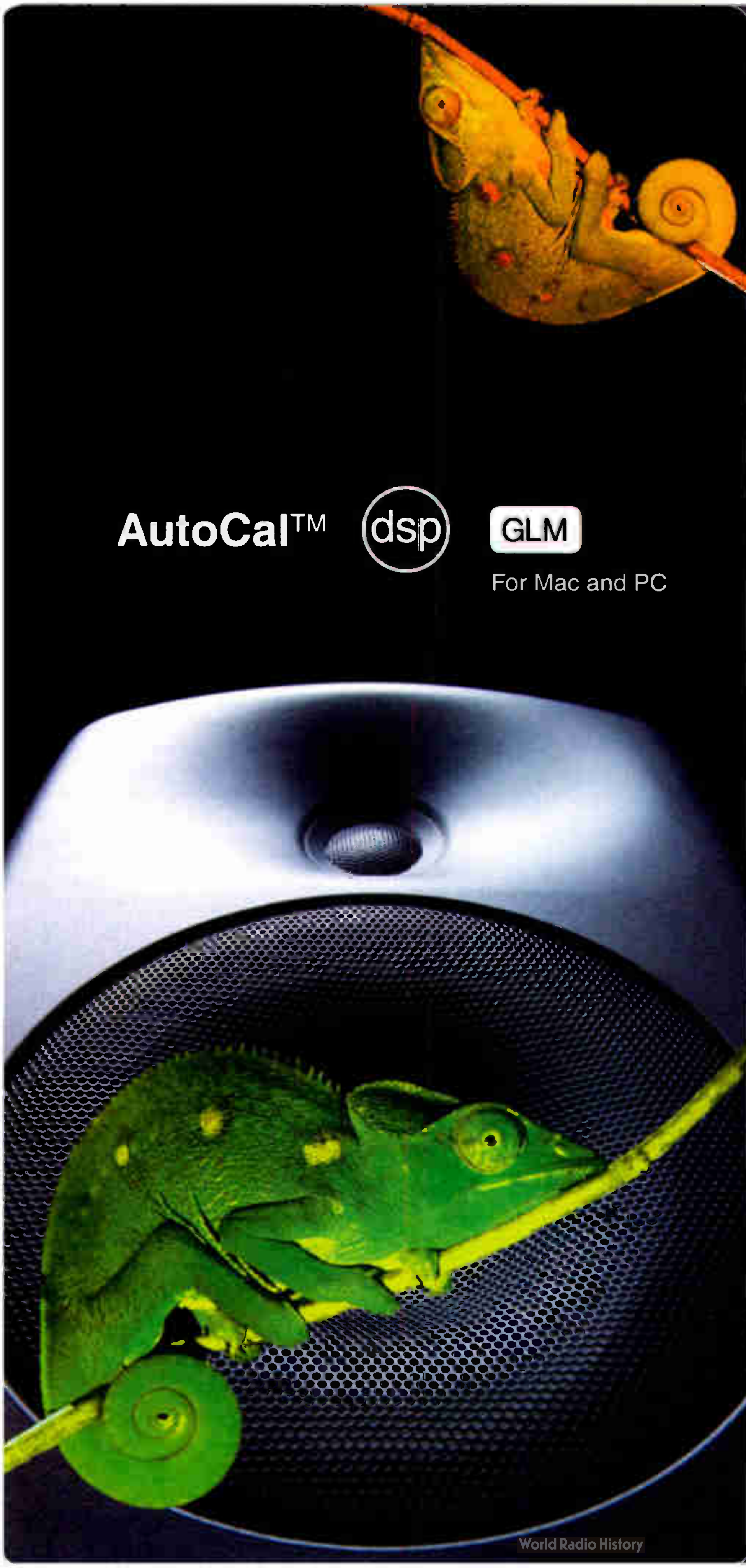
FAST FACTS

APPLICATIONS
Studio, project studio, broadcast, post production, location recording

KEY FEATURES
48-tracks of 24-bit 96kHz recording; FireWire capability

PRICE
$3,995.00 base

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



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

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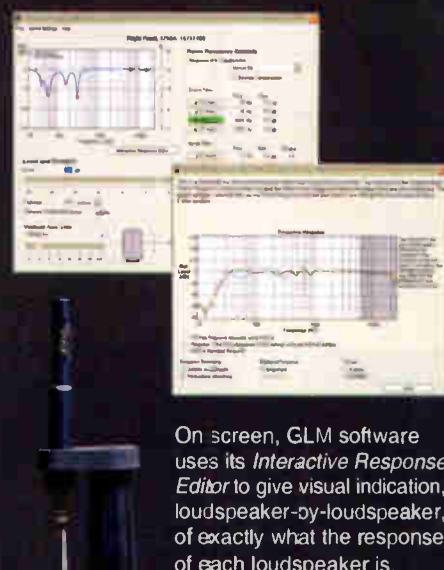

On screen, GLM software uses its *Interactive Response Editor* to give visual indication, loudspeaker-by-loudspeaker, of exactly what the response of each loudspeaker is.

In 1978 Genelec brought active monitoring to the professional audio world. An essential part of our active design is the room response controls. They are included in every Genelec analogue model to help integrate them to the listening environment. To further this, Genelec Product Specialists travel the world providing system calibration services to ensure optimum monitoring performance for our large system customers.

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TASCAM Continued From Page 16

lights indicate the current sample rate and the Timecode Rate indicator displays the current frame rate. System Lights (Sample Lock, Dest Rec, and Varispeed) illuminate to indicate sample lock to an external source, destructive recording mode enabled, and varispeed enabled.

PRODUCTPOINTS

- Great sound & features
- Ease of use
- Low price

- No headphone jack
- No AES/EBU or optical output

SCORE

The X-48 should be a serious consideration for any professional needing a multi-track hard-disk recorder.

In comparison to other DAWs, the X-48's mixing features are fairly limited (e.g. the faders are the only function that can be automated and the automation has to be drawn in with a mouse) but they can still adequately handle most situations, especially since most people will be using the mixer for reference mixing. The X-48's 32-bit floating-point mixer features include grid-style editing and varispeed (+/-6%). Each channel includes Dynamics, 4-band parametric EQ, and 4 VST plug-in inserts. The dynamics section includes controls for threshold, ratio, attack, release, and makeup gain as well as a button to activate the soft knee mode for extreme ratio settings. The Equalizer section includes a four-band full-parametric EQ. Each band has full Q control and is sweepable from 20Hz to 20 kHz. Each band and can also be set to Low Shelf, High Shelf, Peaking, Low Pass and High Pass. The knobs can be turned to make adjustments or you can simply grab the dots in the graphic display. The X-48 officially supports Antares Auto-Tune and the Waves plug-ins, though most VST plug-ins should work trouble-free. There are also buttons for solo, mute, record arming, and input monitoring as well as pan controls.

In addition to the 48 audio tracks, the mixer includes six stereo returns providing a total of 60 inputs at mixdown. The dynamics and EQ sections of the digital mixer sound good and I found that in both instances it was quick and easy to attain my desired results. There are also 12 stereo groups, and six aux sends and a stereo master buss. The TASCAM US-2400 can be used as a control surface for the X-48 providing physical access to the X-48's transport, fader levels, pan, aux sends, bank select channels, etc.

IN USE

I've always been a fan of TASCAM's converters. Back in the day of DTRS machines I always found the TASCAM boxes to be leagues beyond the competition. At first we all thought that digital sounded terrible (it turned out that ADATs sounded terrible) but there actually were some good sounding converters out there, even in the 90's. All that said, I really like the way the X-48 sounds.

The most obvious use for the X-48 is recording live concerts. This is the situation where it truly shines. I can walk into a venue with a rack of 48 mic preamps, the X-48, a TASCAM DV-RA1000HD or Alesis Masterlink and a pair of headphones and walk away with a live album ready to be mixed. If TASCAM had included a headphone jack on the box, I could do it without the DV-RA1000 or the Masterlink.

When recording longer shows with high track counts, especially at higher sample rates, hard disk space becomes an issue with an internal drive of only 80GB. The X-48 flawlessly records to external hard drives; I have been using a 500GB Glyph GT050Q drive and haven't had a single problem. I recorded 48 tracks at 24-bit, 96kHz for over two hours and the rig never hiccupped. I did have an issue with another hard drive while recording a David Phelps live Christmas DVD that turned out to be a faulty hard drive. Interestingly enough, I ran the Drive Benchmarking application on the drive and it shows that the drive was fine; this makes me question the quality of the X-48's application, but I haven't been able to duplicate that problem so I'm not exactly sure what the issue was.

I love the fact that the X-48 records Broadcast WAVE files. This makes it simple to import recordings into another DAW for editing and mixing. Projects can also be exported

as AAF files which retain automation, pan and level settings. The X-48's editing screen uses a familiar DAW interface that includes snap-to-grid editing and crossfade features. I found that the 32-bit floating-point resolution mixing within the X-48 resulted in a great sound. As someone that is used to automating everything — from compressor thresholds to aux sends to panning to reverb times — I felt fairly limited, though. I wouldn't use the mixer to do a final mix but it works perfectly to create a mix from a live show for video people to use as a reference for their editing. I'm sure there will be some users that will find themselves right at home mixing on the X-48; having a complete turn-key system will be great for them.

The machine will work well as an analog machine replacement in a normal tracking studio situation. I like the feel of recording to a purpose-built recorder instead of to a computer adapted for recording. Aesthetically, TASCAM and SaneWave did a fine job with the X-48. The screen layout is logical and the meters are easy to see from across the room.

Another strength of the X-48 is as a playback machine for theater, live performance and presentation applications. The X-48's Theatre Mode allows one audio section to be played at a time while automatically cueing up the next section for instant playback. This mode was intended for theaters playing back multiple music segments and/or sound effects and with the X-48; cueing the next scene is as simple as tapping a footswitch.

| SUMMARY

The TASCAM X-48 48-track recorder provides a high quality audio workstation with the GUI, editing features and plug-in compatibility of a computer-based digital audio workstation, all contained within a single box. The machine provides a fast and smooth workflow with quick and easy backup. Anyone considering a multi-track hard-disk recorder should give serious consideration to the X-48.

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

| REVIEW SETUP

Focal Twin6 monitors; Yamaha MSP10 monitors; PMC AML-1 monitors by Heather Johnson

SSL Matrix SuperAnalogueConsole

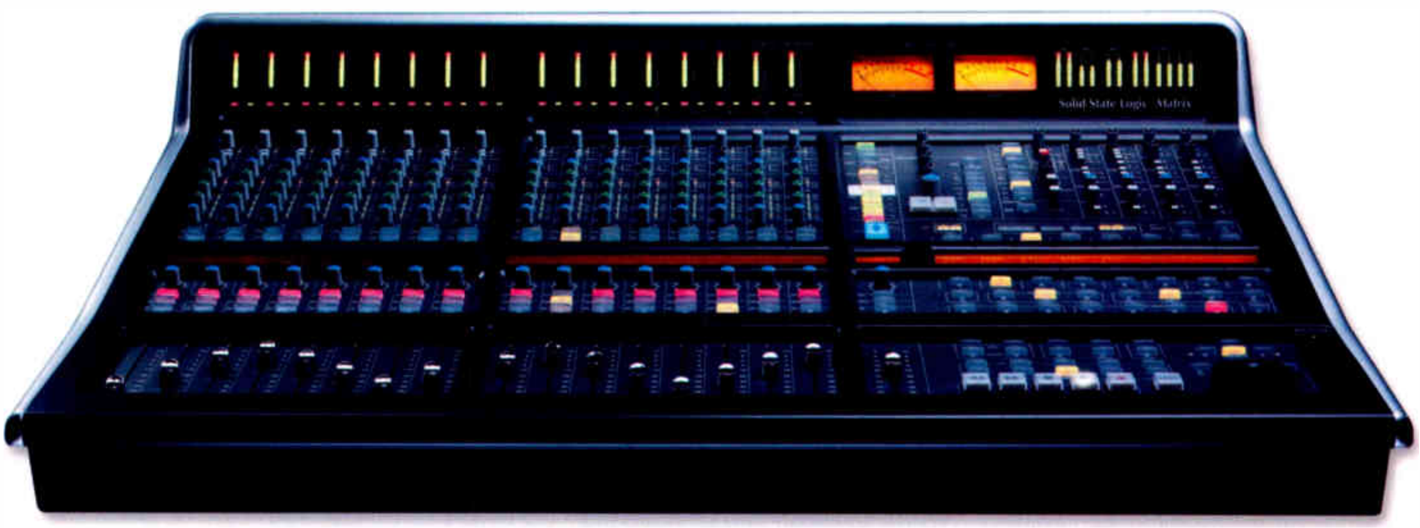
Solid State Logic roseraised more than a few eyebrows at this year's Frankfurt Musikmesse and subsequent Las Vegas' NAB Convention as they announced Matrix — a compact 16-channel, 40-input mixing console with built-in SuperAnalogue signal router and multi-layer digital workstation control. The small-format desk turned heads for its ability to integrate the analog and digital domains, allowing an engineer to incorporate his existing racks of boutique signal processing in a DAW-based studio environment.

Matrix doesn't come with a lot of frills, because it's designed for recordists who don't need, and don't need to pay for, a lot of extras. Instead, SSL designed Matrix as a high-quality hub from which engineers can incorporate the mic pre's, EQs, and other effects that they already have with their existing workstation, and operate the whole lot from one set of automated faders. Aside from its function as a central nervous system for hybrid recording, Matrix can serve as a true analog SSL recording and mixing console. And its price tag ($25,995 MSRP) makes it a relatively affordable alternative to a more fully loaded desk, provided one has the gear to support it.

Due to its myriad functions, Matrix can benefit a variety of recording environments, from the video game composer to the recording musician to the established engineer or producer to the mastering engineer. Its small size (947 x 724 x 218mm) and flexibility make it suitable for project studio and 'B Room' environments, while its light weight (about 55 pounds) makes it suitable for portable touring rigs.

"When we discussed some of our existing products with producers, engineers, and other customers, we would hear things like, 'Well, the contentpackage is fine, but I don't need everything that this system provides,'" says SSL Director of Product Marketing Niall Feldman. "They had built up a collection of their own equipment that, in many ways, defined their signature sound if you will. These people also moved around a lot, so the concept of a fixed installation console wouldn't suit the way they did much of their work. Operation with workstations was also quite important. So we developed Matrix as a tool that would integrate all of the pieces of equipment and the workflow that these producers and engineers had to work with."

The applications for Matrix extend beyond traditional music production, however. Post production sessions, particularly Foley and sound effects recording, can benefit from Matrix's efficiency. Those types of sessions don't require a huge number of inputs, compressors, or EQs," says Feldman. "They need a small number of inputs with high audio quality and workstation control." A music composer who works in a mostly solitary, computer-driven environment would appreciate Matrix's ability to control multiple workstations simultaneously.



MATRIX AT THE HEART

Just as the Matrix itself serves as the heart of its recording environment, Matrix's SuperAnalogue router serves as its central and most unique feature. The 32 x 16 x 16 routing matrix manages the sends and returns of up to 16 analog outboard processor units together with the assignment of the console's channel strip inputs and insert points. By using the Matrix Java control panel software, any combination of outboard gear can be inserted into any of the console channels. A precious tube compressor can then act like a plug-in, but still sound like a precious tube compressor.

"The name Matrix comes from the fact that it has this routing matrix at its heart," says Feldman. "The SuperAnalogue router allows you to route your outboard gear to the inserts of the Matrix console. The Matrix itself is controlled from a Java application that runs on any PC or Mac. The Java control panel software gives you a visual, graphic way to control the router, so if you want to insert your LA2A on channel three, you open a window, select the insert for channel three, pick the LA2A from a list, and the job's done. That then could get saved as part of the project settings to be recalled later."

An engineer can follow the same process to select and save outboard chains. "With one recall of a preset you can set up that chain instantly from the control application. Matrix would be a good analog mixing console with workstation control even if it didn't have the SuperAnalogue router, but that feature really

SSL continues on page 22 ➤

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- 2-channel WAV (BWF) and MP3 recording and playback
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- balanced ¼" TRS inputs with line inputs and 48V phantom-powered mic preamps
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Includes stereo electret microphone, earbuds, 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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SSL Continued From Page 20

sets its apart. In a workstation, you have digital plug-ins. In many ways what we offer with Matrix through this routing feature is an analog plug-in type architecture. It turns your outboard gear into analog plug-ins that you can operate from the console."

With HUI or MCU modes, the 16 motorized faders and channel controls allow the user to mix using just the control surface. From the desk, the user has access to transport and navigation functions, plug-in and virtual instrument parameters, and essential DAW commands.

Producers and engineers who work primarily 'in the box' will appreciate Matrix's ability to control up to four DAWs simultaneously. For example, the video game composer can switch between Logic, Cubase, and Pro Tools with a press of a button, using Matrix as the control surface.

"[Propellerhead's] ReWire technology will allow you to connect the transports and stream audio between the DAWs, but by having simultaneous workstation control, you can, select Ableton Live orfor example, select Reason as a workstation control options, and your faders will then control your AbletonReason mixes. Press another button and you're back in your Logic mixer. It's very fast way to work with several different applications simultaneously."

The Matrix claims 16 faders, with one additional "Focus Fader" mode (and accompanying V-Pot) that proves useful for a variety of functions. "Maybe you want to ride the vocal level as you're listening to the mix," Feldman explains. "That extra fader can be assigned to control a different parameter to the 16-faders on its left. So the additional fader gives you an extra level of control. It's like a wild card that you can assign to a variety of different tasks."

Other convenient features include an A/D converter available as XLR/AES format and S/PDIF optical TOSLINK connector. "The A/D converter provides a very pure monitoring power from the computer," says Feldman. The digital I/O can be selected via the front panel to be fed from the REC bus, Mix bus or the Pre Monitor pot signal. The input can be selected from the front panel's programmable softkeys. The softkeys allow a user to program essential DAW commands, up to 350 per DAW layer.

Another proprietary feature called the iJack allows the user to plug an iPod or other portable device right into the console. "Quite often people have ideas they've put onto an MP3 player or an iPod," says Feldman. "This allows you to plug it straight into the front panel, call it up on monitoring, and then listen to it right away."

MATRIX AS A MIXER

Routing and controlling aside, Matrix can act like a high-quality SSL mixer when it wants to. Each of the 16 SuperAnalogue channels offers two inputs per channel, a dedicated Channel Output, Stereo Aux Send and four mono Aux Sends. Dual Stereo Mix Busses and four Stereo Returns with full stereo mix bus routing bring the total available mix inputs to 40. Matrix also features independent Main and Mini monitor outputs, an Artist Monitor output section with independent EQ and source selection, and three External Monitor inputs with source summing.

Keep in mind, however, that Matrix doesn't come with inboard mic pre's and EQs. Matrix users would need to either integrate their own or add on one or more of SSL's XLogic channel strips. "Our idea was to create a tool that would allow people to make better use of their personal tool kit that they've already developed," says Feldman. "Another reason was pure economics. People said, 'I've already spent thousands of dollars on all of this boutique gear. I'm not going to get rid of it. It's the focus of my workflow.' Matrix took those considerations to heart."

For those who are still building their tool kit, SSL offers several options. "An engineer can purchase one or two X-Racks and have a nice powerful production system," says Feldman. "In the X-Rack system we have mic pres and EQs of a couple different flavors, we've got dynamics units, and a bus compressor. You can build close to a small SSL console entirely from X-Racks. Obviously you get a more production-focused system by combining that with a Matrix. That's one of our ground-up solutions."

This interconnectivity — or "Logictivity," as SSL describes it — is one of the company's key concepts in product design. "By allowing the units to talk to each other, we can add a bit more value, so if you had an entirely SSL solution, you could conceivably do more than if you were using a rack full of other boutique pre's and EQs."

And while there are parallels between the products, each serves a different purpose. It would be easy to describe Matrix as a stripped-downsmaller, simpler AWS900, but Matrix is actually more complex than that. "The"At launch, AWS900 was a pioneering product in combining high quality analog audio and digital workstation control in a single unit," says Feldman. "And Duality, its bigger brother, and Matrix, its younger brother, are all parts of the same family. You can see that there's a DNA resemblance in the three family members. But their feature sets are quite different."

The SSL Matrix ships in May 2008. For more information on Matrix, visit *www.solid-state-logic.com*.

Heather Johnson is a San Francisco-based journalist and author whose books include "If These Halls Could Talk: A Historical Tour Through San Francisco Recording Studios."

by Steve Murphy

Toft Audio Designs Series ATB Analog Console

Ok kids, gather around while I tell you the tale of The Little Console That Could...

The model designation "Series ATB" that Malcolm Toft bestowed upon this ambitious, affordable and impressively adorned analog console line draws a less-than-subtle connection to the 1980s-era Series 80B large-format analog console by Trident Audio Designs, the respected UK console company Toft founded back in 1972. While I can't imagine anyone is under the illusion that the project-studio-priced ATB console range is the next generation of any of the storied Trident console lines (and perhaps the factor-of-10 relative reduction in cost from the last Series 80 to the ATB was the result of a clerk's pricing gun error?), there's no way to miss the clear influence of the Series 80 pedigree. The ATB does include several large-format features and provides plenty of bang for the buck, but what I like equally well is that its has great potential to reinvigorate the semi-stagnant eight-bus market with its unique personality.

FAST FACTS

APPLICATIONS
Studio, project studio

KEY FEATURES
16, 24 or 32 channel strip input modules; six aux busses; eight submaster groups; EQ derrived from the Trident Series 80; full in-line monitoring; modular per-channel PCB construction with through-hole and socketed components for easier maintenance

PRICE
$4,499 (16-channels), $5,499 (24-channels) and $6,999 (32-channel)

CONTACT
PMI Audio | ☎ 877-563-6335
www.pmiaudio.com

| FEATURES

The Toft Audio Designs Series ATB is an in-line, eight-bus console available in 16-, 24- and 32-channel frame sizes (listing for $4,499, $5,499 and $6,999 respectively). Unfortunately for the manufacturer and board owners alike, both options have suffered delays and are not available as of this writing. From what I can gather, the initial run of meters is not far off, but it seems that an OEM issue on the Digital I/O has scuttled the effort to date and the company is in the process of developing an alternative.

The console has 100mm mono faders for all input channels. To the right of each channel fader the routing assignment buttons and simple two-stage LED metering providing signal present and nearing-peak indication. Immediately above the channel faders are a center-detented pan knob, mute button with LED and a solo button. The ATB ships globally configured for non-destructive AFL (post-fader, post-pan) soloing, but a switch located internally on each channel's circuit board moves its solo bus tap to pre-fader (PFL).

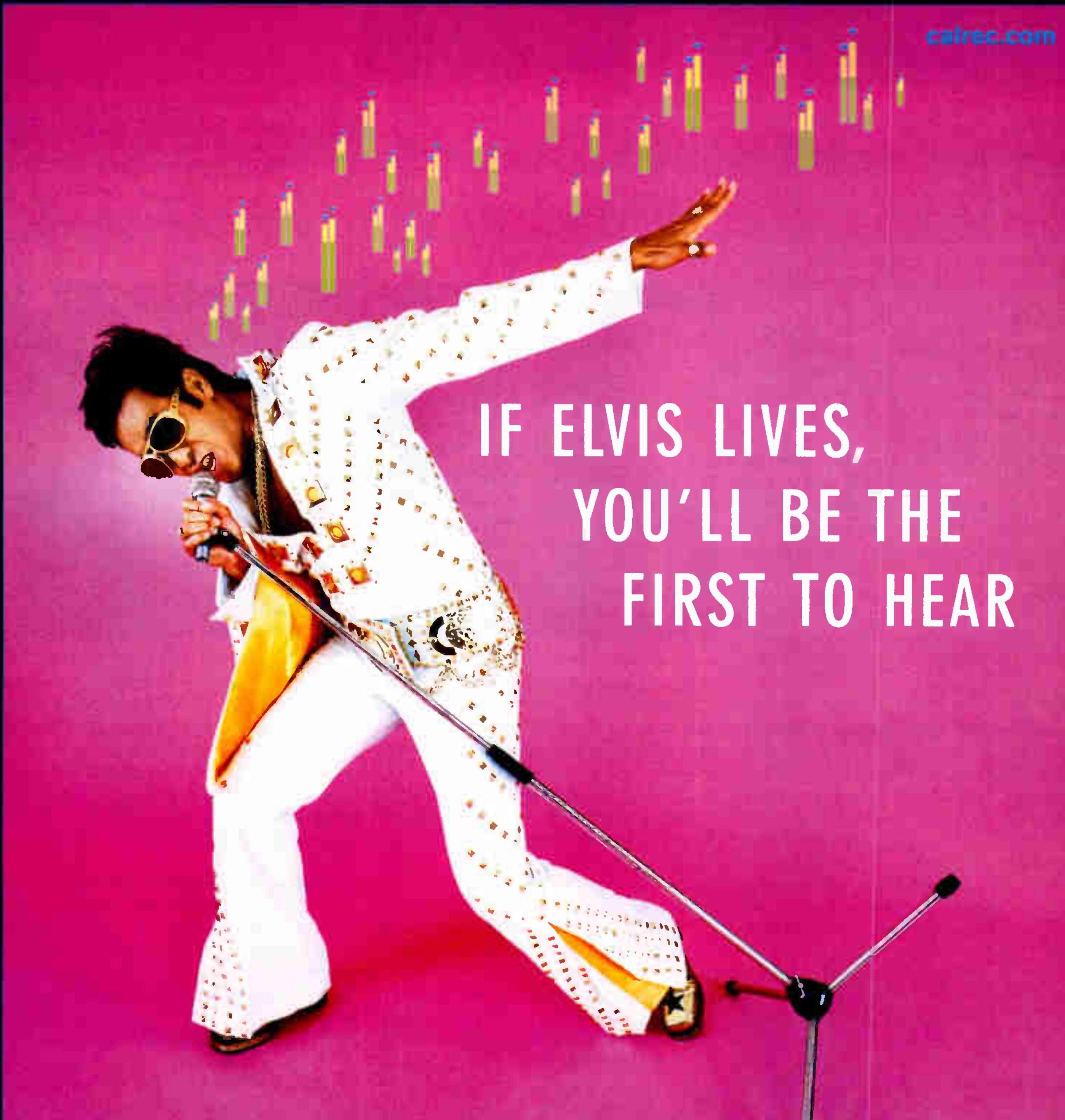
The preamp input section at the top of each channel strip provides controls for gain, mic/line selection, 48-volt phantom power and polarity reverse. An input reverse button swaps the line input and monitor section input signals, the primary purpose of which is to allow use of the full set of sends and the EQ section for mix downs with a minimum of board or patch reconfiguration.

Certainly one of the star attractions of the Toft ATB is its equalizer section modeled on the original Trident Series 80B. Its four bands – each capable of +/-15 dB of gain change – are comprised of two shelving filters that offer a choice of two fixed-frequency turnover points each (8/12 kHz, and 60/120 Hz), and two semi-parametric mid-bands with slightly overlapping frequency ranges (100 Hz to 1.5 kHz and 1 kHz to 15 kHz). Also provided are an 80 Hz high-pass filter and EQ section bypass switch.

The full EQ section can be moved into the channel's inline monitoring path via the "EQ to Mon" button adjacent to the monitor rotary fader, pan and mute controls. Likewise, the "Aux 5/6 to Mon" button allows the pair of aux sends to be placed in the monitor path (pre/post selection conveys to the monitor fader).

There are six mono auxiliary sends available on each input channel, with sends two through six pre-/post-fader selectable and send one fixed at pre-fader. The six send bus master level controls are located at the top of the central control section and the corresponding send output jacks (balanced 1/4-inch) can be found on the rear panel along with the rest of the console's I/O connections. The ATB's generous brace of eight stereo effects returns, each with a stereo level and pan knob plus mute button, are located just under the

TOFT continues on page 26 ➤



When you're mixing sound for light entertainment, it's best to expect the unexpected. You need to set up and handle multiple sources quickly and easily, access pre-sets, network your i/o and instinctively control the whole thing without missing a beat.

At Calrec we've been dedicated to live production and on-air broadcast audio mixing since the first Elvis impersonator wiggled his pelvis. Today, as well as all the practical functions you could want, we also offer the most innovative and economical solution for surround sound.

Bluefin technology provides twice the signal processing capacity in a fraction of the space of conventional systems, at no extra cost and with 100% redundancy.

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CALREC

Putting Sound in the Picture

TOFT Continued From Page 24

aux output masters. Signal input to the returns is via stereo 1/4-inch jacks; all return outputs feed into the main L/R bus summing amplifiers. Under the eight echo returns are the eight submaster controls, each with a 12-segment LED bar graph display, aux 5 (pre/post) and 6 (post) sends, solo switch, pan knob and fader.

The ATB ships globally configured for non-destructive AFL (post-fader, post-pan) soloing, but a switch located internally on each channel's circuit board moves its solo bus tap to pre-fader (PFL).

The master section of the ATB is where you'll find the two-track input source selector (for two stereo analog sources, and a third reserved for the digital I/O option), solo master level, the headphone jack and level control, talkback controls. A set of stereo analog VU and LED bar-graph meters monitor the the main bus signal leaving the master module's 100mm stereo fader; the main bus monitoring knob adjusts the control room volume of up to two speaker sets.

| IN USE

I had been waiting patiently for an ATB console to review ever since I spotted it on display at the NYC AES convention – in 2005. I had no doubt it was going to be in demand, and my enthusiasm to review one (due in part to the fact that I started my professional engineering career with several years on a Series 70) was obvious. So I did what any self-respecting engineer-disguised-as-writer would do: begged Mr. Toft to let me get the jump and take the display unit to review. He politely but firmly rejected the idea with some vague excuse about "prototype" and "completely devoid of electronics." It's OK...I can take a hint.

But then an amazing thing happened: shortly after unpacking the ATB – all dollied up in its handsome wood trim and row after row of colored anodized knobs in the old Trident style – well, let's just say all is forgiven, Malcolm.

I spent several weeks working with an ATB 16-channel model, and have a pretty good grasp of the big picture at this point, so let's start there: This console is going to make *a lot* of people happy. And it may leave a few people frustrated, particularly when expectations and intimate familiarity with large-budget console features don't match the ATB reality. The trick, of course, is to be realistic and mind where you fall on the trajectory.

I think most engineers, no matter what experience level, will be quite impressed by the overall concept of a familiar large-format console distilled into a commanding, small-format package that provides better fidelity, more functionality and gobs more wow-factor than its cost suggests. That is the conclusion I came to after putting the console through a variety of tracking and mixing tasks, and after exploring every possible way to mate the ATB with my Nuendo system.

It was during the latter that I was guilty of trying to turn the ATB into something it wasn't. I prefer to use the DAW as a multitrack tape machine with the insert loop just after the preamp outputs. That yields the most direct path to tape, and full use of the channel strip features on playback. The ATB doesn't have preamp output jacks, so that wouldn't work, but it does have an insert point but its after the EQ and its the unbalanced stereo-jack type. I went through every other gyration possible, but at this point it was more to have fun exploring console and the limits of its flexibility. Ultimately, using the ATB in the manner that it

| A TALE OF TWO CONSOLES

Can't we all get along? I mean, if two the two gentlemen of Torquay, England can't bury the hatchet and move on, what hope is there for political reconciliation in the States, or for lasting peace in the Middle East?

OK, maybe my sense of perspective is out of whack, but that's what happens when you get caught up in audio forums reading thread after bitter thread on Malcom Toft's ATB console vs. John Oram's 8T console – both of which, you may note from their tricksy model names, are inspired by the large-format Trident Series 80 consoles from that comparatively simpler and harmonious era – the early 1980s. The dust cloud surrounding these two consoles, and fanned exponentially in the ether by, shall we say, very committed forum posters, is unfortunate and most certainly unnecessary.

I've had the Toft ATB and an Oram 8T in my control room for several weeks for comparison purposes and you know what? These are both excellent boards for the intended markets and applications, and the for value vs. price. Sure, there are differences in features. Potential purchasers should look very closely at how each meets their needs when it comes to incorporating them into an existing studio setup and work flow. Other differences have some bearing: the Toft is made in China and the Oram is made in the UK; but the Oram uses surface mount comonents and a more monolithic approach, while Toft uses a more traditional, modular PCB approach and through-hole components. As someone who is capable of circuit mods and repairs, this latter point would win out for me (assuming all other things being equal), but the majority of users will be happy with either. The boards do have some differences in sound and character, but again, they are not even close to "night and day" but far, far more subtle.

Far too much time and space has been taken up endlessly debating what essentially comes down to personal preference, and far too much emphasis on pedigree, who did what on which circuit on the original Series 80, which has this component or EQ width, and on... Far better to be discussing making recordings and music and creative applications. As it turns out, both boards can be used very effectively to do just that!

– Steve Murphy

was obviously designed – as a channel path to monitor path in-line, or by sending submixes back from DAW to open input channels or the subgroups. In fact, this latter method was great for tracking as the eight submasters have secondary inputs that are normaled to the first eight input channel strips.

As for fidelity, its overall throughput is certainly as good as the other conspicuous consoles in this market segment, and it does offer a measure of its own Trident pedigree in the form of what one may call "warmth" and roundness, but is of course most likely the result of multistage THD and EQ phase shift. Nothing necessarily wrong with that, as they are the DNA building blocks of much analog "character." The high-frequency detail was also very good, and didn't exhibit any irritating graininess.

The fairly faithful EQ section of major value to the console's ultimate enjoyment. I tended to be a bit shy of the high- and low-shelves, however. The danger is, with only the two sweepable bands, it's natural to reach for the high shelf for brightening, but do that on too many channels at the same fixed point and you may find yourself starting a mix from scratch. This was much more of a problem on the Series 70, with its bands fixed at (from memory...) 100 Hz and 10 kHz.

I wasn't thrilled with the overall headroom, but again, it is not built to major-console rail spec and its commensurate cost. With greater attention to internal levels, and a bump up on the amplifiers, I was a happy camper.

A few nits I had with the ATB, besides the lack of preamp output points, are its lack of solo buttons on the in-line monitor section of each channel, its lack of a stereo aux pair, and the lack of an LED indicator on soloed channels. I would also gladly trade the input reverse for a proper fader reverse (which would allow recording levels to be set with a rotary knob and the monitor signal to flow down the main channel to the faders. Or just go ahead and flip the current design around so the default is that the monitor path uses the channel strip controls and the tracking/input path uses the rotary knob and pan, as it is on most large-format in-lines I have used. In that way, all the EQ, sends, fader levels and pan settings and tweaks made throughout a tracking day add up to a great rough mix; plus, no recording levels were harmed because they were out of the way on knobs and safe from musicians who will invariably move faders while the engineer is out of the CR.

| SUMMARY

The obvious markets for the Toft Audio Design Series ATB console are project studios, smaller music-oriented studios, secondary rooms in larger studios, private/commercial production suites, fixed installations and broadcast audio suites – in short, anyone who will appreciate having (or learning about) the flexibility and core functionality of a traditional analog in-line console, and whose clients will be more impressed than with any of the usual suspects in the eight-bus arena.

The ATB packs a great deal of features and functionality within its handsome wood-trimmed frame and the appeal of its impressively evocative Trident-esque stylings will go far towards the board's acceptance in a wide range of applications both above and below its price-point station.

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.

by Rob Tavaglione

KRK Systems Exposé E8B Powered Monitor

High-end offerings from KRK offer superior accuracy, imaging and power.

BENCH TEST | Page 31

As I grow older and wiser, and my gear collection improves, I now find myself drawn to products that are not so much innovative, but simple. Or I should say simply beautiful, both in their quality components and straightforward design. My opportunity to review the Exposé E8B powered monitors by KRK Systems has confirmed these values for me and reminded me that top quality monitoring is priority one.

| FEATURES

The Exposé E8B is KRK's top of the line, near-to-midfield monitor, boasting impressive specs, an eye catching look and an equally eye catching MSRP of $6500 per pair ($5k street). First released in 1998, some significant changes have been made to the E8 monitor, although none are major or 'downgrades.' The biggest change is the new "AlbeMet" tweeter, made of a combination of aluminum and beryllium (previous versions used Kevlar or titanium), which is rather rigid, very fast and ready for high sample rate reproduction (response out to 30 kHz!) With a resonant frequency well above human hearing, distortion is reduced, and extended, flatter response is achieved. This new tweet uses the same inverted (concave) E8 design, an interesting look further complemented by the E8B's yellow woofer. This yellow woofer is effectively KRK's 'branding' — here it is two layers of Kevlar with a central layer of Rohacell, which contributes both stiffness and dampening.

The E8B cabinet is not to be ignored; it is hefty and overbuilt to the degree of weighing in at over 70 pounds! The old hexagonal design is now more like a pot-bellied, rectangular cabinet, and these curved and smooth lines serve a number of acoustic purposes. First of all, the external radius edges reduce diffraction around the cabinet, improving imaging and helping create a wide sweet spot. Internally, these smooth lines and lack of parallelism reduce standing waves. Unwanted resonances are reduced through cabinet rigidity, excessive mass and a rubberized 'footing' that helps isolate the E8B from its mounting surface.

Marketed for both near and mid-field monitoring, powerful amplification is in order and the E8B features 120 watts for the tweeters and 140 for the woofers. This bi-amplification is courtesy of two discrete, Class A/AB power amps crossed over at 1.9 kHz. Below 8 watts of output they operate at Class A, above 8 watts at Class AB, reportedly increasing detail at lower volumes.

The rear panel is familiar and traditional, with a singular XLR input, high frequency shelving control (from -2 dB to +1 dB, in .5 dB steps), high frequency level control (again from -2 dB to +1 dB), system level control (from -30 dB to +6 dB), a HPF switch (-3 dB at 45 Hz, 50 Hz or 65 Hz) and the ubiquitous IEC power connector.

The E8B pair was accompanied by KRK's V12S subwoofer, a 12-inch self-powered design, which I incorporated into my system but not this review; the sub was not essential to the review, per se, but essential to my successful work habits and listening preferences. Rest assured, the V12S performed admirably with plenty of output, headroom and a pleasant puffiness to its punch.

| IN USE

Upon powering up the E8Bs I heard one of my favorite sounds: the satisfying click of a sturdy relay switching on, as I noticed that attractive backlit glow of the KRK logo. With no time to interrupt my workflow, I had to jump right into overdubbing and editing a heavy metal project that has been testing the limits of my new JBL LSR4328 monitoring system and the limits of my critical listening abilities. With new monitors, my ears immediately gravitate to the mids; here I was delighted to hear flatness, accuracy and a complete lack of personality (of course, this is meant as a major compliment). The high end was ample

KRK continues on page 30 ➤

FAST FACTS

APPLICATIONS
Studio, project studio

KEY FEATURES
New one-inch "AlBeMet" tweeter; eight-inch Kevlar with Rohacell woofer; hefty and overbuilt hexagonal cabinet; Class A/AB bi-amplified power — 120W for tweeters, 140W for woofers; XLR input, high frequency shelving control, high frequency level control, system level control, and HPF switch

PRICE
$3,250 each

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

KRK Continued From Page 28

but completely unobtrusive; no distortion or shrillness was present as I envisioned long hours of work without fatigue.

Overall, I was looking for a little more sheen when I noticed the rear panel high frequency controls are not flat at the 'midnight' position, but actually at -.5 dB. I adjusted these controls to flat and got all that I asked for.

Many of my clients like an occasional blast of high SPL, so I slowly pushed the E8Bs. At lower levels the bottom end detail and overall balance was impressive and these qualities continued as I slowly cranked the control room volume pot. I'm sure I gave up before max output, but the point is nearly academic as I was at 105 dB; it was all I could stand!

The E8B's delivered quantity and punch in the bottom end that was surprising, but not hyped. Apparently, the measures taken by KRK to reduce cabinet resonances have paid off; the amount of definition in the low end was astounding, allowing detailed mix decisions (i.e. kick drum vs. bass guitar vs. floor tom) with confidence. In particular, I don't think I've heard 200 to 400 Hz with such pleasantness and honesty (outside of mastering).

PRODUCTPOINTS

- Sonic excellence in imaging, clarity, dynamics, and frequency response
- Ample headroom and power
- Non-fatiguing and translatable

- Relatively high price
- Too heavy for meter bridges

SCORE

The E8B has it all: precise imaging, frequency response, accuracy, dynamics, and power ... and a relatively high price.

I worked on a huge variety of projects over the next month, including classical piano, bluegrass, rock, and pop, and the E8Bs always seemed comfortable. As I carefully mastered a bluegrass project — one that had employed far too many cheap condensers and an inconsistent upright bass — lesser monitors would have smeared the top end and blurred the bass notes. Not these KRKs — I applied surgical EQ and meticulous multiband compression to tame the sonic inaccuracies. Detailed changes of less than .5 dB were clearly audible, allowing me to fix problems that would have normally vexed me or been overlooked.

| SUMMARY

The bottom line is that the Exposé E8B is the best sounding nearfield monitor I've ever heard. They rank right up there with mastering monitors for accuracy, imaging and trustworthiness. The switching Class A/AB design is brilliant and will be helpful to those who value truth at low listening levels. Those who value sheer butt-kicking volume will not be disappointed either — in the words of my client (whom I proudly thrust into the sweet spot), "Wow, I can feel the pressure and everything is so there and so real ... I've never heard anything like this!"

The E8B's simply have it all: precise imaging, frequency response, accuracy, dynamics, and power. There were only two negatives for me, and they are not performance related: a lack of room diagnostics and a hefty price. However, those who are now used to 'intelligent' monitors and still desire analysis and corrective room EQ can look to out-of-the-monitor solutions for such features. And, as good as the E8Bs are, their MSRP may prohibit many potential users. That's such a shame because — as one finds truthful monitoring, resulting in effortlessly translatable

mixes — informed listening will inevitably become priority number one. In use, the Exposé E8B will have you feeling very informed, indeed.

Rob Tavaglione owns Catalyst Recording in Charlotte NC and is calling for a cease-fire in the volume wars. Join him at www.myspace.com/catalystrecording.

| REVIEW SETUP

Soundcraft Ghost console; Digital Performer 5.12 DAW; Apple Mac Pro quad-core computer.

| KRK VXT8 VS. KRK E8B

Sure, it's an absolutely unfair fight — a $799 list powered monitor next to a $3,250 list powered monitor — but my recent month-long audition of the Exposé E8B next to my own VXT8 pair was, to me, a worthwhile lesson in product line quality and placement. In KRK land, there is no optional price point between these two models: respectively, their top-of-the-line and 'mid-level' powered nearfields, both with eight-inch woofers. Just as there's a whole lot of performance in between these two monitors, there's also a huge leap in price. Yes, the E8B justifies this jump, but the budget-conscious pro audio customer in me must ask, "Does it have to be such a leap?" (Sob. Sniffle.) But I digress.

The VXT8, for me, has become an indispensable rock and roll monitor; it's great for all kinds of tracking, critical listening, and most mixing purposes. But over time — and most clearly illustrated in direct comparison with the E8B — the VXT8 has shown some limitations in imaging. The E8B's razor-edge sharp imaging allows the listener to literally reach out anywhere in the space around and between their two ears so as to precisely tap every random transient square in its tiny, hard little belly. Another way to put it is that my eyesight isn't bad enough to require full-time corrective lenses, but switching between the E8B and VXT8 is like peering through, then over, a pair of rarely worn glasses; the difference is transcendent. You get the drift. Most importantly, everything that comes out of an E8B pair sounds absolutely exquisite and organically huge. Here, it's the difference in performance that is empirically worth the money.

Thus, I would recommend the E8B to the professional tracking/mixing engineer who is shopping as if they are buying their last-ever powered nearfield monitor. After all, if you folks don't buy an Exposé-level monitoring system now, you'll just want one later.

-Strother Bullins

BENCH MEASUREMENT DATA

Frequency Response
On-axis 90 Hz to 20 kHz +/- 6.7 dB

Bass Limit
71 dB SPL @ 62 Hz @ 2 meters (<10% Distortion)

Control Action
HF Shelf Half dB Steps +1.0 to -2.0 dB Steps > 10 kHz

Actual Response
+1.1 dB to -2.0 dB > 8 kHz

LF Level
Half dB Steps +1.0 to -2.0 dB > 1.9 kHz

Actual Response
+0.8 dB to -2.6 dB >1.9 kHz

LF Adjust
-3 dB @ 45 Hz; Actual Response 90 Hz
-3 dB @ 65 Hz: Actual Response 103 Hz
-3 dB @ 50 Hz: Actual Response 117 Hz

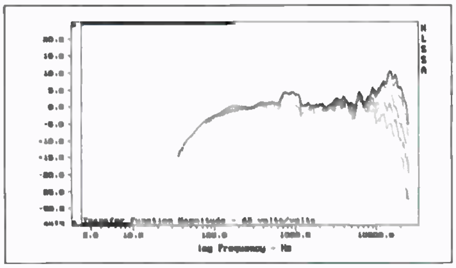
BENCH MEASUREMENT COMMENTARY

The Bass Limit of the speaker is the Sound Pressure generated at 2 meters in a 7600 cubic foot room with less than 10% distortion. The 10% 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% 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 gives equivalent anechoic results above 200 Hz including front panel reflections and cabinet diffractions. Overall results give true acoustical summation of all drivers and passive radiating elements.

The E8B has less low frequency extension and dynamic capability than the large and relatively heavy cabinet and specifications would lead one to believe. The Low Frequency Adjust function has a curious effect when the switch is set to -3 dB @ 50 Hz; unlike the 65 and 45 Hz settings, response begins falling below 300 Hz and is less than half power by 117 Hz. The high frequency controls are reasonably close to marked action, which serves to make them easy to use, but the limited range probably makes them less useful overall than one might think.

Directivity is well controlled but the frequen-



cy response basic shape has some obvious errors; the largest of these are a large +10 dB peak centered on 14.8 kHz and a 4 dB bump between 700 Hz and 1.1 kHz.

— Tom Nousaine

NEW PRODUCTS

MARANTZ PMD580 Solid State Recorder

D&M Professional, manufacturer of Marantz Professional, has introduced the rackmount PMD580 Solid State Recorder, the newest addition to its highly successful family of rackmount digital recorders. Most importantly, the new PMD580 features network connectivity; via its Ethernet port, the PMD580 can be positioned as a network device, allowing users to set menu parameters, schedule recording events, and transfer and archive audio files, all by using a web-based GUI interface from any PC or Mac in the network. The new unit offers balanced XLR inputs and outputs, S/PDIF as well as AES/EBU digital interfaces and RS232 control. The new PMD580 offers MP3 and WAV Recording formats, and users can select between 16-bit and 24-bit resolution.

PRICE: $1,399

CONTACT: D&M Professional | ☎ 630-741-0330 ➲ www.d-mpro.com

MINNETONKA SurCode For Dolby E

Minnetonka Audio Software, Inc. has added to its SurCode product family with the introduction of SurCode for Dolby E. The SurCode for Dolby E product line will offer Dolby E software decoding and encoding for various platforms. SurCode for Dolby E Decoder is the first product available for both Pro Tools and Minnetonka AudioTools AWE. Dolby E, as the standard for surround audio using stereo delivery, has grown into a standard exchange format and is therefore also required for non-broadcast facilities. According to the manufacturer, SurCode for Dolby E Decoder is the perfect way to integrate Dolby E into existing workflows. One or more instances of the SurCode for Dolby E Decoder or Encoder on an iLok allow studios to use their Dolby E licenses on different audio workstations in different studios and in various plug-in or standalone formats.

PRICE: $3,195

CONTACT: Minnetonka Audio | ☎ 952-449-0187 ➲ www.minnetonkaaudio.com

HOLOPHONE N-CODE portable multi-channel encoder

Ideal for larger remote productions requiring surround recordings, the Holophone N-CODE takes six channels of audio from the H2-PRO or H3-D and converts them to two channels using Dolby's Pro Logic II technology, allowing full 5.1 channel surround sound audio to be captured or transmitted to virtually any stereo recoding device, or broadcast over the existing stereo infrastructure. The N-CODE features six XLR quarter-inch inputs with mic/line selection and comes equipped with two XLR outputs that connect the encoder to the recording device or transmission line to a remote broadcast truck. Further, it provides 48v of phantom power through six high-quality microphone pre-amps and is battery powered for portable, real-world applications.

PRICE: TBA

CONTACT: Holophone | ☎ 416-362-7790 ➲ www.holophone.com

AUDIO-TECHNICA AT8004 AND AT8004L ENG Microphones

A-T has introduced its AT8004 and AT8004L Omnidirectional Dynamic Microphones, two new microphone solutions designed specifically for the broadcasting market. The AT8004 (5.93 inches) and the longer AT8004L (9.43 inches) both offer exceptionally natural sound reproduction; both are ideal for handheld interviews, ENG/EFP and sports broadcasting applications, or as the "mono" mic when used in conjunction with a stereo mic. Additionally, the AT8004L's longer handle easily accommodates a microphone flag while still providing sufficient space for the talent to grip the microphone. The AT8004 and AT8004L offer a frequency response of 80-16,000 Hz and an omnidirectional polar pattern. In addition, they each feature a rugged housing and a hardened-steel grille as well as internal shock mounting

PRICES: $135 (AT8004) and ?$149 (AT8004L)

CONTACT: Audio-Technica | ☎ 330-686-2600 ➲ www.audio-technica.com

NEWS

With the exception of the once-a-year telecasts of the Super Bowl and the Oscars, FOX's American Idol is the most highly watched program on television.

This year, all American Idol contestants are using the **Sennheiser** MD 5235/SKM 5200 combo. Shown is David Cook, one of leading contenders for the coveted title.

WBRZ-TV has invested in Ross' newest switcher platform with the purchase of a **Vision** Quattro 3.5 Multi-Definition Video Production Switcher, along with Ross openGear terminal equipment, to equip their new HD build out. Vision Quattro provides 4 keys per ME and up to 4 MEs, available in both a small and large chassis.

Symetrix, Inc. has appointed Pearson and Pearson Marketing, Inc. (PPM) to represent its products in the Rocky Mountain Territory that encompasses Utah, New Mexico, Colorado, Wyoming, southern Idaho, and eastern Montana.

For its extensive NFL and NASCAR coverage, FOX Sports conducts on-site audio/video production from a multi-truck unit provided by Game Creek Video. FOX Sports Audio Consultant/Senior Mixer, Fred Aldous, chose **Tannoy** for all of the unit's monitoring environments. Mixing is provided via **Calrec** Alpha Bluefin and **Sigma** audio consoles.

by **Bruce Bartlett**

ZOOM H2 Portable Digital Recorder

This recorder is simple, well equipped, sounds good, and — best of all — priced right.

Priced at $199 street with built-in stereo mics, the Zoom H2 Handy Recorder is the lowest cost pro flash-memory recorder available. But its sound quality is more than adequate for its intended uses: recording interviews, conferences, ENG, gigs, band practices, podcasts, music lessons, and sound effects. It even works well recording orchestras and acoustic instruments.

Compared to the Zoom H4 (which I reviewed in the November 2006 issue of PAR), the H2 has a mini phone-jack mic input rather than XLRs. You could use a mic preamp or small mixer as a front end to the H2 if necessary. Smaller than the H4, the H2 is simpler to operate and more intuitive to navigate the menu items.

The H2 records WAV or MP3 files onto an SD card. Select up to 24-bit/96 kHz WAV files with 128 times oversampling, or MP3 files up to 320 kbps (or VBR). You can record a maximum 2GB file size on an SD card up to 4 GB. The unit runs for four hours on AA alkaline or rechargeable batteries, so it's ready to grab and go.

FAST FACTS

APPLICATIONS
Studio; project studio; music, spoken word, and environmental location recording; rehearsal and songwriting documentation

KEY FEATURES
Stereo WAV or MP3 recording onto a SD card up to 24-bit/96 kHz resolution; comes with earphones, 512 MB SD card, USB cable, tripod stand, mic-stand adapter, carrying strap, foam windscreen, mini-phone to RCA cable, and an AC adapter

PRICE
$334.99 list

CONTACT
Zoom/Samson Technologies |
☎ 631-784-2200
www.samsontech.com

FEATURES

Zoom supplied all the accessories you might need: earphones, 512 MB SD card, USB cable, tripod stand, mic-stand adapter, carrying strap, foam windscreen, mini-phone to RCA cable, and an AC adapter. The unit itself is the size and shape of a deodorant stick and has a high-impact plastic chassis.

On the front panel are a high-contrast LCD screen, front/rear/surround mic pattern buttons, Menu button, Record/Enter button, Play/pause button, and forward/backward buttons. The left side of the unit contains a headphone/line-out mini phone jack, record/playback level buttons, power switch, and power connector. On the right side are a USB port, line-in mini stereo phone jack, L/M/H mic-gain switch, and an external mic input jack (mini stereo phone). A battery access panel is on the back, while a covered slot for the SD card is on the bottom.

As an aid to setting the mic-gain switch, a Mic Active indicator flashes if the input signal is clipping, and stays lit otherwise.

An unusual feature is the built-in mic array. The unit has two stereo pairs of XY mics aiming forward and backward. While holding the unit upright, aim the front panel at the sound source for a 90-degree pickup; aim the back panel at the source for a 120-degree pickup. You can record two sources, or a band, with the front and back mic pairs simultaneously. Or record in surround 4-channel mode. You use the mic pattern keys to select Front/90 degrees, Back/120 degrees, front plus back, or surround. LEDs indicate the mode of operation.

Four-channel recordings are created as two stereo WAV files. For playback, all the channels are mixed to stereo. After a 4-channel recording is done, you can adjust the front/back/left/right level balance (3-D panning) within the H2. The 4-channel signals can feed a surround sound encoder for playback on a 5.1 system.

The unit also accepts a stereo unbalanced line-level source or a stereo unbalanced mic that uses plug-in power (DC bias, not phantom).

Using the supplied USB cable, you can transfer recorded files to a computer for editing or burning to a CD. Edited files can be copied back to the H2 as well.

Zoom always has lots of features in their products. The H2 is no different. It includes:

- Metronome
- Guitar tuner with various modes
- Audio interface for a computer; record the input signal directly to a computer, and play it back via the H2 (16-bit/44.1 or 48 kHz format only)
- Delete, rename or split files. Display file information. Put files in folders
- Check remaining recording time on the SD card
- Set date and time
- Low cut filter
- Record mode (set wave and MP3 resolutions)
- Automatic gain control/compression/limiter with several modes
- Auto record function (voice-operated recording)
- Pre-recording mode, in which audio is stored before you press Record so that no music is lost
- Various playback modes (repeat, play all, play one, A-B repeat of a section)
- Normalize audio files
- Convert a wave file to MP3; convert a 4-channel file to stereo file
- Monitoring can be always on, or off until you set recording levels (to increase battery life)
- Check for dropout points

ZOOM continues on page 36 ➤

ZOOM Continued From Page 34

PRODUCT POINTS

- Easy to use
- Good value
- Many extra useful features, such as surround miking
- Very good sound, wide stereo

- None noted

SCORE

The Zoom H2 is one of the best values on the market.

- Adjust display contrast, turn backlight on/off
- Key hold (lock controls)
- Update system software

All these features are accessible via the Menu and don't get in the way of recording operations. The H2 has a limited 1-year warranty.

IN USE

The unit is well designed for ease of use. Batteries and SD card are easy to access, as are the connectors and controls, which are clearly labeled. Navigating the menu features is intuitive. To access the features, simply press the Menu button, then press the arrow keys or the Record button to select and set various parameters. The buttons feel solid.

I first used the H2 to record an acoustic guitar at 18 inches. Here are the steps:

- Set the mic pattern to Front/90 degrees
- Set the mic gain to high
- Press the Record button
- Strum a few chords and check that the Mic-Active LED does not flash
- Set the recording level. The LCD screen meters are easy to read
- Press the Record button again and record a tune

That's all there is to it.

Q: How did the built-in mics sound? *A:* Smooth, wide-range, and uncolored. I also recorded a standup bass with the Zoom H2 and with a flat-response omni condenser mic. The two recordings sounded the same except that the omni had a little more deep bass and about 4 dB less hiss. When I recorded a voice, the H2 and the omni mic sounded essentially the same.

ZOOM continues on page 38 ➤

SECOND OPINION: ZOOM H2

I liked what I saw of the H2 mockup at the January 2007 NAMM show and have eagerly awaited its arrival. While I wouldn't expect a recorder like this to be one I'd take to a professional gig, I've been looking for something I could carry in a pocket or banjo case to record a good jam session or grab a tune that I wanted to learn. The H2 fits this niche nicely. It's small, the case is smooth with no pointy things or fragile projections, the built-in mics are well protected, and there don't seem to be any obvious mechanically weak parts waiting to break. I really can carry the H2 in my pants pocket if I'm careful not to sit down on it.

The display is clear and the menus are, for the most part, self-explanatory. It took a while to get used to using the left-right buttons to scroll up and down through the menu, but otherwise menu operation is just fine.

Typical of this genre, the record volume up/down buttons adjust the digital level after the A/D converter, so it's important to understand the H2's gain structure. Using the "Rec Level Down" button to bring the meters on scale will almost surely result in a clipped recording. The scale on the (digital) record level runs from 0 to 127, with 100 apparently being the unity gain setting. I'd have preferred this to be calibrated in dB, and, of course, anything below 100 (0 dB) is the danger zone. The bright "Mic Active" LED flashes to indicate analog clipping, but it's only active when using the internal mics.

There's a three-position analog attenuator switch (0, -10, and −24 dB) to get the analog level into the ballpark. It works on the built-in mics and external mic input, but not on the Line Input. When recording from a mixer, I needed a 20 dB pad ahead of the H2's Line Input in order to avoid clipping.

The mics sound fairly good and provide a reasonably well-defined and stable stereo image. I found the front-and-back 2-channel "surround" mode to be particularly useful for interviews. Sitting across the table from my subject with the recorder in the middle and turned about 30 degrees off axis yielded a stereo recording with the two of us separated left and right.

My gripe list is fairly short. Some of these things can be fixed with a firmware update if Sam Zoom thinks they're sufficiently important, while others are inherent in the hardware architecture and we'll have to live with them.

The 2000 mA-H NiMH batteries that I used for testing provided just over three hours of recording time on a charge. This is a near perfect match for 44/16 recording on a 2 GB memory card (both power and file space run out at about the same time), but I didn't get significantly more recording time from a charge using the 192 kbps MP3 mode, a mode that I'd be more likely to use for informal recordings. My battery gripe is that there's no provision for charging the installed batteries when operating on AC power or from the USB port. Since there's a menu-operated switch to select between alkaline and NiMH batteries (this changes the characteristics of the battery life indicator), if the hardware was there, the charging function could be enabled when the NiMH mode was selected.

The H2 records time stamped Broadcast Wave files (BWF), but the time stamp of each file is 00:00:00 rather than the BWF standard time based on the number of samples after midnight. There's a clock/calendar which time-stamps the files, so the recorder knows what time it is and could use this information for the BWF time stamp as well. In a related issue, when viewing File Information, the "Time" displayed is the length of the recording. It would be useful if the file date and time were also displayed. Since files are named automatically (they can be renamed, up to 7 characters, with patience), when browsing the file list, knowing when the recording was made might help to identify the material.

Although the clock/calendar keeps alive for several hours without batteries, the "preferences" (sample rate, pre-record mode, compressor/limiter mode, low-cut filter, etc.) are stored in a file on the flash memory card. When you install a new or freshly formatted card, the recorder reverts to its default settings, some of which you may need to change before recording again. This can increase your "reel change" time.

My one totally out-of-scope wish is that the H2 could write data to an external USB storage device (like a hard-disk drive) as an alternative to the flash memory card. I've dreamed of this for a while now (I do a lot of all-day recording gigs), and my crystal ball tells me that we may start seeing this feature in the next generation of portable recorders.

There's a lot to like with the H2. It makes perfectly good casual recordings with its internal mics, it sounds pretty good when fed from a good quality line-level source as long as you don't let it clip, and its size, shape, and weight make it no hassle to carry along if I think I might want to record something. At the price, it's going to be hard to resist.

— *Mike Rivers*

ZOOM Continued From Page 36

Left and right signals are audibly reversed from the audience's perspective unless you mount the unit upside down.

I recorded a loud rock band with the mic gain set to low. Stereo separation was wide. There was no audible distortion on playback. The H2 preamp clips at about 120 dB SPL but is clean at 116 dB SPL.

Even though the mics are cardioid, handling noise was not excessive. Neither was wind noise with the included windscreen in place. The H2's stereo stage is much wider than that of the Edirol R-09 recorder, which uses two omni mics about 2.5 inches apart.

To test the line input and A/D converter in the H2, I recorded an acoustic guitar with a top-quality mic into a good mic preamp, which fed both the H2 line input and a PreSonus Firepod audio interface line input. In an A-B test, the two recordings sounded remarkably similar. The Firepod might have been slightly more open or transparent in the high frequencies, but the difference was subtle.

I wanted to transfer my recorded files to a computer for editing. I plugged in the USB cable with the H2's power off, pressed the REC key, and dragged and dropped the files to my computer. Simple. The transfer rate was about 0.83 MB/sec.

The operation manual is very clear and explains everything a user would need to know about the H2's operation. It also includes an index and sections on troubleshooting and error messages.

SUMMARY

I was very impressed with the H2's sound quality, features, and ease of use. I feel that it is one of the best values on the market. If you're looking for a low-cost portable stereo/surround flash recorder, the Zoom H2 is definitely a winner.

Bruce Bartlett is the author of "Practical Recording Techniques Fourth Edition" and "Recording Music On Location" published by Focal Press.

WHAT IF IT IS TOO LOUD?

I haven't seen this mentioned in many reviews, so I'm guessing that I must be attending the loudest concerts in town. My problem is that I'm finding many performances which are too loud for your average portable recorder to handle. A few online forum users confirm my observation. Rap and metal fans also know what I'm talking about. Earplugs can protect my hearing, but sometimes I wish my recorders had their own earplugs. Sony's two popular portables (PCM-D1 and PCM-D50) smartly feature a true -20dB pad and they both handle loud volumes well. Zoom and other manufacturers should follow Sony's lead. It's easy to confuse pad dB with preamp gain when looking at specs. Too many portable recorder and USB microphone manufactures wrongly believe that -10db is a sufficient pad strength.

Analog mic gain on the Zoom H2 is determined by a 3-position input switch. Buttons (+, -) conveniently placed on the face of the unit digitally fine-tune the recording level from there. Moderately-loud band practices with the H2 and it's built-in mics were not a problem. Rock club environments, with the H2 placed back at the soundboard area, also recorded without clipping. Even the punishingly loud snare hits of a legendary 1980's drummer, on a reunion tour, failed to spike maximum level. I'll call him "Stu" and he came very close to clipping the input.

Being stuck in the back of the room sometimes yeilds a too-ambient recording. I prefer to be up where the action is in order to capture better stereo. This usually involves bringing along external, unpowered mics. The Zoom H2 recorder has an 1/8" stereo jack for external microphones, so as a work-around I have occasionally used Nady RSM4 ribbon mics to capture the loudest concerts. The naturally reduced output of the ribbons is easier on the analog front end of the recorder. Unexpectedly, their unique ribbon coloration even suited some of the more abrasive concerts. Low-energy dynamic mics will work almost as well. Do remember to turn off the H2's trickle-power when using ribbons. Phantom power is bad for ribbon mics. A dual-XLR to stereo mini-jack adapter is also required.

— *Davis White*

Producing Surround Sound at 50th Grammy Awards

Embracing an Inevitable Digital Broadcasting Future

by **Mel Lambert**

With just 10 short months until the analog turnoff for terrestrial broadcasting in February 2009, the major networks are anxious to promote the advantages of all-digital 1080/720 transmissions with 5.1-channel surround sound. To this end, the Recording Academy's annual Grammy Awards ceremony is a unique opportunity to showcase surround broadcast at its very best. And this year's 50th Grammy Awards, broadcast in mid-February from the Staples Center, Los Angeles, chose to spotlight the legacy of the past and the challenges of the future. Co-produced for The Academy by Cossette Productions in association with AEG Ehrlich Ventures LLC, the show included a number of innovations, including live concert performances by the Foo Fighters and winners of My Grammy Moment contest from the nearby Nokia Plaza, plus a remote performance in London by multi-Grammy winner Amy Winehouse.

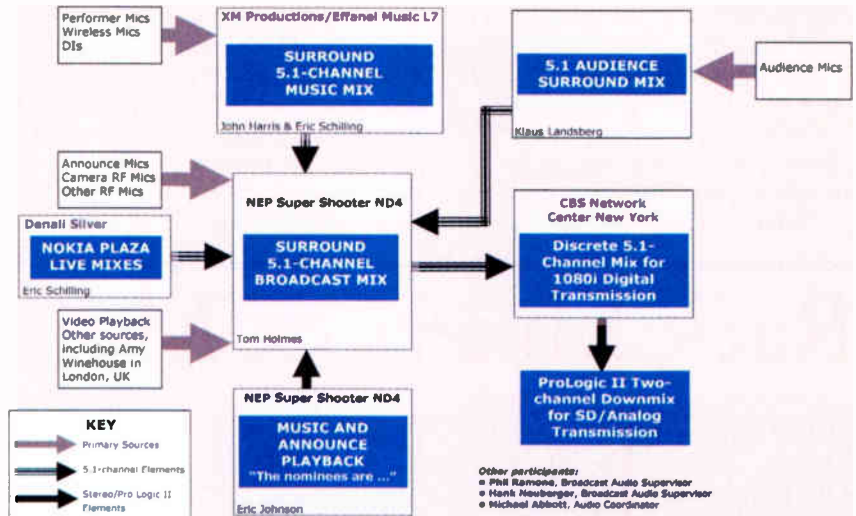
While the February show marked the sixth year that the telecast had been broadcast in HDTV/5.1 surround sound, this year the organizers decided to make life a shade simpler. CBS asked the show's producers to prepare a single, no-compromise 5.1-channel mix, and accommodated analog and SD viewers with a two-channel downmix prepared at the network's broadcast center in New York. Again, Audio Coordinator Michael Abbott was responsible for infrastructure that supported a trio of broadcast mix rooms required for the CBS broadcast, plus pre-records at nearby Capitol Studios, the remixing of legacy recordings for the 5.1 performances of The Beatles Across the Universe soundtrack and Love segments with Cirque du Soleil]. "We also had the opening segment remix of original multitracks from the first Grammy Awards broadcast featuring Frank Sinatra, a remote music mix from the Foo Fighter and a last-minute Amy Winehouse UK remote." Abbott worked closely with Hank Neuberger of Springboard Productions and Phil Ramone, Chairman Emeritus of the Recording Academy's Producers & Engineers Wing, both of whom supervised the broadcast audio.

XM Productions/Effanel Music provided its L7 Mobile Recording Studio for music mixing of live acts within Staples Center, with John Harris and Eric Schilling taking turns with alternate acts. Joel Singer served as Effanel's Engineer-in-Charge. A total of 192 Aphex 1788A Remote Controlled Mic pre-amp channels were provided for L7 from the stage area, routed via a fiber-optic "A" and "B" system. "That configuration allowed us to have the next setup recalled and ready to go when we switched to it," Singer points out. The Denali Silver broadcast mobile was on hand in the nearby Nokia Plaza to mix the Foo Fighters' concerts. Eighth Day Sound Systems handled sound mixing for the band, using a Digico D5 Digital Live Console for front-of-house; the main PA comprised a D&B line array powered by Lab:gruppen amps.

"Since the Foo Fighters' stage [in the Nokia Plaza] was 300 yards away from L7," explains Neuberger, "it was a challenge mixing in a separate video truck. Eric Schilling went over to the truck by golf cart during the middle of the live telecast! Also, the Amy Winehouse remote from London came together late in the day - the first time we heard it was during Dress Rehearsal, on show-day morning."

As on previous occasions, a second mixing environment was provided for the music mixers to refine various mixes memorized during rehearsals on the L7's Digidesign ICON Assignable Console and Pro Tools HD|6 Core. The Offline Remix Booth (ORB) housed an identical ICON control surface and Genelec DSP 8200 Series monitors. Pro Tools Session files and QuickTime reference

SIGNAL FLOW FOR 50th ANNUAL GRAMMY AWARDS



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The discrete 5.1-channel mix was Dolby-E encoded for satellite transmission to CBS Network Center, New York.

video from rehearsals were reloaded and static 5.1 surround-mix levels, EQ and dynamics settings developed for the musical acts, often with the artist's mix engineers in attendance. Updated ICON settings were loaded onto portable drives and returned to L7 for the broadcast. TC Electronic supplied

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several reverb effects plug-ins, in addition to the new TubeTech CL1-B Compressor.

Readily conceding that the ORB Gelco trailer, as a remix area, left something be desired, Singer looked for a cost-effective solution. While the operation's Genelec DSP 8200 Series loudspeakers feature built-in digital equalization, "that only goes part way to overcoming acoustic anomalies," Singer concedes. "The untreated ORB environment meant that the [DSP] was working pretty close to its limits. To help it along, we asked Media Specialty Resources [MSR] to provide a series of [StudioPanel] absorbers, diffusers and bass traps for the walls and ceilings." The treatment made a big difference, Singer says, "and meant that we had available more accurate monitoring within the second mix area."

Grammy veteran Klaus Landsberg helmed a separate Audience Reaction Mixing room, which received feeds from 34 mikes within Staples Center that were mixed on a Yamaha DM2000 digital mixer and a bank of 32 channels of Focusrite OctoPre pre-amplifiers. Monitoring was via JBL Professional LSR-4328P active systems. Audience mikes comprised Audio Technica BP4029 stereo shotguns, AT4051a cardioid condensers, AKG C547 hypercardioid boundary models on the stage front and four Sennheiser MKH-416 hypercardioid shotgun models positioned on the stage apron covering the left and right sides of the audience. Landsberg produced a 4.0-channel mix: front left-right and rear left-right. The final mix passed through a pair of CEDAR DNS1000 Dialog Noise Suppression unit "for room shaping," Landsberg says, and then to a TC Electronic DB-8 Broadcast Processor used as a 5.1-channel multiband compressor "to make a cohesive whole" of the multiple mike sources.

Stage microphones included wired and wireless models from AKG, Audio-Technica, DPA, Earthworks, Sennheiser, Shure and Neumann. Soundtronics' Dave Bellamy again served as the show's frequency coordinator, working with Bill Kappelman, as more than 40 wireless mikes were re-used during the Grammy Awards. Sennheiser models included a custom decorated SKM5200 with an MD5235 capsule for Rihanna, and a SKM5200 mic with MD5235 capsule for Beyoncé. Sennheiser SKM5200 hand-helds were used by presenters Ludacris, Joe Mantegna and Bonnie Raitt, and an SKM5200 mike with MD5235 capsule for The Black Eyed Peas' Fergie, who joined John Legend to perform "Finally." The Foo Fighters' Dave Grohl used his usual Sennheiser MD431II dynamic, backed by drummer Taylor Hawkins on an Evolution Series e945 mike, while performing "The Pretender" backed by the Grammy Philharmonic Orchestra conducted by Led Zeppelin bassist John Paul Jones. Andrea Bocelli honored Luciano Pavarotti with Josh Groban during "The Prayer" on twin SKM5200s with MD5235 capsules. In London, Amy Winehouse used an Evolution 935 condenser mike during her two-song set.

More than 250 Audio-Technica microphones were available, including Artist Elite 5000 Series wireless systems. Artists using AEW-T5400 handheld wireless mikes included Kanye West, Morris Day and The Time with Rihanna; Kid Rock with Keely Smith; Aretha Franklin and BeBe Winans;

John Fogerty with Little Richard and Jerry Lee Lewis; and The Beatles Sgt. Pepper finale. Backline A-T mikes included AE5100 for ride cymbal, hi-hat, overheads and strings; AE2500 for kick drum; AE5400 and ATM250 for Leslie cabinet; AT4050 for horns, overheads and guitar cabinets; plus ATM350 for strings.

Carrie Underwood used a custom Shure SM58/UHF-R wireless mike on "Before He Cheats," while Alicia Keys opted for a Shure KSM9/UHF-R wireless during her opening collaboration with Frank Sinatra on "Learnin' the Blues."

THE CRITICAL 5.1-CHANNEL HD BROADCAST MIX

Multichannel submix stems from the audience reaction area and Effanel truck were received by NEP Super Shooter's ND4 Video Production Truck, where broadcast mixer Tom Holmes - using a 86-fader/dual-layer Calrec Alpha with Bluefin Digital Console - balanced these with playback sources and announcer microphones to generate a 5.1-channel broadcast mix. The Calrec console also received an LCR mix from Amy Winehouse's live appearance in London, together with a 5.1-channel mix from the Foo Fighters stage.

"My biggest challenge was to maintain consistency between the 5.1 HD mix and the stereo mix for analog and SD viewers," Holmes considers. "In the past, two mixers worked in [a pair of] control rooms to handle that task; this year there was only a 5.1 mix, which was folded down to a stereo [ProLogic II matrix-encoded surround] mix at CBS [New York]. To check compatibility, we monitored several consumer setups." These four mixes, according to Holmes, were:

a. 5.1 discreet for the people that have an HD signal, and a home theater surround audio system;

b. A flat fold down of that same 5.1 mix, for people that have an HD signal, but only use stereo speakers (maybe on the TV set);

c. Lt-Rt, the encoded 5.1, for people that receive an SD signal, and only use stereo speakers to listen to; and

d. ProLogic II, the decoded Lt-Rt, for people with an SD signal, but who are listening in surround on a home-theater system.

"We also needed to get the most audience response from an audience of industry attendees," which very often is far more subdued than a fan-based audience. "The show needed to sound as live as possible without washing out the music. But the rehearsals - with stand-ins, no audience and an empty arena - never really sounded very much like the real show. Although we are there for six or seven days, there is a lot of content. So you end up rehearsing a band or a production item for an hour and a half, but only do [a run through] two or three times. I always wish we had just one more rehearsal!"

All Grammy show numbers were performed in real-time, Neuberger stresses. "Moving over to a separate remote truck for the Foo Fighters was slightly awkward," he offers. "But since it was used for a single dedicated number, it was clearly the best solution." The Amy Winehouse live insert from the UK was mixed at The Riverside Studios, West London, by Tim Summerhayes. "For transmission capacity and reliability reasons it was decided to only transport three channels of audio from London," continues Neuberger. "Tim sent us

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BROADCAST | REVIEW

by Rob Tavaglione

JBL LSR4312SP Linear Spatial Reference Powered Subwoofer

Room control, low frequency performance make LSR subwoofer ideal for studio use.

The LSR 4300 series of monitors from JBL — including the 4326 and 4328 with 6- and 8-inch woofers, respectively — have been widely praised for their sound quality and modernistic networking features. They first gained my attention in 2006, when I test-drove a 4328 pair (sans sub) and found the monitors quite pleasing in both timbre and construction. I thought the RMC (Room Mode Correction, JBL's system of reducing low-frequency room resonances) was interesting, but ultimately useless to me as I had a 15-inch powered sub from another manufacturer that could not be controlled by the software. Philosophically and practically, I insisted on full-range monitoring, so I knew that I had to wait until JBL had an LSR sub that could truly complete the system. This brings us to today.

FAST FACTS

APPLICATIONS
Studio, project studio, audio post, and broadcast recording and mixing environments

KEY FEATURES
I/O includes six analog inputs (L, C, R, LS, RS and LFE) on XLRs (and their accompanying XLR pass-thru outputs), AES/EBU I/O, S/PDIF I/O, two quarter-inch inputs (L/R only); -10 dB gain selector at LFE in; two HiQnet connections with RJ45 connectors; six dipswitches (for switching polarity and crossover point), an IEC power connector, and five LEDs indicating input selection; measures 19.75 inches high x 16 inches wide x 19.5 inches deep

PRICE
$1,139 list

CONTACT
JBL Professional | ☎ 818-894-8850
www.jblpro.com/LSR

FEATURES

The LSR 4312SP is the logical extension of the previous LSR models in every way. Matching in material and design, this self-powered, 450-watt RMS, forward-firing, bottom-ported subwoofer (12-inch woofer, self-shielded, neodymium motor structure, response -6 dB at 27 Hz, -10 dB at 24 Hz, 116 dB continuous SPL) incorporates into Harman's HiQnet protocol, allowing adjustment of the sub's volume, bass management and RMC selection from either your computer or the supplied remote control. All of these controls are also found on the front panel of the 4312, which is simplistically attractive with its grill-free design and "soft touch" rubber-paint surface. The 4312 measures 19.75 inches high x 16 inches wide x 19.5 inches deep, and has an MSRP of $1,139.

The rear panel is where the 4312's many I/O's are contained, including six analog inputs (L, C, R, LS, RS, and LFE) on XLRs (and their accompanying XLR pass-thru outputs), two-channel AES/EBU I/O, and, finally, S/PDIF in and out. There are two balanced quarter-inch inputs on the Left and Right inputs only, as well as a -10 dB gain selector at the LFE input. Additionally, there are two HiQnet connections (using RJ45 connectors), a series of six dip-switches (for switching polarity and settings for digital inputs), an IEC power connector, and five LEDs indicating input selection.

For further detail on the construction, design, and theory behind the LSR monitors, please see Russ Long's excellent and thorough review for *PAR* (found online and in both October 2006 print and digital editions). With such an understanding of the LSR system and the RMC calibration, I was set to disassemble my current 2.1 system, a pair of Event 20/20 nearfields and accompanying 20/20/15, 15-inch, 250-watt RMS woofer. This system was comparable to the JBLs in many ways — internal biamplification, similar amounts of RMS power to the drivers, same driver sizes (except for the larger sub) and even tweeter construction (1-inch soft-dome silk) which had been a primary factor in selecting the 20/20s (a non-fatiguing top end is quite important to me).

IN USE

Upon first powering up, the LSR4328s indicated that RMC calibration had not yet been performed, but I proceeded to do some listening anyway. Without the 4312 sub, the 4328s were full and admirable in their bass, but not fully extended in their response. Selecting Bass Management on my remote control engaged the sub, kicking in more of that below-60 Hz information I'm so finicky about; it completed the sonic picture for me. Engineers with smaller control rooms than my 17 x 25-foot space may find the 4312 sub unnecessary, especially if they are not frequently working on music with ample low-frequency information. This review is not aimed however at such applications, but indeed maximization of the control room experience, as close to a flat 20 Hz to 20 kHz as I can get.

[From the manufacturer: "Although the LSR4328P and LSR4326P are flat to 50Hz and 55Hz, respectively, we recommend a subwoofer for any size room. Smaller rooms that do not have real bass traps may be more sus-

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ceptible to LF problems. In addition to extending LF response below 30Hz, crossing over LF to the sub allows flexibility and optimum placement of the LF in the room, improving LF performance in a small room. The LSR4312SP RMC system overcomes associated boom typical in small rooms."]

That being said, the LSR system sounded

PRODUCTPOINTS

- Deep, full, extended bass with adequate amplifier headroom
- Definite improvements in mix consistency and bass definition
- Remote control and LSR software are extremely helpful

- Driver is exposed and could be damaged easily

SCORE

Accurate monitoring in acoustically challenging environments is a hurdle now clearable thanks to a system anchored by a LSR4312SP.

well powered and loud enough for "rock band" monitoring, amply deep and full, but it still created that bottom end inconsistency I'm so unfortunately used to (two notes in the scale seemed disproportionately loud and two spots close to the mix position had quite untrustworthy bass response). At this point, I'll confess to needing more/better bass trapping in my control room (and I'll be getting some more soon), but the question is can the 4300s with RMC be an improvement for now?

I set up the supplied condenser mic with its head right smack in the mix position, connected it to the front left speaker (which supplied 15v phantom power) and started the RMC calibration. A series of blips and sweeps sounded at various levels and within 15 seconds RMC was done. Playback confirmed that something had been changed, as the bottom end was absent of those humps, less cloudy, a little tighter ... not as bassy overall, but clearly a more even and trustworthy bottom end. The RMC works through the attenuation of 73 low-end frequencies (at 1/24 octave centers), with Q values from 1 (1.4 octaves) to 16 (1/11th of an octave), a maximum attenuation of -12 dB and the RMC overrides the rear-panel polarity dipswitch setting (when deemed necessary). I rapidly switched back and forth from non-RMC to RMC with my remote control while identifying the trouble frequencies that were notched out.

The LSR Control Software was a little confusing at first, but proved to be a valuable part of the LSR system. After connecting the left front speaker to my Mac Pro with a standard USB cable (provided), the LSR Control software allows the muting or soloing of any individual speaker, as well as other system controls. Most importantly to me, it allows you to see what changes the RMC made to your system. My curiosity was piqued, as I'm sure many of yours would be too, and here's the numbers I found for my system:

Frequency, Q, & Gain
Left: 121; 5.01; -4.5 dB
Right: 140; 2.31; -5.5 dB
Sub: 59; 4.40; -3.5 dB

[From the manufacturer: "The correction at the left frequency listed above looks like the system compensated for proximity to a wall, or work surface — a 'boundary condition.'"]

Notice the additional gain reduction at a higher frequency in the right channel; this is to be expected as I have a large equipment rack on the right side of the mix position, which is creating a standing wave at 140 Hz as bass gathers around it. With my 4312 sub crossed over at 80 Hz (the most natural-sounding setting to me), one should note that I had frequency buildup both above and below my crossover point, requiring both the mains and the sub to be adjusted. Keep in mind that if one is using bass management, as I am, that it must be selected prior to the RMC calibration for accurate results.

There's more to audio life than bottom end, and I must say the LSR4300 system took some getting used to at first. Even though the important midrange was quite flat, the overall sound was slightly forward, almost aggressive. The highs were crisp and uncluttered, but more piercing and revealing than I was used to. I had a number of critical projects coming up that would surely test these monitors translatability, the sub's accuracy and my flexibility.

First up, were mixes of a five-song EP of dynamic indie rock. Although the band had a straightforward drum/bass/two-guitar attack, I was concerned with my ability to get melodic bass lines consistently audible without excessive flab and getting their dynamic female lead vocal to sit in the mix — always dominant, but seriously challenged by loud guitars. Here, 4328s and their 4312 sub simply blew me away. I got more translatability and "predictableness" than I ever got with my old system. Snare sat exactly where I knew it would (for once), guitars were perfectly even "L to R" in their almost hard-panned glory and vocals fit right into that narrow little niche I was aiming for. My "kick drum: bass guitar" balance ratio was nice and even, but overall bass levels were a touch inconsistent from mix to mix, although much better than normal for me.

Next up was mixing a full length of neo-metal, which incorporates enough double kicks and jagged-edge bass guitar to require a super-accurate bottom-end balance. I was concerned with keeping kick drum tight, yet large and getting bass guitar audible, despite layers of Mesa Boogie-powered guitar crunch. Again, the LSR system really impressed me. Bass guitar was not ever a

problem, staying audible in the mixes without muddiness, a testament to the truthful low-mid response of the 4328s. Kicks sat right where I hoped they would without any frequency response humps like I'm often used to getting. Still, the overall bass content of my mixes was a little less than consistent, so I tried EQing the system a bit.

Using the Control Software (although I could have used the remote) I fiddled with the low and high shelving EQ, experimenting with different corner frequencies and value. Only two dB of boost or cut is offered, but a slight bottom boost and top cut got me where things sounded a little more pleasant to my ear ... just in time, as I had to master a compilation CD for the local entertainment weekly. Here, the 4328s and the 4312 sub really rose to the occasion, allowing me to hear detail and imbalance in the mixes that I quickly corrected. The music wildly veered from country, to rap, from indie to pop, but "the moving bullseye" of consistent bass was easy to hit. On a 12-song CD, I only retouched the EQ (after some real-world testing in my car, computer, headphones, etc.) on two songs — way better than my typical results. I found my QC procedures to be much more "double-checking" than revealing.

JBL was kind enough to provide a second 4312 subwoofer for this review, and the performance was predictably positive. With such a 2.2 system, bass management must be selected to engage the subs, but their information is not summed, the bottom stays in true stereo, like the top end. The second subwoofer allowed the system to have more headroom and achieve higher bass output levels, but performance was not improved much at lower monitoring levels. The use of a second sub would be highly recommended for audio post applications, especially if there were a true LFE output.

Be forewarned: with a second sub and RMC, one can spend a lot of time experimenting with placement and analyzing the results the myriad of combinations can be both useful and overwhelming.

SUMMARY

Now that I've tasted the sweet fruit of total monitoring system control, I'm never going back. I have always found it hard to get levels exactly matched from left to right speaker and even harder to calibrate my sub properly (what with subjective crossover frequency, level, placement and polarity adjustments). The ease of the RMC setup and the consistency of the results has been an eye-opener for me. A perfectly tuned system, with speaker muting, dimming and soloing, remote control and bass management gives you very significant tools that will help you work better and faster.

Those of you with 4326s or 4328s should seriously consider adding the 4312 sub and completing the system. Then, for those of you who are considering affordable, widely available professional bass trapping products for your room, rest assured that the final hurdle, more accurate monitoring, is now clearable thanks to a system anchored by a LSR4312SP.

Rob Tavaglione has owned and operated Catalyst Recording in Charlotte, NC since 1995. He welcomes your questions and comments. Contact him at rob@catalystrecording.com.

REVIEW SETUP

Soundcraft Ghost mixing console, Digital Performer 5.12 workstation, Mac Pro computer, Lucid Gen X 6-96 wordclock by **Heather Johnson**

Audio Production For Gaming

PlayStation Portable's God of War: Chains of Olympus

With a storyline based on Greek mythology combined with an ex-Spartan warrior lead character named Kratos and more than enough enemies and evil creatures to keep a warrior busy for a few centuries, Sony Computer Entertainment America's (SCEA) single-player action-adventure game God of War, exclusively available for PlayStation2, was considered one of the top video games of 2005, receiving accolades from gamers and reviewers alike for its story-driven puzzles, cinematics, graphics, and audio. The game lent itself to a sequel: God of War II, issued in March 2007.

Now, a year later, SCEA brings an all-new story in God of War saga to the PSP (PlayStation Portable): God of War: Chains of Olympus, which features the same characters, creatures, and villains as the previous titles, along with more bloodshed than a slasher flick and the same high-quality standards as its multiple award-winning predecessors.

Developed by Ready at Dawn Studios in Santa Ana, Calif., in collaboration with SCEA's Santa Monica studio, God of War: Chains of Olympus features several different vicious beasts and enemies from the dark and brutal world of Greek Mythology, which leads to a game with tens of thousands of lines of recorded dialog, combat and creature sounds, as well as multiple layers of sound effects and more than five hours of original score.

Much of the God of War: Chains of Olympus musical score incorporates cues from God of War and God of War II. Roughly 15 minutes of new music for the game was composed by composer Gerard Marino (one of four composers used on God of War and God of War II) using Cubase 4 and integrated into the larger body of work. Much of the previously recorded material featured live orchestra and other instrumental parts recorded in London, Prague, and San Francisco—a common practice in most SCEA titles. "There is no substitute for recording live musicians," says SCEA music director Chuck Doud. "The intensity and drama of both the musical compositions and the gameplay could only be realized and brought to life by an ensemble of extremely talented players, conductors, arrangers, orchestrators, and recording and mix engineers."

For the dialogue, the SCEA crew convened at Soundelux in Hollywood, Calif., where dialog manager Greg deBeer spearheaded a team that includes dialog recorder Justin Langley, and voiceover director Keythe Farley. Over a span of eight months, they auditioned and recorded a solid cast of voices, including lead characters Terrence "TC" Carson (Kratos), Corey Burton (Zeus), Lloyd Sherr (Cronos), and Linda Hunt (narrator), as well as artists as soldiers, delivering background yells, and chiming in for Walla sessions.

DeBeer pays special attention to the tone of each voice, no matter how minor. "The God of War cast was primarily warriors, gods and demigods," said deBeer in an earlier interview, "so all of those voices needed to have a lot of weight behind them. The main character was a mortal, but the mortal had to sound strong enough to beat the Gods, so he needed to have those qualities in his voice to make the story convincing." Langley used the Soundelux U95 microphone to capture the myriad lines of dialog and recorded directly into Pro Tools using minimal processing on the front end.

Keeping detailed records of mics used, mic placement, and the settings used on mic pre's and in Pro Tools becomes as crucial as the recording itself. If an actor has to record additional parts months later, they need to sound exactly as they did on that first session. Excellent organizational skills become a priority as audio files get passed on to the editors and for localization (adapting the language for a different country or region). "The best tool we have is a solid and well-thought-out naming convention," said deBeer in an earlier interview. "Every line in our script gets a file name and is put into a folder structure, usually organized by character name, and also put into spreadsheet format."

Sound designer David Farmer, who also worked on God of War II, created many of the creature sounds for the PSP title. When traveling to local zoos and other remote locations, Farmer often uses Fostex FR2s, Sound Devices, and Tascam field recorders, along with a wide range of field mics such Sennheiser 416s and Sanken CSSs. "We don't typically use very many library sounds," says Gene Semel, senior manager for SCEA's

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sound group. "We record everything live. If our sound designers need something, they can go out and get it either in the field or create it on our Foley stage. We then add it to our library, so our library is huge. Basically we've created a God of War library that is constantly updated, especially now that our field recording is in high-def."

MIXING AND MANAGEMENT

All of the God of War: Chains of Olympus audio—including score—eventually made its way to SCEA's Foster City, Calif. headquarters for mixing and mastering. The complex's

Chuck Doud

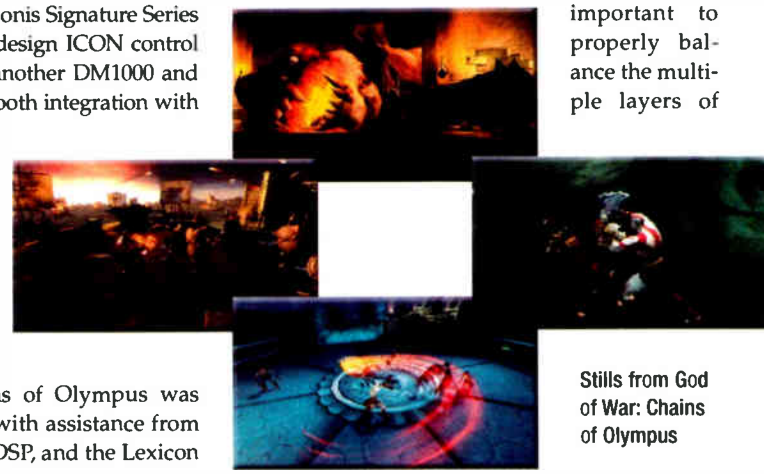
newly revamped studio, designed by Chris Pelonis, includes 11 "Pods" (edit bays) equipped with Pro Tools | HD, Digi's D Control surface and Yamaha DM1000 digital production consoles. The 5.1-equipped Pods integrate with the facility's THX-certified master control room, which features 7.1-surround monitoring courtesy of Pelonis Signature Series monitors and a Digidesign ICON control surface, along with another DM1000 and HD rig to ensure smooth integration with the Pods.

A full gamut of plug-ins handled audio processing, including Waves bundles, Pitch Shifter, Time Blender, and various VST plug-ins. God of War: Chains of Olympus was mixed in Pro Tools with assistance from Sonnox Oxford, McDSP, and the Lexicon 960, among other plugs.

To manage the enormous volume of files shared throughout the SCEA facility, the company utilizes Studio Network Solutions' SANmp SAN sharing software. "We've been using the SANmp with much success," says Semel. "It allows for one constant state of backup, which is very important to ensure that the data is always retrievable if a crash happens. And we can pull hundreds of tracks off the SANmp with no problem."

Of course, storage within the game is also important. The PSP titles hold about 2MB of audio RAM, used mainly for dialog, while Sony's proprietary AC3 audio files, are streamed direct to Blu-ray disc. A 24-bit/96 kHz sample rate gives the PSP titles a dynamic range comparable with the early PlayStation2 titles, but the medium still comes with challenges. "The PSP in particular presented its limitations on this project, so we had to work a little harder to make it work," says music engineer Joel Yarger. "But overall, there isn't much difference sonically between what we accomplished on the PSP versus what we were dong previously in PlayStation 2."

Aside from shaping the sounds in the mix, it's also important to properly balance the multiple layers of



Stills from God of War: Chains of Olympus

sounds. "Part of the art is figuring out what we want the player to hear," says Semel. "Because if the player hears everything that's going on at the same level, it's going to be a cacophony." Audio programmers handle the bulk of this task, writing lines and lines of code to integrate the music with the graphics. "Half of having a successfully scored video game is how that music is integrated," says Doud. "When it's done right, I don't think there's anything like it, because video games are so immersive. The goal of the music is to enhance the gameplay. The score needs to adapt to that, and deliver that emotional connection."

Heather Johnson is a San Francisco-based journalist and author whose books include "If These Halls Could Talk: A Historical Tour Through San Francisco Recording Studios."

by **Mel Lambert**

Euphonix S5 Fusion Digital Post Production Console

Derived from the best-selling System 5 series, Euphonix S5 Fusion Digital Post Production Console is a complete mixing package that combines the firm's new processing DSP SuperCore engine with EuCon Hybrid; the latter allows the console surface to control its own DSP channels in addition to assignable channel strips implemented within external DAWs. In other words, S5 Fusion combines the control-surface features of the well-respected System 5 with the usefulness of enhanced EuCon- and HUI/Mackie Control-based workstation connectivity. The basic S5 Fusion surface incorporates 24 multi-format faders, eight assignable knobs per channel, TFT multiformat metering, Master Module and an integral DAW display screen and twin motorized joysticks. The assignable, touch-sensitive knobs can be set to control EQ, filters, compressor, expander/gate, aux and pan settings, as well as TDM, VST and Audio Units DAW plug-ins. EuCon protocol currently is supported by Apple's Logic Pro, Merging's Pyramix and Steinberg's Nuendo; control of applications such as Pro Tools, Digital Performer and Final Cut Pro is via HUI, Mackie Control and similar protocols. High-resolution screens display multi-format metering, track info and routing display.

Folded into this package are Euphonix Automation, PatchNet Routing software, serial and MIDI Machine Control (MMC). For larger installations, the base 24-fader configuration is expandable within a larger frame size that can now accommodate an additional eight extra faders, a Submaster Assignable Module, plus a PEC/DIRECT Panel and Producer's Desk. (Extra Euphonix or third-party I/O plus DSP also can be added to provide more channels and buses.) MRP of the 24-fader base configuration is $150k, including Monitor Matrix.

DAW-SAVVY CONTROL FOR TV, POST, AND FILM

S5 Fusion is targeted at TV facilities, post houses and film stages that need an ultra-compact, DAW-savvy control surface and console for mix-to-picture surround-sound production. Given that such facilities need to turn the room more efficiently to handle a wide range of productions, a medium-cost, multi-function system like the S5 makes a great deal of financial and operational sense.

The console's DSP busses and channels are accessed via a quartet of MADI-format I/Os - 224x168 channels at 48 kHz -including eight analog and 16 AES/EBU-format outputs for monitoring. Other digital formats, including Core Audio or ASIO including Pro Tools, also can be accommodated. A fully loaded S5 Fusion will handle over 150 DSP channels. Monitoring is provided via a new digital surround-monitor matrix built into the DSP SuperCore, complete with analog and digital speaker outputs. A new Studio Computer handles eMix file management and PatchNet software, plus Euphonix mix automation software and EuCon Hybrid DAW control application for up to five workstations.

ALL-IN-ONE PACKAGED AUDIO MIXING SYSTEM AND DAW CONTROLLER

As will be readily appreciated, S5 Fusion is most definitely a wolf in sheep's clothing – it combines the integrated power of a full-function EuCon-compliant mixing console with the ability to mix DAW tracks from the control surface, complete with plug-ins. And you soon become so used to having DAW controls under your fingers at the touch of button – my evaluation system was simultaneously accessing Nuendo 4, Logic Pro and Pyramix, as well a multiple hardware channels implemented by the DSP Core – that it makes any other way of working pretty mundane. Mixing with a mouse is, at best, slow and tedious - mapping DAW functions to knobs and buttons makes mixing and signal adjustments seem so much more intuitive with virtually instant response times.

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EASE OF USE

The S5 Fusion control surface is extremely easy to learn and use, with its truly intuitive interface and screen display. It provides plenty of visual feedback, with convenient high-resolution LED meters beside each channel fader. A new charcoal finish for the channel-strip panels improves legibility and display contrast. Euphonix' unique illuminated and touch-sensitive knobs instantly reset when settings are changed or recalled from memory. The center of each knob includes a switch that can be used for certain select functions, and/or to punch-in automation. A Spill function provides easy access to multi-format sources, by unwrapping them onto adjacent faders for trim and update. A total of 240 SnapShots of static console settings can be stored and recalled, together with 48 Layouts for different surface configurations. The external PatchNet router/patch bay accommodates from 256-by-256 for the base system to 1,024x1,024 signal paths for a fully loaded DSP SuperCore. Of course, comprehensive surround panning and monitoring is provided as standard, plus up to 7.1-channel metering via TFT high-res displays at the top of each channel that also map EQ, dynamics, pan and routing. Machine control also is available for Soundmaster, Colin Broad, Tamura and JSK systems.

INSIDE THE MODULES

The S5 Fusion control surface is supplied standard with three different types of modules: CM401-T Master Control Module with talkback; CM408-T eight-fader Channel Strip Module (three in the basic 24-fader setup); and CM411-J Joystick Module. Optional modules include the CM403- F/J Film Monitoring Panel with twin Joystick Module (to replace the CM411-J module) and a CM409-F Blank Module. While the standard frame supports a total of five modules, a larger frame accommodates seven modules. The surface connects to the equipment rack via standard Cat5 Ethernet cables, enabling remote location. MADI ports handle the primary I/O highways.

The banks of channel strips are laid out much like a traditional analog topology. Each source offers two A/B inputs, phase reverse and gain trim. If mic pre-amps have been specified, the strip now features additional remote controls for remote gain, high/low input impedance, high-pass filter, phantom power on/off, etc. The four-band 20 Hz to 20 kHz parametric EQ section offers ±24 dB of gain (switchable shelving for upper and lower bands) and local variable EQ control. It is extremely flexible and sounds like Euphonix EQ; what's more to say? Filters can be set to high pass, low pass, band pass or notch mode. Usefully, the later includes Q control and a "boost/listen" function that enables the audio to be auditioned without the filter in-circuit while identifying the problem frequency range – maybe a 60 or 120 Hz hum, or an unexpected noise problem.

The dynamics section includes a combination compressor, expander and gate with hysteresis, key input and side-chain filter. An assignable section handles sends to 16 aux buses and group assign to 32 mix busses with multiformat panning, plus two direct outputs selectable pre/post-fader. Sources to each channel strip can be mono signals up to 7.1-channel stems; the processing elements can be inserted in virtually any order between input and output, including 20 seconds of delay. Usefully, a choice of APL, PFL and SIP solo modes are available.

> The S5 Fusion is a full integration of multichannel mixing and DAW playback for post, film, and broadcast.

Up to 32 mix buses can be configured as multichannel stems - Mono, Stereo, LCRS, 5.1, 6.1 and 7.1, plus custom configurations - for different production formats, such as a stereo mix section with two individual busses and a 5.1-channel stem. The targeted signal can be routed to all busses in that stem with/without pan inserted. Usefully, it is also possible to route to individual busses – a.k.a. Direct Assign – for each stem. (Maybe the center bus of a 5.1-channel stem.) Up to 16 mix stems may be configured and named. Pan mode available for multichannel stems comprise Front Pan, Surround Pan, Rear Pan, Boom Level, Non-Boom Level, Divergence, Focus and Rotate. Stereo or surround sources – maybe a 5.1-channel pre-mix or audio sub-group – can be controlled from a single control strip referred to as Multi-Format Master, which dramatically simplifies routing. The 16 Aux Sends can be configured in mono or stereo pairs, with Aux Masters controllable from the center section.

Single or dual meters above each channel strip can be set to monitor levels for Mix, Group or Aux bus. Metering is accurate to within 0.25 dB from 0 dBfs to -48 dBfs with clip indication. A handy green bar located to the left of each meter shows the approach to peak level. When set to be Multi-Format Master controlling stereo or surround stems, the metering automatically changes to reflect that mode. Up to 24 meter presets can be stored and recalled.

Using the supplied eMix application, the Fusion's seven processing sections - delay, metering source, insert point, EQ, filters, dynamics, fader and mute - can be arranged in any order, even as audio is flowing through the signal paths, the only limitation being that at least two adjacent channels must be re-ordered at a time. (Maybe EQ to follow dynamics, insert after the EQ or dynamics, and metering after the fader on channels one and two, or channels 1 thru 8?)

Swap buttons enable single or multiple strips to switch between two layers, allowing 24 strips to control 48 sources. In addition, most critical sources can be moved to the central sweet spot – maybe dialog elements to the left and music sources to the right - with different layouts being saved for later re-use, complete with all appropriate knob and fader settings.

The CM401T Master Module includes not only a fully assignable channel strip but all of the necessary master facilities, including monitoring, communications, bus masters solo, transport and automation controls. Selectable solo modes include Solo-In-Place (SIP), After Pan Listen (APL), Pre Fader Listen (PFL), Fader Backstop PFL and Solo Safe – useful for isolating effects returns from Solo-In-Place. A bank of eight knobs and function switches control the master faders between Group, Mix and Aux Bus Masters. A Master Fader handles controls of Mix Section output levels.

In addition to the various console monitor sources up to 7.1-channel, as many as 24 external devices can be accessed from the center section. The monitor matrix automatically selects the appropriate format to match the source. Each of the monitor outputs - CR, Mon A, B, C, D – features individual level and source select buttons with eight-character labeling plus Dim and Mute. The primary Control Room can be routed to the Main Monitors, Alt 1, or Alt 2; this output also can be folded down – maybe 5.1 to LCRS, stereo or mono. (How these multichannel outputs combine, and the appropriate level adjustments, are usefully controlled via a bank of 24 user presets.) Other functions include talkback, listenback, oscillator and slate.

In addition to SnapShots – up to 240 per title - all important S5 Fusion parameters can be automated dynamically to timecode, including faders settings, EQ, pans, aux send levels, dynamics and DSP in/out. Automation is writable in sections or as continuous moves throughout the length of a Title. Read/write modes include Isolate, Read, Write Absolute,

mode that provides a soft-ramped change between the level at punch out and previous levels. Overall automation modes comprise Preview (a preset value can be set prior to punching into write), Suspend Preview, Capture (stores in memory the values of objects that are writing, allowing these objects to be punched back into record at the stored values - handy!), Join, Fill and All Match. The pair of motorized joysticks can be assigned to any channel.

For film and video mix-to-picture project, a Conform utility re-aligns the automation data to match the edited images using imported Edit Decision Lists that contain start time, end time and duration for each edit, together with the mode (insert, delete, move, etc.). The S5's Conform then creates a new mix that integrates with the EDL changes. The optional CM403-F/J Film Monitoring Panel includes a film-monitor panel with conventional PEC/Direct switching plus space for custom controls. An application-specific display shows monitor status and multi-format stem meters.

THE OPERATIVE WORD: FUSION

As I discovered, EuCon Hybrid extends S5 Fusion's control to encompass familiar DAW systems, whose controls now map to the familiar on-surface controls. Everything is fluid and intuitive. You soon forget what channels are via the DSP Core and those from the DAW Engine - everything is seamless and fully integrated. Plug-ins and workstation functions respond quickly and accurately – it is a delight to move from control to control, without MIDI-induced command lags. Mixing and sample-accurate editing – you still have full access to the DAW from the surface via the keyboard and music, of course – is now a single operation, without the need to turn to a companion surface and screen. Everything is where you would expect it to be, ready for adjustment by hitting the minimum number of keys.

Multiple tracks from separate DAWs can be arrayed across the surface adjacent to Euphonix DSP channels – maybe dialog tracks from Pro Tools, music from Logic Pro and effects from Nuendo – plus Foley and ADR from DSP SuperCore sources. If a cue isn't correct, it's just a matter of a few seconds unfolding to access the elements, correct a sync point, for example, or even substitute an alternate take, and be back into the mix again. No interruptions, no fuss – S5 Fusion is definitely the way to go.

BOTTOM LINE

The S5 Fusion is a full integration of multichannel mixing and DAW playback for post, film and broadcast. If I seem to be impressed by the latest offering from the Series 5 innovation series, you would be correct. S5 Fusion offers what it advertises – powerful DSP SuperCore Mixing functions plus full integration of digital audio workstations via both HUI and Mackie Control, plus the faster and more elegant EuCon protocol. As one would expect from such a serious and committed developer of medium-format consoles, Euphonix has a winner on its hands.

My sincere thanks to Ozzie Sutherland, Euphonix Console Product Manager, and Andrew Wild, VP of Marketing, for providing access to an S5 Fusion at the company's LA offices within a specially-equipped mobile demo truck.

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Pro Audio Review

HOT Gear

2008 NAB SHOW

At the April NAB convention in Las Vegas, our editors and contributors found several impressive new, or recently introduced products that we predict will do well in the broadcast production/post niches. To recognize those products we have created the inaugural ***Pro Audio Review*** *Hot Gear* — NAB 2008 list. The *Hot Gear* list will follow every major U.S. pro audio trade show except for AES, which ***PAR*** will continue to bestow its ***PAR*** Excellence Awards. We added some commentary why we picked em and web URLs, so readers can peruse, at their leisure, the company PR in more detail.

Audio-Technica 8004L ENG Microphone

Price:$149

"Nice ENG dynamic omni for less than $150.00. For those who like the big stick mic." *www.audiotechnica.com*

Audio Technology Inc. (ATI) DM500 Digital Monitor

Price: $999 (for complete kit)

"ATI debuts its new digital monitor which checks status bits, sample rate, word length, and gives you a listen via the headphone jack. You know it's good when the techy engineers come up to the stand and say they gotta have one." *www.atiaudio.com*

Audio Precision ApX585 HDMI Interface Test/Measurement System

Price: (depends on configuration)

"The big boys of audio testing are the first ones on the block to provide an HDMI interface and testing for this up and coming A/V connection scheme." *www.audioprecision.com*

Auralex Eco-Friendly StudioFoam Acoustic Treatment

Price: (depends on quantity)

"What could be better: making effective sound shaping material with soy-based raw materials that don't cause pollution during manufacture." *www.auralex.com*

Aviom 6416 Microphone Preamp Module

Price: $7,500

"Aviom adds another critical component to its audio distribution system, a high quality, remote controllable mic preamp module." *www.aviom.com*

Bias Peak LE 6.0 Editing Software for Apple

Price: $129.00

"This powerful, way easy-to-use, streamlined version of Peak Pro is ideal for those Mac-based broadcast edit suites." *www.bias-inc.com*

D-K Technologies MSD100C Stereo Loudness Meter

Price: $2,690

"The standard in post, broadcast and music production facilities, the metermen at D-K Technologies realizes the value of an accurate stereo "loudness" meter for the variety of program levels that broadcasters now have to deal with." *www.dk-technologies.com*

Euphonix S5 Fusion Digital Mixing System

Price: Starting at $149,000

"Euphonix adapts the best selling System 5 into a lower cost, powerful system for DAWs that is at home at well-equipped broadcast HD facilities as well as post." *www.euphonix.com*

Genelec 5040A Powered Mini-Monitor,

Genelec 6010A Compact Powered Subwoofer System

Price: (TBD)

"Pint-Sized powered Genelec pedigree main monitors with a min-sub that digs in at under 50 Hz." *www.genelec.com*

Henry Engineering Six Broadcast Mixer

Price: $1,199

"Leave it to longtime audio designer Hank Landsburg to come up with a compact, high-quality USA-made mixer that offers just the right number of channels and I/O. Check out those knobs." *www.henryeng.com*

Holophone PortaMic 5.1 Camera Mount Surround Microphone

Price: $599

"Holophone's obsession with capturing audio in surround has netted it several innovative products in recent years, including this compact 5.1 system, that mounts right on the camera." *www.holophone.com*

IZotope ANR-B Noise Reduction for Broadcast

Price: $4,995

"It ain't cheap, but this sophisticated, powerfully adaptive noise reducing tool was designed for the various kinds of noise that broadcasters want to eliminate or reduce, yet leaves the primary audio relatively unchanged." www.izotope.com

Lectrosonics D4 Digital Wireless Transmitter/Receiver System

Price: (TBD)

"Lectrosonics new D4T and D4R digital wireless systems had significant wow factor at NAB 2008 with the debut of its 24-bit systems utilizing a proprietary coding scheme in the ISM band." www.lectrosonics.com

Marantz PM620 FLASH Memory Recorder

Price: $399

"With a glut of FLASH digital recorders on the market, Marantz gives us an easy-to-use, robust handheld that records at 48 kHz." www.marantz.com

Minnetonka SurCode Dolby E Decoder

Price: $3,195

"With Dolby Es increasing use in broadcast and other audio environments, the innovative software designers at Minnetonka Audio have introduced a Dolby E decoder for Pro Tools and Audio Tools AWE. Expect an encoder software package by AES 2008." www.minnetonkaaudio.com

Samson G-Track USB/Interface Microphone

Price: $139

"So it's ridiculously inexpensive. The G-Track is still a budget podcaster/broadcaster's dream, saving some major coin and getting a USB mic with some control and extra connectivity." www.samsontech.com

Sennheiser MKH-800 Twin-Capsule Microphone

Price: $3,356

"Not a new concept, but Sennheiser gives us the MKH-800 sonics in a twin-capsule design with separate outputs. The mix engineer can can adjust and fine-tune the polar pattern from the desk, offering a wider palate of sound possibilities." www.sennheiser.com

Sound Devices 788T Eight-Channel Multi-Format Recorder

Price: $5,995.00

"Known for its robust, and excellent sounding location recorders, Sound Devices adds an 8-channel, 48 khz sampling recorder to its multi-channel family. Hard drive, DVD-RAM or FLASH are storage options." www.soundevices.com

Sony DWT-B01/Sony DWRS01D Wireless Transmitter/Receiver System

Price: $2,500 (DWT-B01), $3,150 (DWR-S01D), $900 (Adapter)

"Sony means business with its flagship digital wireless systems that puts out 24-bit audio capability from the camera to the receiver." bssc.sel.sony.com

SSL Matrix SuperAnalogue Console

Price: $25,000 (16-channel version)

"SSL is pushing the Matrix to broadcasters as well as music studios. The Matrix SuperAnalogue signal path, combined with its digital control, offers the classic analogue console sound and the advantages of digital that broadcast facilities expect." www.ssl.com

TASCAM DR-1 Portable FLASH Recorder

Price: $399

"TASCAM' new low-cost recorder with built-in mics. It is easy-to-use and offers very good sonics for just a little money." www.tascam.com

Yamaha PocketTrack 2G Miniature FLASH Stereo Recorder

Price: $399

"Outstanding sound in an extremely small package with built-in speakers. Just don't lose it." www.yamaha.com

ZAXCOM Deva Fusion Recorder/Mixer System

Price: $10,000

"Complete 16-channel, recorder and mixer with 192 kHz sampling recording and legendary ZAXCOM build. There is a reason this company is a top choice for cinema recording." www.zaxcom.com

by Mel Lambert

Soundcraft Vi6 Digital Live Production Console

The English console firm Soundcraft enjoys an enviable reputation as a long-established manufacturer of analog designs. So, when the time came to make the inevitable move to large-format all-digital topologies, it was perhaps inevitable that the Harman-owned division would look within the brand family for DSP ideas and in-development technologies. Given Studer's long-term experience with Vista and earlier D950 Series digital broadcast and production consoles, a synergy of effort was available to Soundcraft hardware designers and software writers. (And recall that, until late last year, with the introduction of the Vista 5 SR console intended for touring systems and fixed installations, Studer hadn't put a toe into the live-sound marketplace.)

In essence, the Soundcraft Vi6 Large-Format Digital Live Console features a unique integration of touch screens and encoders that eliminate complex mental mapping. The 32-fader control surface handles mixing of 64 mono inputs into 35 outputs, with 24 insert send/return pairs assignable to any of the I/O channels via the powerful Vistonics II touch-screen based user interface. For added flexibility, in addition to internal routing to 32 Group/Aux/Matrix busses, all input channels can be set up with direct outputs and routing to the main LCR and Left-Right/Stereo busses. Exemplary sound quality is guaranteed through a combination of Soundcraft's ultra-low noise analog mic pre-amp designs and wide ranging experience in control-surface design, plus advanced 40-bit floating point DSP magic from Studer. (In essence, the Vi6 console was designed using the proven reliability of Studer's Vista and OnAir 3000 Series consoles.)

THE VISTONICS II REVELATION

Vistonics II is a revelation; it enables users familiar with analog designs to be up to speed quickly and easily. A combination of touch-screen color TFT monitors with integral, on-screen rotary controls and switches provide a particularly intuitive and easy-to-remember Point-and-Click GUI that can be learned in minutes. The user is presented with all the system information he/she needs to make critical decisions, but without unnecessary clutter – what ergonomic designers might refer to as "perfectly optimized control density." And since it controls the functions being accessed, Vistonics II's unique integration of touch screens and encoders eliminates the fatigue of mentally mapping each control to its corresponding function when channels are moved freely around the control surface, or re-assigned to other layers.

The color-coded, context-sensitive graphics around the assignable and relabeled rotary controls make it clear which type of function is being adjusted, while a high-contrast, white highlight provides a reminder of which specific channel is being controlled. To move to another area of the channel strip, or to close down the control area, simply requires a touch on the TFT screen. Color-coding also streamlines system setup and mixing. For example, the Blue input stage and routing screen provides adjustment of input delay, mic gain, digital trim, HP/LP filter frequency, channel patching, channel naming and stereo pairing.

MAIN ELEMENTS

The Vi6 Digital Live Console system consists of three main elements: the control surface; a local rack containing the SCore Live processing engine; and a stage box that connects to the local rack via reel-mounted CAT5/7 cable for dramatically simplified system hookup. Amphenol RJF connectors are standard for enhanced reliability. The mixing control surface can be located up to 300 feet from the stage box; in fixed installations, the use of CAT7 cable increases that distance by 100 feet. (The optional fiber-optic interface enables cable runs up to 1.25 miles/2 km between stage box and the local rack/DSP engine.)

The stage box houses 64 mic/line analog inputs (with 48V phantom power) and 32 analog line outputs. As is to be expected, mic-amp gain is controllable remotely from the Vi6 control surface. The local rack offers 16 analog line inputs, three analog mic/line inputs, a talkback mic mounted on the surface, plus eight pairs of AES/EBU-format inputs. Outputs include 16 analog line-level, eight pairs of AES/EBU-format digital, three LCR local monitor A lines, two LR local monitor B lines and talkback. For added flexibility, the 64-channel MADI I/O with optical SC connectors is standard but can be replaced with a variety of optional I/O cards, including CobraNet, Ethersound or Aviom A-Net 16. A total of 16 GPIO contact-closure inputs and outputs are available at the local rack, plus eight inputs and outputs on the stage box. Finally, the control surface features a single MIDI input and two MIDI outputs.

Soundcraft considers the Vi6 to be a third-generation design that abandons the layering and assignability of earlier offerings in favor of the more intuitive Vistonics II configuration. The dedicated Vistonics II panel controls eight input channels, with a TFT touch screen that contains 16 processing rotary encoders and companion switches. Sections of the screen around each control are relabeled to indicate what each control and switch currently is han-

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SOUNDCRAFT Continued From Page 58

dling, including a four-band parametric EQ section, compressor/limiter/gate/de-esser section, auxiliary controls and full I/O routing.

The four-band EQ is displayed graphically with boost/cut, frequency and Q/bandwidth settings, plus a main screen that displays the overall EQ curve. For easy assimilation, the frequency settings mimic a radio-tuner scale; HF and LF bands also can be switched to shelving response. The EQ sections, as one would expect, sound very nice, with plenty of cut/boost for even the most troublesome sound sources. Adjustment is very precise, with plenty of headroom in following sections even when working with extreme settings. Touching a target function on the Vistonics II channel strip initiates a corresponding control panel within the lower section, that area being highlighted to identify the active controls. Soundcraft refers to this capability as "Where you look is where you control," which pretty well sums up the design philosophy.

The Dynamics Section comprised a full-function Compressor with threshold, ratio and release controls and an independent Limiter, plus a switchable De-Esser or Noise Gate with attack, hold and release controls, with a frequency-conscious key input. Gain reduction metering is provided within the fader area.

Routing and control of the 32 main busses – designated Aux, Group or Matrix outputs - is achieved via two areas on the channel strip, arranged in two banks of 16; additional stereo controls come into operation for stereo sends. Busses also can be switched on/off with level trim, selectable pre/post fader (for direct sends), while a global mode enables pre/post EQ switching. The remainder of the channel strip controls Pan (assignable LR and C, or LCR mode), Channel Insert (switched pre/post-EQ/dynamics) and direct output. Very usefully, the latter send is assignable pre-filter, pre-EQ/dynamics, post EQ/dynamics and post-fader. Useful for today's remotely controlled PA systems, the built-in HiQnet protocol integration enables the Vi6 to communicate directly with HiQnet-savvy amplifiers and signal processors.

Also included in the Vi6 system software is a suite of very useful plug-ins that comprises Lexicon reverbs and multi-FX plus BSS graphic equalizers. The DSP Package provides eight Mono or Stereo Effects units that may be patched to Aux Buss outputs and then back into a channel input, or inserted into input or output channels. For full control integration, all effects are controlled from Vistonics II graphical displays. The Lexicon DSP offers a total of 14 reverb types, including various plates, halls/spring types and delays, plus chorus, flanger, tremolo and pitch shift. The BSS DSP provides a 30-band graphic EQ on every output, controlled via the first 30 channel faders. Simultaneously, a composite EQ curve is generated and displayed on the Vistonics screen for that output path.

BASK IN THE GLOW

FaderGlow is a new feature from Soundcraft that dramatically reduces the potential confusion of assignability by integrating the fader-track colors used with the Vistonics II GUI to show, at a glance, what channel function a target fader currently is controlling. Also, within the output section, a bank of eight faders are assignable to Aux, Group and Matrix outputs, and very useful VCA control groups - FaderGlow applies, respectively, orange, green, cyan and blue illuminations to each fader to instantly designate the corresponding output species. (No need to read small text labels in poor lighting at the FOH or monitors position.) And, also useful, in conventional input mode each of the Vi6's 32 motorized faders remains unlit. But when the user hits solo for an Aux, Group or Matrix buss master – assuming Follow Solo function is activated – each channel fader becomes a contributor to that soloed buss and is lit in orange, green or blue, accordingly.

THE LATEST SOFTWARE

Version 2.0 software, released in late-Fall last year, added an eagerly awaited Copy/Paste facility to the Vi6's OS, in addition to a very handy Library function. The new Copy/Paste capability enables the settings of any channel, bus, FX section or processing section to be copied and then pasted onto single or multiple channels; an UNDO reverses the last paste operation. Blocks or individual parameters within a channel can be selected for copying using Vistonics II's touch screen. The software update also includes a number of useful libraries of EQ and Dynamics settings. The new Library function lets a user select any set of parameters currently in use – maybe EQ setting on a group of channels handling drum-kit mics - to be stored and recalled from an internal library. The contents of this library can be transferred using a USB memory stick. (However, this process is independent of archiving/recalling Show Files of the entire Vi6 system settings.)

Other important enhancements include linking of Master output graphic and parametric equalizers to streamline LR or LCR EQ adjustments, plus individually adjustable left/right Pan and Gain controls on stereo input channels. Equally essential, a new auto-backup system ensures that, in the event of power loss interrupting a show, the last console settings are retained and restored automatically when power returns, with no audio changes. (Also planned for March 2008 release are support for multiple stage boxes and offline parameter editing. Phew!)

THE BOTTOM LINE

Put simply, the Vi6 Large-Format Digital Live Console, with a price tag of $83,250, is extremely versatile, sounds great and is extremely easy to use. Offering a fully optimized control density, the third-generation console's compact user surface features a unique integration of touch screens and encoders that dramatically reduces complex mental mapping. The system's Vistonics II touch screen user interface, which places rotary encoders and switches directly onto an ergonomically designed touch screen, is a revelation. It remains uncluttered yet is powerful enough for even the most complex live-mixing assignment, with a strong analog-like look and feel that results in a shallow learning curve. The combination of 32 Group/Aux/Matrix busses with direct outputs and LCR and Left-Right/Stereo busses is very powerful. Soundcraft's ultra-low noise analog mic pre-amps ensure pristine signal capture, while Studer's 40-bit floating point engine handles the DSP heavy lifting. And with a continuing software upgrade path, users are assured of remaining competitive in the heady world of all-digital console technologies.

The Soundcraft Vi4 Format Digital Live Console offers the functionality and facilities of the Vi6, including Vistonics II and FaderGlow, but in a smaller, more compact footprint aimed at space-conscious applications within theaters, houses of worship and venues with smaller mix positions. Measuring less than five foot wide, the Vi4 provides access to 48 inputs on 24 faders, with a total of 27 output busses available for use as masters, groups, auxs or matrices.

My sincere thanks to Tom Der, Soundcraft USA's National Sales Manager, for providing access to a Vi6 live-sound console at Harman International's corporate HQ in Northridge, CA.

Mel Lambert has been intimately involved with production and broadcast industries on both sides of the Atlantic for more years than he cares to remember.

by **Strother Bullins**

Go To A Show!

Photo credit: Doug Rice/Allure

Last night I saw two great performers and friends do what they love to do — play music to a loud, enthusiastic rock 'n roll crowd. I took the relatively short drive from my home in Northwest North Carolina to Charlotte in order to see The Dreaming (www.myspace.com/thedreaming) — a Los Angeles-based rock act led by Christopher Hall, former lead vocalist of Stabbing Westward — currently finishing up a 60-plus date national club/theater tour.

Friend number one, Carlton Bost, plays guitar for The Dreaming; I met Carlton in college and we've been very close friends for well over a decade. Other than when he was a student, I've never known a time when he wasn't a professional musician. He's played venues such as Madison Square Garden, the LA Forum, etc., etc.

Friend number two, Jesse Garber, is the vocalist for The Dreaming's local support act, Audacity (*www.myspace.com/audacitycharlotte*). I met Jesse as a 'little brother,' a high school kid and younger sibling of another very close friend who happens to be an incredible singer himself. Jesse has also become a very close friend of mine. I've never known a time when Jesse wasn't talking about or interested in music. Now he's rocking stages like a pro; who knows where he'll end up performing in the future.

Other than the fact that these guys are two of the most passionate, kind, and cool people I know, they're also great contributors to my knowledge of our industry. Carlton really is that guy deciding what instrument, amp, or microphone to pick up and play on the national stage; Jesse really is that guy deciding what microphone best works on his voice for his local (and potentially regional and national) stage. I can't think of better examples of the modern pro musician/recordist and the modern aspiring-pro musician/recordist than this pair.

On this particular night, both Carlton and Jesse were quite complimentary of the house engineer and house PA at Amos Southend, the stalwart local rock club that hosted the tour. FOH and monitors sounded great; as a result, both guys and their respective bands played great sets and felt great about it afterwards.

Pro audio gear manufacturers may often ask, even if only rhetorically, "What do guys like this — the aspiring pro, the pro, and the house engineer — really need?" My answer? Go to a show and ask them. I'm sure they'd be happy to share. And while you're there, you could probably even thank them for their business.

BUSINESS BEGINS WITH A SHOW

After all, our business is all about 'live' — whether you're a professional in the studio, out in the field, or providing live sound to the masses. For the perfect example, look no further than a guy like John McBride, truly a Professional Sound Man at the top of his game, featured in the March issue of PAR. In this particular touring feature written by *PAR* contributor Heather Johnson, McBride was highlighted as a successful live sound engineer for a major country music artist and his wife (the same person, actually). As most of you should know, McBride has been running an incredible live sound business for years. And today he runs the biggest and baddest recording studio in Nashville — Berry Hill, to be exact. (Shout out, Berry Hill!) Anyway, McBride is a guy who made his fortune in live sound and now spends his fortune on his studio. (Ask any pro freelancer in Nashville about Blackbird Studios — they'll tell you.)

Guys like McBride are incredibly good for the industry, mainly because guys like McBride get it: you have to reach out to the music community, in person and on location, to make such a good living as an engineer. Inside the recording studio is a great place to be, but 'live' is where the music is always happening. Be the best live sound engineer in town, and you'll know every good musician in town that needs to record an album. Then, in no time, all the musicians will be at your place to record.

TAKING 'LIVE' WITH YOU

The burgeoning handheld digital stereo recorder market is arguably the hottest segment in pro audio product sales today. There may be many reasons why seemingly every manufacturer in pro audio that has ever made a hardware-based recorder is now making a tiny stereo recorder with built-in mics and a USB port.

The Olympus LS-10

Back on topic, and to tie this one up: last night, while rocking in Charlotte, the most important and lucid reason for these products came to mind — they allow you to easily take the fun of 'live' with you. To be fair, all of these products sound pretty darn good — at least the ones I've heard and/or used — and, not to spoil my upcoming full review of the Olympus LS-10 PCM Recorder, the best among them are incredibly easy to use.

Thanks to a kind house engineer, I parked the small, intuitive LS-10 perfectly center of FOH for the duration of the show, and I walked away at the end of the evening with audio so good that either band could immediately and proudly play it for absolutely anyone.

In this day and age of MySpace.com immediacy, products allowing this sort of instantaneous content creation are windfalls for everyone: manufacturers, musicians, and music fans everywhere — at home, at the studio, or at the live show.

Strother Bullins is the Reviews and Features Editor for Pro Audio Review *and a former Berry Hill resident.*

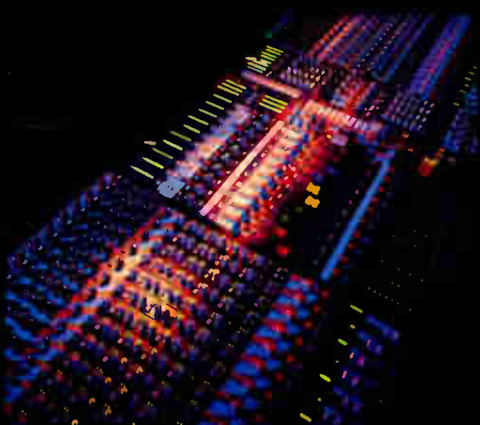
XL8
digital goes Midas

DL431
3- way digital mic splitter

HERITAGE 4000
monitor evolution

HERITAGE 3000
the industry standard

HERITAGE 2000
dedicated front of house



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LEGEND 3000
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SIENA
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design
performance

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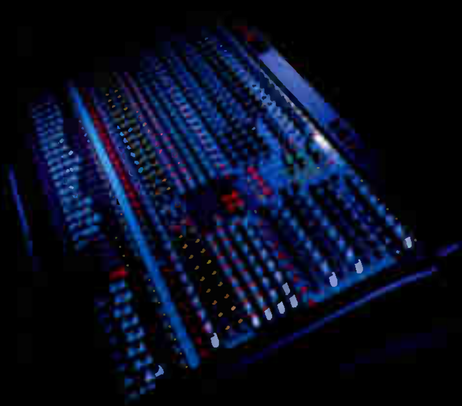
VERONA
8 buss excellence



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CONTRACTING

The latest news and products

NEW PRODUCTS

AVLEX Superlux PRA-618M Professional Gooseneck Microphone

The Superlux PRA-618M professional gooseneck microphone has been tailored specifically for reproduction of the human voice and is targeted for a range of installed sound projects — conference facilities, boardrooms, lecture halls and houses of worship. The PRA-618M features a miniature dynamic capsule that has the advantages of a supercardioid polar pattern. Reputed benefits of this design include a tailored vocal frequency response and the ability to pick up from a greater distance while delivering higher gain before feedback. The microphone has a frequency response range of 100 Hz-15 kHz and a sensitivity rating of -56 dBv/pa @ 1.6 mv/pa. Its rated impedance is 500 ohms, while the maximum SPL rating is 140 dB (THD 1% 1,000 Hz).

PRICE: $46

CONTACT: Avlex Corporation | ☎ 816-581-9103 ➲ www.avlex.com

LAWO mc2 56 Digital Audio Console

Lawo has introduced the new mc2 56 digital audio console, a smaller desks offering the performance of the larger mc2 90 and mc2 66 consoles. Based on the existing mc2 technology, the mc2 56 offers the Lawo HD core with up to 512 DSP channels, 144 summing buses and a routing capacity of up to 8,192 crosspoints. The complete functionality of the mc2 series is available, including the transfer of snapshots within the mc2 family, as well as dynamic automation and networking with other Lawo products. The newly designed control surface offers direct access to essential operating elements. Functions that are rarely used are handled via the touchscreen graphical user interface. In addition, the new construction reduces fader width to 30 mm. Every 16-fader bay offers fully featured metering on high resolution TFT.

PRICE: POA

CONTACT: Lawo North America Corp. | ☎ 888-810-4468 ➲ www.lawo.ca

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LIGHTVIPER Shadow Fiber Optic Transport System

FiberPlex, Inc. has announced the LightViper Shadow, a combined media fiber optic transport system designed for live sound and broadcast production, fixed media installations and remote recording applications. The 2.5-GHz system will handle a variety of media, including audio, intercom, Ethernet, RS422/232/485 control, composite video (EIA-250C) and TTL data. With optional component modules, DMX lighting control, MIDI, CANBUS and other control data can also be handled. The Shadow's modular components may be used independently or in combination to achieve a virtually unlimited variety of system designs for transporting various media data over a fiber optic network (up to its maximum 2.5-GHz bandwidth with a total of 256 bi-directional audio channels at speeds up to 192kHz).

PRICES: $8,500 to $30,000

CONTACT: FiberPlex, Inc. | ☎ 301-604-0100 ➲ www.lightviper.com

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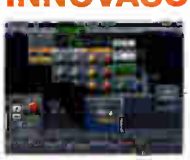
INNOVASON FM-8VB Effect Module



The release of Sensoft v12.0, InnovaSon's latest version software, coincides with the release of the internal effects module, the FM-8VB, which is now shipping as a standalone unit to be added to existing consoles, and as part of the company's new Pro FX console package. Sensoft 12.0 offers management of the range of effects that comprise the FM-8VB. Compatible with the range of the company's audio racks, FM-8VB has four effects engines installed between each pair of I/Os, all of which are configurable to receive the algorithm of choice; all the I/Os are accessible via the patch grids within Sensoft. From there, the user declares I/Os as insert points, or to feed inputs via an aux on the console while using a temporal effect such as reverb, echo, and so on.

PRICE: POA

CONTACT: InnovaSon | ☎ 888-DIG-DESK ➲ www.innovason.com

NEWS

The new Jaffe-Holden Acoustics-designed Dell Hall of the Long Center for the Performing Arts in Austin, Texas is the new home of the Austin Symphony and their chief recording engineer, Andy Murphy. **Sennheiser**'s new MKH 8000 Series have quickly become the favorite orchestral

recording mics for Andy Murphy, chief recording engineer for both the Music School Recording Studio at University of Texas and Austin Symphony delivering clear, un-hyped high-end with a "ribbon" like low-end.

Granite Rocks Live, a fast-growing audio production company is located in Southern New Hampshire not far from Boston, a hotbed of live musical and concerts. In business for five years, Granite Rocks has been mastering the art of multiple inputs and multiple mixes for stage shows with the help of **APB-DynaSonics**. "We added the Spectra Ti48 console specifically to handle some very large channel count shows we had schedule for the fall and spring." comments Devino. "The very fist show we used it on was Stagecoach Production's performance of Parade in Nashua New Hampshire, which used all 48 inputs."

Further reinforcing its reputation as a world leader for microphone solutions, **Audio-Technica** was selected to provide microphone products for Pope Benedict XVI's recent visit to New York City, which took place April 18-20, 2008. Audio-Technica provided a wide range of microphones for Benedict's appearances and related events in New York City, even going so far as to create special white-colored microphones and accessories, in order to be consistent with the color scheme and solemn tone of these events.

API Vision Surround Mixing Console

FEATURES: 24 main, three stereo buses and a dedicated 5.1 bus; two 100 mm automated faders on each input, five-channel panning on each fader; comprehensive multi-format monitor facilities; re-settable switch assignment; API Legacy Series mic premaps,

equalizers and dynamics modules; custom built to specification; five-year warranty on all parts.
CONTACT: API at 301-776-7879, www.apiaudio.com.

SOLID STATE LOGIC AWS 900+

FEATURES: 24-channel; SSL G series compressor; SSL SuperAnalogue mic preamps; SSL Twin Curve four-band parametric EQ; 5.1 surround sound; compatible with DAWs; TFT touchscreens; VU meters; includes legs, Total Recall and AWSomation.
PRICE: $93,500.
CONTACT: Solid State Logic at 212-315-1111, www.solid-state-logic.com.

HARRISON Trion

FEATURES: Post, live or broadcast architectures; 64 – 360 digital audio channels; 48 main buses, 16 aux. buses, eight control groups, four stereo program

outputs; 16 – 56 motor-driven faders; 40-bit internal signal path; XEngine Native DSP with 32- or 64-bit floating point audio at 44.1, 48 or 96 kHz sampling rates; 8-Band Parametric EQ, dynamics with compressor, limiter and gate; TFT displays; IKIS technology; XRouter.
PRICE: $90,000 - $200,000.
CONTACT: Harrison at 615-641-7200, www.harrisonconsoles.com.

YAMAHA DM2000 VCM

FEATURES: 96-channel; 24-bit/96-kHz processing; onboard DSP effects; Virtual Circuitry Modeling simulates '70s analog; Interactive Spatial Sound Processing (iSSP);

LCRS, 6.1 surround sound; joy stick; 100 mm motorized faders; LCD screen; SMPTE; word clock; linkable; Windows/Mac PC control software; I/O expansion slots.
PRICE: $19,500.
CONTACT: Yamaha at 714-522-9011, www.yamahaca.com.

SOUNDCRAFT Ghost LE

FEATURES: Analog 8-bus with Mix B path; 10 auxes plus two stereo pair; 24/32-channel frames; 24-channel expander; four-band British EQ; ProMic preamps with 60 dB range; onboard computer; SMPTE timecode generator; 100 mm faders; machine control.
PRICE: Starts at $6,295.
CONTACT: Soundcraft USA at 818-920-3212, www.soundcraft.com.

STUDER Vista 5 Digital Mixing Console

FEATURES: Modular design; up to 240 channels; 24 bit/96 kHz; LCR, LCRS, 5.1

Studer Virtual Surround Panning; SCore Live DSP engine; Vistonic LCD screen control surfaces; 32 100 mm faders; talkback section; rackmountable I/O.
PRICE: Starts at $120,000.
CONTACT: Studer USA at 818-920-3212, www.studer.ch.

AMS NEVE 88RS

FEATURES: 96 analog channels; > 24 bit/192 kHz; Encore Plus automation; two motorized faders per channel strip, with fader starts and Overpress PFL; "Classic Neve" processing; enhanced spectral formant equalization; soft knee compression; 5.1 monitoring; 24-step precision volume; acoustically optimized frame.
CONTACT: AMS Neve at 425-454-9966, www.audio-agent.com.

EUPHONIX S5 Fusion

FEATURES: 224 x 168 paths at 48 kHz; DF66 DSP SuperCore engine with 38 channels, scaleable to over 100; expandable System 5 surface design with 24 multi-format faders, 8

touch-sensitive knobs per channel; eMix (with optional joystick); TFT multi-format metering; Euphonix converter, digital surround monitoring; EuCon Hybrid protocol for controlling up to five DAW workstations.
PRICE: Starts at $150,000.
CONTACT: Euphonix at 212-889-6869, www.euphonix.com.

MACKIE VLZ3 Series

FEATURES: 12 – 16 channels; compact and ergonomic; XDR2 Extended Dynamic Range mic preamps; independent EQ controls; ultra-clean summing bus with extended headroom; built-in universal power supply; rugged steel chassis.

PRICE: $389.99 - $1,099.99.
CONTACT: Mackie at 800-258-6883, www.mackie.com.

TRIDENT Series 8T

FEATURES: 16-, 24-, 32-channel frames; Trident S20/4T Celebration channel mic preamp; Series 80B replica EQ; 8-aex sends; monitor bus; mono

selector; analogue VU meters; optional meterbridge; talkback section; two-year warranty.
PRICE: Starts at $3,498.
CONTACT: Trident Audio at +44 1474-815-300, www.tridentaudio.co.uk.

BUYER'S GUIDE continues on page 68 ➤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Continued From Page 66

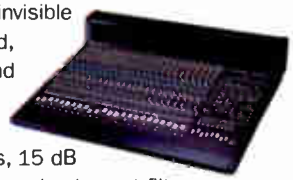
TL AUDIO M1 Tubetracker

FEATURES: 8, 12 input channel frames; tube preamps; three-band EQ with sweepable mid/bypass; two aux sends per channel; effects return; alternate monitoring options; 100 mm K Series faders; stereo VU meters; optional 8-channel ADAT interface; 24-bit/96-kHz mix output option.
PRICE: Starts at $4,320.
CONTACT: TL Audio at +44 1462-492-090, www.tlaudio.co.uk.

ALTO Typhoon4800

FEATURES: 48 inputs with 40 mic preamps (48V Phantom Power); inserts on every channel; four band parametric EQ with sweepable center; eight aux. sends/returns; eight subgroups; talkback functions; 66 pounds; includes plywood flight case.
PRICE: $4,718.
CONTACT: ALTO/The Yorkville Group at 716-297-2920 ext. 32, www.altopa.com.

BEHRINGER EURODESK MX9000



FEATURES: 24 channels; Balanced Mic/Line Input path with invisible mic preamp, pad, inserts, four-band EQ with semi-parametric mid-frequency bands, 15 dB boost/attenuation plus low-cut filter, pan pot, solo and mute; Mix B/Tape Return path with two-band shelving EQ, pan, level and mute controls; six aux. sends; 100 mm faders; ULN circuitry; LED meters; external rack-mountable PSU.
PRICE: $1,629.99.
CONTACT: Behringer at 425-672-0816 or www.behringer.com.

CALREC Omega with Bluefin

FEATURES: 160 channels processing paths packaged as 48 stereo and 64 mono channels on one DSP card; up to 64 dual layer faders; 19.6 minutes audio delay; 8 x mono, stereo of 5.1 groups; additional VCA grouping; 2 x main stereo or 5.1 outputs; 20 auxes; 48 multitrack outputs;
CONTACT: Calrec at 917-825-3728 or www.calrec.com.

LOGITEK Artisan Console

FEATURES: Router-based for versatile operation and easy selection of sources to faders; modular design allowing 2 – 30 fader; two master mixes; eight submaster mixes; four aux. mixes; 24 mix-minus busses; one-button capture/recall of console layouts including EQ and dynamics settings.

PRICE: Under $60,000.
CONTACT: Logitek Electronic Systems at 800-231-5870 or www.logitekaudio.com.

LECTROSONICS Venue Receiver System

FEATURES: Dock for up to six UHF receiver modules; 258 UHF frequencies; antenna multicoupler; reception compatibility modes; Digital Hybrid Wireless technology; compander-free audio with 400 Series transmitters, compatibility modes for older analog; LecNet2 PC interface.
PRICE: starts at $1,850.
CONTACT: Lectrosonics at 800-821-1121, www.lectrosonics.com.

PEAVEY Pro Comm PCX U-1002

FEATURES: 100-channel; true diversity; AutoScan technology resolving abrupt frequency changes; Channel Control System; LCD display; lightweight belt-pack transmitter; handheld, lavalier and headset configurations.
PRICE: $799.99 - $899.99.
CONTACT: Peavey Electronics at 866-443-2333, www.peavey.com..

REVOLABS Solo Wireless Microphone Systems

FEATURES: UHF system; 1.9 GHz; 128-bit encryption; wideband audio frequency response; MaxFlex technology allows wearable, tabletop or XLR handheld adaptor-type Solo Microphones; up to 16 simultaneous microphones; single-, four-or eight-channel versions.
PRICE: Starts at $249 per channel.
CONTACT: Revolabs at 978-897-5655 ext. 111, www.revolabs.com

SABINE SWM7000 Series 2.4 GHz Wireless Microphone System

FEATURES: UHF system; True Diversity 2.4 GHz Smart Spectrum technology; 70 channels; FBX Feedback Exterminator; parametric EQ; compressor/limiter; de-esser; Mic SuperModeling; ships with Audix OM3 capsule; handheld or lavalier/headset bodypack transmitters; rechargeable batteries; digital audio output.
PRICE: starts at $1,259.99.
CONTACT: Sabine at 386-418-2000, www.sabine.com.

ZAXCOM TRX900 Series

FEATURES: Digital Wireless system; time code stamped internal recording for back-up of transmitted audio; optional internal IFB receiver; graphic LCD display; stereo wireless transmission available.
PRICE: $3,800 per channel.
CONTACT: Zaxcom at 973-835-5000, www.zaxcom.com.

AKG WMS 400 Pro Wireless Microphone System

FEATURES: Available in two 30-Mhz bands with 1,200 frequencies each; up to 12 simultaneous systems in each band; Integrated frequency management system with automatic scanning; infrared link for frequency/setup data uploads.
PRICE: Starts at $479.
CONTACT: AKG Acoustics at 818-920-3212, www.akgusa.com.

SENNHEISER SKM 5200 Wireless Transmitter

FEATURES: For use with six Sennheiser and two Neumann interchangeable microphone heads; two channel banks, one fixed one variable; 20 pre-set frequencies and 20 programmable UHF frequencies in 5 kHz steps; sensitivity switchable in 1 dB steps; HiDyn plus noise reduction; backlit LC display for settings; automatic lock mode.
PRICE: $2,115.
CONTACT: Sennheiser at 860-434-9190, www.sennheiserusa.com.

SONY DWT-B01 Digital Wireless Transmitter

FEATURES: Compact bodypack transmitter accepts mic or line input (attenuator level: 0 dB to 48 dB in 3-dB steps: Selectable output power for stable and long-distance transmission (1/10/50 mW); Up to 12-channel operation (at 6-MHz bandwidth); and Easy-to-read, full dot-matrix organic EL display.
PRICE: $2500.
CONTACT: Sony Pro Audio 1-800-686-SONY or visit www.sony.com/professional.

SONY DWR-S01D Digital Wireless Receiver

FEATURES: World's first two-channel slot-in receiver: Mounts directly into PDW-700 HD 422 two-channel digital slot; Rear-mountable to HDCAM/XDCAM/Digital Betacam/MPEG IMX camcorders with DWA-01D adaptor; Supports analog or AES3 digital audio output; Rugged, lightweight compact design; and Easy-to-read, organic full dot-matrix EL display.
PRICE: $3,150.
CONTACT: Sony Pro Audio 1-800-686-SONY or visit www.sony.com/professional.

SHURE KSM9 Vocal Microphone

FEATURES: Dual gold-layered. low-mass Mylar diaphragm; switchable supercardiod/cardiod patterns; premium electronics; gold-plated connectors; champagne or charcoal grey; 50 Hz - 2- kHz response.
PRICE: $850.
CONTACT: Shure Inc. at 847-866-2200, www.shure.com.

AUDIO-TECHNICA Artist Elite 5000 Wireless Microphone System

FEATURES: UHF system; up to 200 channels; True Diversity reception; dual compander; Tone Lock squelch; Intelliscan; handheld, bodypack transmitter packages; dynamic and condenser handheld transmitters.
PRICE: Starts at $3,119.
CONTACT: Audio-Technica at 330-686-2600, www.audio-technica.com.

CARVIN UX1000-MC

FEATURES: 960 user-selectable channels with diversity UHF receivers; four assignable groups; Diversity PLL Synthesized technology; two antennas; handheld microphone; low-battery indicator; mic on/off status indicator.
PRICE: $429.95.
CONTACT: Carvin at 858-487-1600, www.carvin.com.

BROADCASTING
| Feature

GRAMMY Continued From Page 43

an LCR mix with the vocal isolated in the center, so that we would to be able to put his three channels at unity onto our Production desk, and we would have his mix. It worked just as expected, and his balance was superb."

"Our biggest challenge came with the decision to supply only a 5.1 soundtrack," Neuberger concludes. "How should we best monitor the various mixes and downmix compatibility at our most mission-critical locations? With wonderful support from Dolby Laboratories we were able to provide our online and offline music mixers and our production mixer with the same encode/decode units, carrying the same metadata settings as CBS used at [New York] Master Control. The pragmatic approach was to have our mixers prioritize monitoring in 2.0 - the downmix - which is the way most of the audience listens. Both John [Harris] and Eric [Schilling] were disciplined in this regard, and checked both 2.0 and 5.1 throughout the performances."

Dolby Laboratories supplied a number of monitoring units to the production, including LM100 Broadcast Loudness Meters, DP570 Multichannel Audio Tools and DP563 Dolby Surround and Pro Logic II Encoders. "The DP570 let us check metadata settings and compare our discrete 5.1 mixes with other delivered mixes and decoded surround balances," Holmes says. "We could to monitor exactly what the consumer would ultimately hear." For added confidence, a DP563 was configured exactly as a unit being utilized by CBS New York, and used to encode the discrete surround broadcast mix into Pro Logic II Lt-Rt for monitoring of stereo and mono compatibility. Each of the music and production mixers used an LM100 meter to assure that the program loudness matched the Dialnorm metadata setting used by the network, which translates to the best possible performance of AC-3 audio that accompanies HDTV broadcasting.

ATK SOUND REINFORCEMENT - JBL VERTEC ARRAY WITH QSC AMPLIFICATION

ATK AudioTek once again flew a JBL VerTec system powered by QSC amps within the Stapes Center; XTA Electronics DP226 processors provided system equalization and routing. The four main clusters were composed of a dozen VT4889 line-array cabinets, augmented by an array of 12 JBL VT4880a subwoofers flown above the center of the main stage. QSC PowerLight 4.0 amps fired the HF/MF cabinets with PowerLight 9.0 models for LF cabinets and subs. Three clusters of delay loudspeakers comprised a total of 22 VerTec 4889 Line Array Elements, plus a dozen ATK C-6 three-way cabinets.

Two Yamaha PM1-D Digital Production Consoles handled front-of-house; two more PM1-Ds provided monitor mixing for the performance stages. Ron Reaves helmed the right-hand PM1-D console and delivered a stereo/L-R music submix, subwoofer/LFE mix and a vocals stem to ATK Vice President Mikael Stewart, who operated the main PM1-D and added stage announcements, prerecorded tracks and other elements for the stereo house mix, plus delays and subwoofer. Tom Pesa and Michael Parker developed stage-monitor mixes.

According to Jeff Peterson, ATK AudioTek Design Engineer, "We put up 24 QSC PowerLight amps onto a flown platform [above the mix position and camera platform] to power the nine delay clusters. Since these amps require the Spiderman routine to get to, we prefer to control them remotely. We established a fiber Ethernet network between FOH and the amp platform. Then, using QSC's CM16a, we are able to control level, muting and power of the PowerLight amps from the ground. This year, we replaced the CobraNet network used to drive the delay amplifiers with a new [Media Numerics] RockNet 300 digital snake system. RockNet performed flawlessly - we had it set up in a completely redundant fashion, where any single device failure would not affect out audio path."

"As always, our biggest challenge was the room," offers AudioTek's Sound System Manager Andrew "Fletch" Fletcher. "Staples Center is big and cavernous and certainly sounds that way. The suites between the 200 and 300 level cause some very unpredictable reflections and resonances. We point the main PA down to cover the floor and the 100 level only, hence missing the suites. We can pretty much crank that up as much as we like without causing too many problems. The side-pointing clusters and the delays cover the suites and some of the 300 level. We adjust those so that reflections to the main floor are hardly noticeable while still giving good coverage to the people in the suites and the 'nose bleeds'. Everything else went very smoothly this year."

"The wonder of the Grammy Awards is due in equal measure to the pageantry and to the technical complexity of producing one of the world's most famous events in live entertainment television," concluded John Cossette, Executive Producer of the 50th Annual Awards Ceremony. "We have consistently expanded the impact of the Grammy's to our viewing audience."

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

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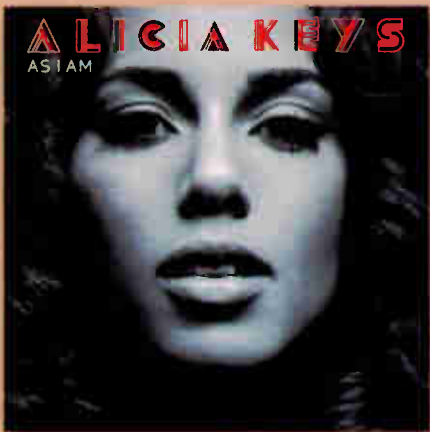
Single Slice

by Strother Bullins

"Superwoman" | Alicia Keys



SINGLE: "Superwoman"
ALBUM: As I Am (RCA)
DATE MASTERED: Over four months over the Summer of 2007 at The Oven Studios in New York City
PRODUCERS: Alicia Keys, Kerry "Krucial" Brothers, Linda Perry
ENGINEERS: Manny Marroquin and Ann Mincieli with Chad Franscoviak and assistance from Keith Gretlein, Glenn Pittman, Zach Hancock, Kristofer Kaufman, Vincent Creusot, Brendan Dekora, Seamus Tyson, and Seth Waldmann.
MIXERS: Manny Marroquin with assistance from Christian Baker and Jared Robbins
MASTERING ENGINEER: David Kutch
OTHER PROJECTS: Kutch has mastered a wide variety of music for artists including Whitney Houston, Lyfe Jennings, Sarah McLachlan, Anthrax, Kanye West, The Roots, Luther Vandross, and others.
SINGLE SONGWRITER: Alicia Keys, Steve Mostin, and Linda Perry
MASTERING MONITORS: Focal Solo 6 powered nearfields
MASTERING WORKSTATION/RECORDERS: Magix Sequoia sequencing and editing software, Pro Tools|HD, and half-inch Ampex analog two-track recorder with Aria heads
SELECT MASTERING PROCESSORS: Dangerous Music Monitor; Dangerous Music Master; Prism Sound A/D and D/A converters, Manley Pultec EQP-1A EQ, Prism Maselec MEA-2 mastering equalizer, API 2500 compressor, GML 9500 Mastering EQ plug-in.

Strother Bullins is a North Carolina-based freelance writer specializing in the professional audio, music and entertainment industries.

ENGINEER'S DIARY

Last summer, mastering engineer David Kutch received the sad news that his digs, NYC's Sony Music Studios, would be closing its doors. But as serendipity would have it, Kutch was considering his next permanent spot when a mobile mastering job became his most important gig of the year.

David Kutch

"At that point I got a call from Alicia Keys and her engineer, Ann (Mincieli), to book me for mastering," Kutch recalls. "Of course, I was happy to be booked for the album, but Sony was starting to look like a ghost town. Then, about three days later, they called me back and asked, 'Would you be interested in doing it here?' I went out, took a look, and found the best place to master this record — in the live room (at Oven Studios, Keys' private recording facility). It's a great sounding room designed by John Storyk. So, I said, 'I'm in.'"

Kutch's entire rig — centered on a Dangerous Music Master and Dangerous Music Monitor combo — was transplanted to Oven for a four-month run. "The beauty of the whole scenario was that there were two studios there," he offers. "I could be mastering something, go into one of the rooms and listen, go out to my car and listen, go to (producer) Kerry Brothers' Range Rover and listen, and so on. We had tons of listening environments. Meanwhile, (engineer) Manny (Marroquin) was mixing upstairs while I mastered downstairs. We'd just go back and forth."

Other than having all the time in the world, the team enjoyed the peace of mind of knowing their masters were safe. "We're all so proud that the album never leaked," says Kutch.

With Keys' larger live tracking room serving as workspace, Kutch realized that his normal large monitors would be out. "I knew that I would rely on nearfield monitors almost exclusively. About a month prior, I had discovered the Focal Solo 6 monitors. They really saved my butt on this project."

All songs were printed to both analog half-inch tape and back to Pro Tools|HD. "It was all about vibe," he explains. In fact, for the mastering of 'Superwoman,' the final came via a few plug-in squeezes in Pro Tools to tape, then back from tape into Pro Tools. "I tried anything and everything. It's about whatever I have to do to get this sound."

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