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DECEMBER 2006

ProAudio Review

The Review Resource for Sound Professionals

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Midas XL8 Digital Console

Best Gear of the Year!

2006 Reviewer's Pick Awards

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Reviewed:

- **Mytek Digital 8X192 Converter**
- **DPA SMK4061 Stereo Microphone Kit**
- **Headphone Nirvana from AKG and Benchmark Media**

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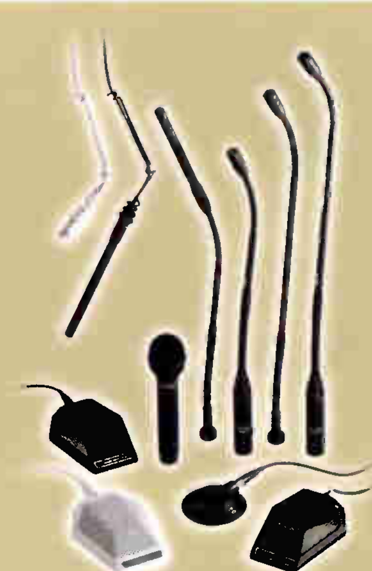

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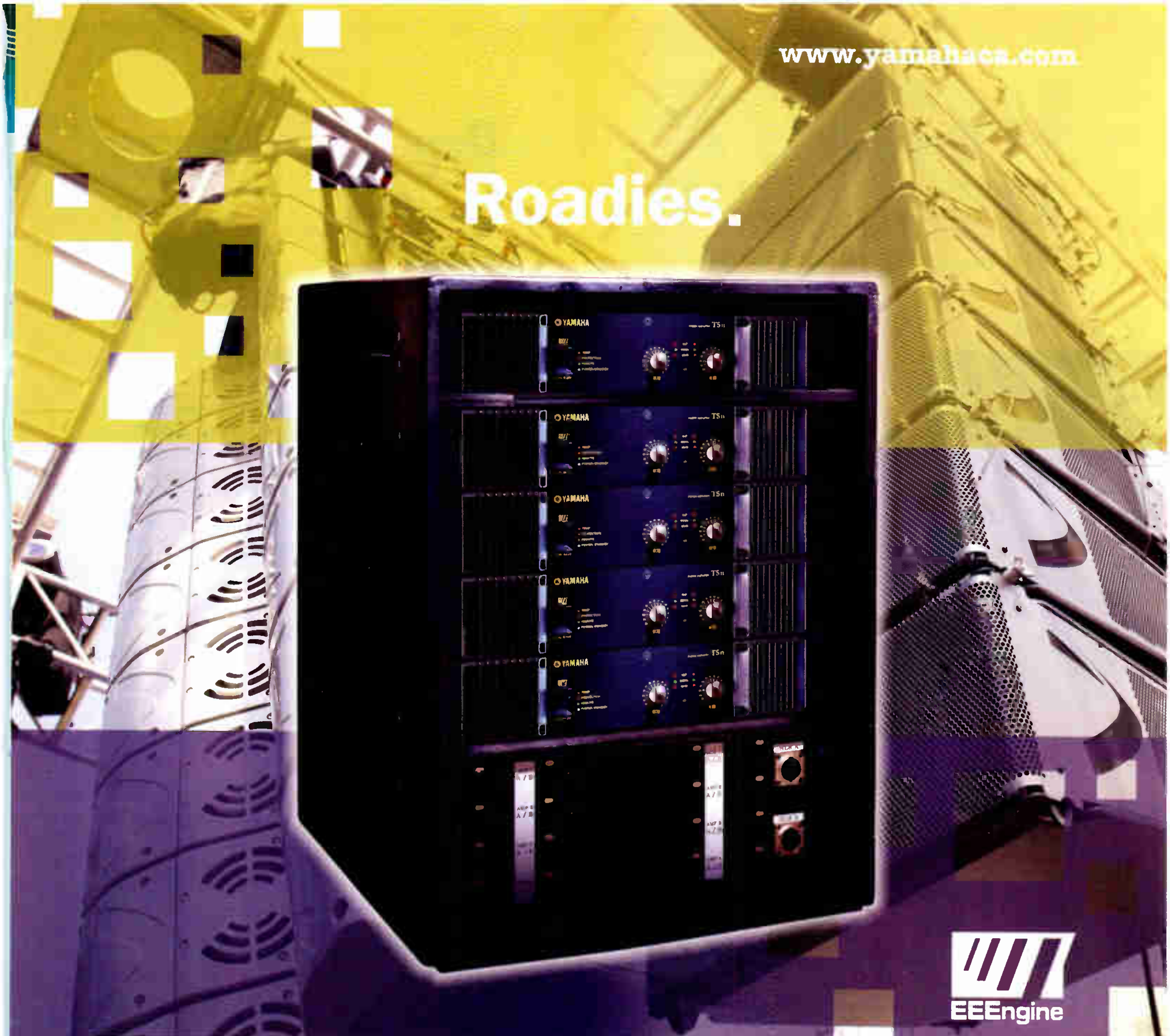
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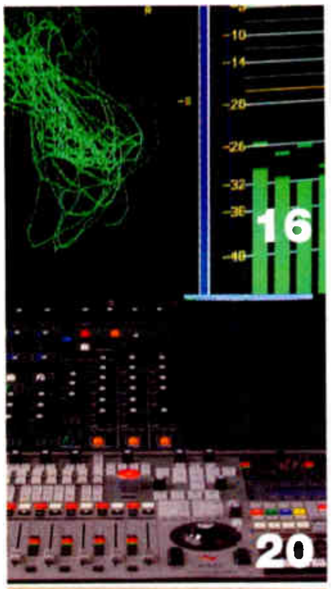

Evaluating audio products for professionals in commercial recording, broadcast production, audio for video/film, project studios, live sound, contracting and multimedia.

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Dionne Warwick (with Mya) "Close to You"
by Strother Bullins

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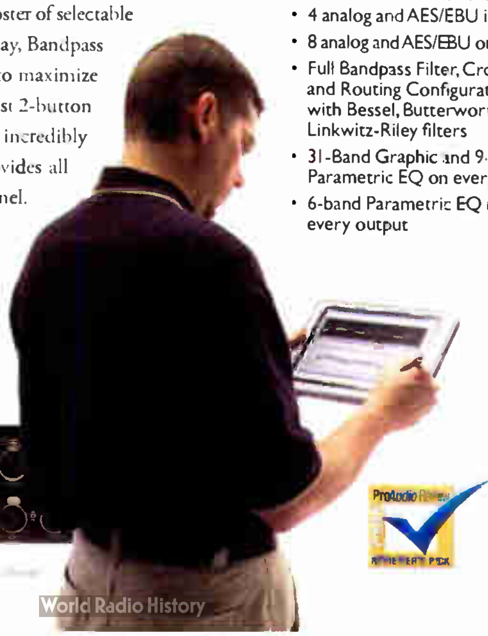

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ProAudio Review PREVIEWER'S PICK

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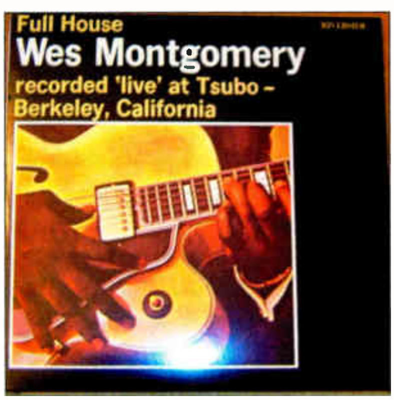
FROM THE PUBLISHER

John Gatski

For the Record

I recently was sent a new phono cartridge to check out from Audio-Technica. Actually, it is an improved version of its ML150 moving magnet cartridge that came out in the 1990s.

Despite digital audio advances over the last 25 years there is a persistent and even slightly growing element out there that still listens to vinyl. Albums are still released on vinyl; turntables and cartridges continue to be made. Even mid-priced turntables are cropping up from companies such as Music Hall, Thorens, etc. LP- fanatic companies — like Acoustic Sounds in Salina, Kansas — not only sell audiophile pressings (and DVD-As, Dual Discs and SACDs), but actually produce new pressings of old classics, as well as recording and pressing newer releases, onto vinyl. They even sell turntables and hi-fi gear so you can hear them at their best.



I have to admit that when you play a well-made audiophile LP (direct-to-disc, half-speed mastered on high quality virgin vinyl) without the high noise and ticks and pops, there is something in the audio that often sounds "more real" than an original digital recording. With all the advances in accuracy and digital resolution, there is something still appealing about the analog LP — even with all of its faults.

Recently, I have been having fun combining LP technology and digital technology. I have been doing transfers of records at 24-bit/96 kHz (or 192 kHz). To my ears, the high resolution digital copying of the first playback of an LP is more open than a CD dub of the LP.

I have been buying up great sounding pressings of jazz recordings from Acoustic Sounds. For example, I recently paid $50 for Wes Montgomery's *Full House* — a live recording from 1963, remastered and pressed onto four sides of two LPs at 45 rpm. Besides the virtuoso straight-ahead jazz guitar performances by Montgomery and his band, captured on tape by the Riverside Label, the analog sound is magnificent. The CD version sounds decent, but the LP is better.

I made a great sounding dub of the LPs with my LP dubbing chain, which consists of a Rotel belt drive turntable, the A-T cartridge, my custom Audio-by-Van Alstine FET Valve preamp with tube phono preamp, Benchmark ADC1 A/D, TASCAM HD-P2 digital recorder and a set of Kimber cables.

Since an LP's biggest negative is the stylus destruction of the fragile high frequency grooves, I recorded the records on the first pass. I played the four side LPs onto my trusty Rotel belt drive turntable with the new A-T cartridge, tracking at 1.25 grams. I took the preamplifier audio signal into the TASCAM HD-P2 via the Benchmark A/D at 24/96. I then took the digitally-copied LP cuts, transferred them to my G5, and burned the vinyl-to-digital tracks onto DVD-A, preserving the best possible playback of a record before repeated plays eventually wear down the grooves.

The digital transfer does a very good job of preserving the analog sound's seductive simplicity and the open, smooth top-end of a well-cut record. Funny thing about digital — I think it does a great job at preserving original analog recording.

In this issue we have a look at the new Midas XL8 digital board by Dan Wothke. Some of you know Dan from his work with our sister magazine, ***Audio Media***. Welcome aboard, Dan.

John Gatski is the publisher/executive editor of ***Pro Audio Review***. *His first record player was a Christmas gift in 1968 — a red/white checkered Sears Silvertone manual play with the speaker mounted under the tone arm — and a brand new copy of the Beatles' "Yellow Submarine."*

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MMADP BPA WORLDWIDE SPARS

NO, SAY WHAT YOU REALLY THINK

Caught your article in the July 2006 issue of *Pro Audio Review* ('Ode to the Record Store,' Publisher's Page).

What I see happening in the music business is more a paradigm shift in the way they think about customer service (in terms of eliminating it) and the quality of employees they hire... or don't hire because the bean-counter says the knowledgeable body is too expensive for their taste.

What few stores that are left are catering to the Wal-Mart model of sales - i.e. go only for the volume sales and to hell with the little stuff because of cost-of-sale on them is too high. I understand it, and see it as a symptom of a far bigger problem. Look at the store rents and property taxes these days, and it doesn't take long to realize that a small operation has the same fixed expense every month that ends up being a good chunk of their sales before they even make any money for themselves. When you're working three out of five days for the government, you gotta do what you gotta do!

Now as far as the music companies go, they have so dumbed-down the music that quality (to them) is a non-issue. Artist development is too expensive for them, and they wish to add to the bottom line as much as they can each quarter, so quality cannot be a part of the business plan. It is the ultimate goal of style-guy management. (Style guys are anyone who came up through sales, marketing, or finance and know little about the day-to-day operations of the business itself. They wear expensive suits, are constantly working on their golf swing, drive expensive, leased Nazi cars, and in general are leveraged to the max.) In other words, they know the cost of everything and the value of nothing. A friend of mine once said that when Madonna and Mariah Carey tracks are the best-sounding and best-produced tracks available, there is something wrong.

I have given up on big music to offer me quality. I make a point of rescuing LPs and CDs all the time, and generally clean them up digitally before transferring them to CD. This has greatly enhanced my CD collection with titles long out of print that will never see the light of day ever again. So far, I have about 6,000 LPs, 800 CDs, 400 cassettes, and countless 45s and 78s. That's a lot of music, yet I would be willing to BUY more if I could find it on a real company-issued CD that didn't sound like it was overprocessed with a

LETTERS Continued On Page 60

STUDIO

The latest news and products

NEW PRODUCTS

SUPERLUX S241/U3 Microphone

The S241/U3 from Superlux is a true capacitor cardioid condenser pencil microphone. The diaphragm is 1/2-inch gold-evaporated and 3 microns thick. There is also a 10 dB/20 dB pad and a 75 Hz/150 Hz low-cut filter.

PRICE: $125.

CONTACT: Superlux/Avlex | ☎ 877-447-9216 ➲ www.avlex.com

RUPERT NEVE DESIGNS Portico 5043 Limiter/Compressor Duo

Yet another module from Rupert Neve Design's Portico line of half-rack processors, the 5043 is a two-channel compressor/limiter. Controls include threshold, attack, release, ratio and makeup gain. The channels are linkable. The 5043 also features a feedback/feed-forward design for increased musicality.

PRICE: $1,895.

CONTACT: Rupert Neve Designs | ☎ 512-847-3013 ➲ www.rupertneve.com

SM PRO AUDIO PM-8 Passive Summing Mixer

If you have a few too many channels coming towards you and need to do some summing, especially you DAW users, you might want to give the SM Pro Audio PM-8 Passive Summing Mixer a look. The PM-8 is an eight-channel rackmount mixer with level and pan controls per channel. It has stereo outputs along with a 25-pin D-sub.

PRICE: $569.

CONTACT: SM Pro Audio/Kaysound | ☎ 514-633-8877 ➲ www.smproaudio.com

DIGIDESIGN Mbox 2 Mini

Claiming to "put the power of Pro Tools in the palm of your hand," Digidesign's latest member of the Mbox interface family is the Mini. The Mini is a two-input, USB-powered interface for Pro Tools LE. Extra features include a headphone output, 1/4-inch instrument inputs, 48V phantom power and 24-bit/48 kHz converters. A hefty software bundle ships – including software tools and plug-ins from Ableton, Properllerheads and IK Multimedia.

PRICE: $329.

CONTACT: Digidesign | ☎ 800-333-2137 ➲ www.digidesign.com

NEWS

New Orleans-based Dirty Dozen Brass Band used **Royer Labs** R-121, R-122, SF-12 and SF-24 microphones on their latest album, What's Going On?

Talk about hybrids! A customized **Yamaha** Disklavier Pro piano, utilizing **Zenph Studios** digital performance recreation technology, was used to "recreate" Glenn Gould's 1955 performance of Bach's Goldberg Variations for a broadcast by the Canadian Broadcast Co. and a subsequent Sony BMG Masterworks album. A Yamaha PM1D digital console inside a CSP Mobile Productions truck was used for recording Stacey Ferguson singing her new single, "London Bridge," in Portland, Maine. See picture of Yahoo Music's James Kelly and mixer Lance Vardis.

Shure donated a number of SM57, SM58, Beta58 and KSM microphones to New York University's Clive Davis Department of Recorded Music.

San Francisco-based engineer Dustpan has installed a **Geoff Daking** front end for his DAW system. He's installed Daking Mic Pre IVs, Mic Pre/EQs and FET II compressors.

The Counting Crows are using a **Mojave Audio** MA-200 condenser microphone on their upcoming new album.

On their new album, Ringside is using a **Telefunken** USA 251E tube microphone. See the picture of drummer Norm Block and singer Scott Thomas. Thomas has the mic while Block has an original 45+ year- old Telefunken Ela M 250E that is being refurbed.

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ProAudio Review 2008 REVIEWER'S PICK

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STUDIO | REVIEW

by Dave Martin

Mytek 8X192 A/D-D/A Converter

A great-sounding, top-of-the-line converter (with a DSD option).

In the early days of digital recording, most of us didn't even entertain the idea that the converters that came with our recorders were less than they should be. But gradually, folks started to notice that some digital recorders sounded, well, "better" than others. And some sounded worse. Much worse. And so a market developed for standalone converters - at first stereo, but followed soon afterwards by multichannel converters that allowed the engineer to move beyond those offered by the manufacturers of recording equipment. It may seem a bit paradoxical to say that the wide variety of high quality A/D and D/A converters available to today's engineers is due (at least in part) to the sound of Panasonic DAT machines, Digidesign's 888/24 and the original ADAT. Nevertheless, there is a certain element of truth to the concept. One member of the current generation of converters is Mytek's 8X192 converter ($3,495 basic).

| FEATURES

Mytek's 8X192 AD/DA is a single rack-space converter that offers eight channels analog and digital conversion at all rates from 44.1 kHz to 192 kHz. Optional firmware is available to provide both standard and high speed DSD. Both analog and digital I/O is on DB25 connectors, with the analog I/O conforming to the standard promulgated by TASCAM and used by most manufacturers. The AES/EBU digital I/O is on a single DB25 connector that follows the standard used by Digidesign (note that Apogee uses a different pinout than Digidesign). Mytek offers a number of digital I/O cards besides the optional DSD card, which allows the 8X192 to interface directly with Pro Tools|HD, ADAT, TDIF and Sonic HD. There is also a FireWire interface card.

The Mytek 8X192 includes a separate analog stereo output (on XLR connectors). This stereo output can be derived from any pair of channels, from all channels (with odd channels summed left and even channels summed right), or all channels summed to mono. The output of the buss/selector can be run through a precision 24-position 1dB stepped attenuator on the front panel of the unit or the attenuator can be bypassed. A headphone jack on the front of the 8X192 presents the same signal as the stereo outputs and is routed through the attenuator whether or not the stereo output is passed through the attenuator. The 8X192 also has an internal clock generator with multiple word clock outputs to allow synching with other digital equipment in the studio.

The controls and the metering on the 8X192 are logical and, considering the limited amount of real estate available on the front panel, well laid out.

| IN USE

Integrating a pair of Mytek 8X192s into Java Jive's Pro Tools|HD systems was a relatively pain-free operation, though the 8X192 offers enough options that some experimentation was in order; the A room has an HD3 with two Digidesign 192s (with optional D/A cards) and one Digital 192. Once I located a dealer for the eight pair of AES DB25/DB25 cables (these, by the way, are NOT something you can pick up at your local Guitar Center), I simply connected the 8X192s to the Digidesign Digital 192's AES ports, trashed all of the PT preferences and rebooted the G5. The added interfaces worked like a charm.

Since a stable word clock is an important issue when interfacing digital equipment, we explored a couple of avenues with regard to clocking the system. The A room's Pro Tools setup includes a Digidesign Sync I/O, which is normally used as the loop master clock source. We tried clocking the 8X192 from the Sync I/O, clocking the Sync I/O from the 8X192 (thereby letting the 8X192's clock be the master for the whole system), and even using the multiple clock outs from the 8X192 to provide a clock for all of the other interfaces. In the end, we finally went with a relatively simple solution – allowing the 8X192 to derive its clock from the AES/EBU signal, leaving the rest of the system alone (with the Sync I/O as the clock master). If there were sonic differences between the various clocking schemes, they didn't make themselves obvious.

The B room has a simpler Pro Tools sys-

FAST FACTS

APPLICATIONS
Studio, field, live sound, post production

KEY FEATURES
Eight-channel; 24-bit; up to 192kHz; AES/EBU digital output; word clock; I/O option cards; DSD option

PRICE
$3,495
DSD upgrade - $1,995; FireWire card - $795; TDIF card - $595; ADAT card - $595; Pro Tools|HD card - $795; Sonic HDSP/USP card - $1,595

CONTACT
Mytek | ☎ 646-613-1822
www.mytekdigital.com

tem; typically, a single interface is hooked up to the HD3. Since one of the two 8X192s used for this review included the optional Pro Tools interface card, the setup was simply a matter of plugging the DigiLink cable and the analog I/O cables to the interface, again

PRODUCTPOINTS

- Sweet, smooth sound from both D/A and A/D
- Interfaces well with a number of formats
- Multiple clock outputs
- Added balanced stereo output
- Low latency
- Can be upgraded to DSD

- Cost
- Runs fairly hot

SCORE

For those willing to pay a premium for high quality sonics, the Mytek 8X192 is a serious contender.

trashing preferences, and rebooting that system. In the B room, the clock on the 8X192 was set to internal.

Because the two rooms at Java Jive are set up for different tasks, 8X192s were used differently in each environment. In the A Room, both 8X192s were used in addition to the converters already present, making a 32-input/48-output system that interfaces to the 48-input D&R Cinemix console. In the B room - designed for editing, overdubs and ITB mixing - one 8X192 was used for monitoring, auxes and outputs to the mastering deck.

Once installed in the systems, the Mytek 8X192s performed very well – just as you would expect any high quality converter to behave. There were no issues moving from 96 kHz sessions to 44.1 kHz sessions to 48 kHz sessions, nor were there any glitches (of any sort) that could be blamed on the Myteks. Of course, the most important question concerns the sound of the 8X192, and there, it also behaved admirably.

On a couple of tracking dates, I routed incoming signals (specifically, piano tracks, organ tracks and lead vocals) through both the Mytek 8X192 and a Digidesign 192 for comparison purposes. The Digi 192 and the Mytek are calibrated slightly differently, with the Digi 192 set for +4 dB = -18 dBFS and the Mytek set for +4 dB = -15 dBFS. This difference means that incoming signals recorded by the Mytek converters are slightly hotter than the signals brought through the Digi 192. After compensating for the volume, comparing the two otherwise identical signals did reveal differences between the two tracks. In a direct comparison with the Mytek, tracks recorded with Digidesign's 192 seemed to have a certain hardness - not "harsh," but edgier than those recorded through the Mytek converters. The tracks recorded with the 8X192's A/D side also seemed to have a little more width than the Digi A/Ds.

This holds true on the D/A side as well - the Mytek converter sounded a bit fuller and (as much as I hate to say it) had more of a "natural" sound. In an A/B situation, it was pretty easy to tell the difference between the two converters when listening to solo piano tracks. Naturally, when a signal was recorded and played back through one brand of converter (either the

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STUDIO | Review

MYTEK Continued From Page 11

Mytek or the Digi converter), these relatively small differences were magnified.

To be fair, Digidesign's 192 converter is much better than its predecessor, the 888/24 (which was itself a heck of a lot better than the generation before it). And while there were noticeable sonic differences between the Digi and Mytek boxes, those differences aren't nearly as stark as they might have been with earlier generations of converters. Nevertheless, they are there. And though it's truly a subjective opinion, I preferred the Myteks to that of the Digidesign converters that I already own.

When using the 8X192 as the sole interface to a workstation, some other advantages of this box came into play; the stereo XLR output can be a real problem solver, and the multiple clock outputs

made synching the rest of that particular system a lot easier. Besides that, the sonics of the 8X192 made long days of editing drum tracks at least a little more pleasant.

I should mention a couple of other things about the 8X192 converters. First, they get quite warm, despite the built in (and pretty quiet) fan. It's recommended that a space be left above each unit for ventilation. Second, the 8X192's conversion latency is quite low; the manual doesn't specify a number but when comparing tracks routed simultaneously through the Mytek converter and Digidesign's 192, I found the Digi converter to be approximately 18 samples slower than the Mytek. This wouldn't be an issue (and it could be argued that it's an advantage) unless the DAW uses converters from different manufacturers (as my system does). But if, for example, you used one company's converter for some drum tracks and another brand for the rest of the drum tracks, this bears watching. Naturally, the same holds true for a stereo signal - it's not a good thing to split left and right into different brands of converters with differing latencies.

Admittedly, the Mytek 8X192 ADDA is not the most cost efficient converter available. But those who care about how everything in the signal path affects the sound should give serious consideration to the 8X192.

Dave Martin would like to thank Dave at Vintage King and David Seymour, the "go to" guy for Mytek Digital, for the use of the converters used in this review.

Dave Martin has been producing music for more than 20 years and has performed as a bassist for even longer. He has recorded bands too numerous to mention and thousands of "sound-alike" tracks and other production music for the karaoke and television industries. He is the owner of Java Jive, a recording studio 20 minutes from Nashville. He can be reached via his website, www.javajivestudio.com.

| REVIEW SETUP

Pro Tools HD3 with two Digidesign 192s and one Digidesign Digital 192, Sync I/O; D&R Cinemix console; Dynaudio Acoustics BM15A monitors

Wireless Innovations.

Sony has a history of breakthroughs in wireless microphones, including the origination of synthesized UHF technology Our newest models continue to lead the way. The WRR-862B camcorder-mounted receiver simplifies field production with two channels of reception. The MB-X6 tuner rack streamlines multi-channel sound with six channels in a single rack unit. The sleek WRT-8B body pack lets you choose high power for maximum distance or low power for maximum battery life. Sony's 800 Series also features diversity reception, legendary build quality and an expanded range of channels to help you navigate an increasingly crowded broadcast band. Outstanding simplicity and agility... that's wireless innovation.

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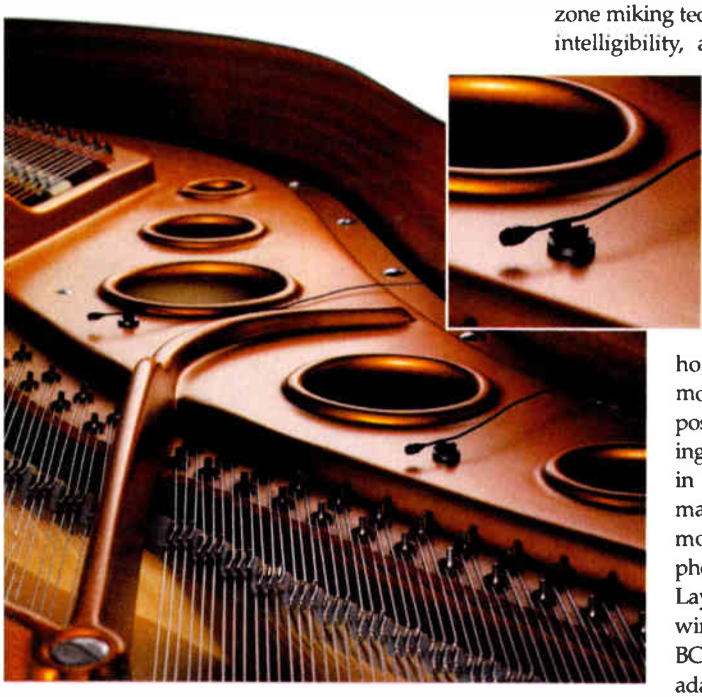
STUDIO | REVIEW

by Russ Long

DPA SMK4061 Stereo Microphone Kit

Two great mics are better than one!

DPA has done it again. With the release of the SMK4061 Stereo Microphone Kit ($995) there is now a way to record piano (and many other sound sources) at the level of quality you've come to expect from a DPA product at a price within closer reach.



| FEATURES

The new SMK4061 Stereo Microphone Kit is a flexible and cost-effective piano miking solution perfectly suited for both live and studio work. The package combines a pair of DPA 4061-BM omnidirectional miniature microphones with a variety of mounting accessories. The 4061-BM is the black version of the 4061.

The DPA 4061 is acoustically identical to the acclaimed DPA 4060 but the sensitivity is adjusted to 6 mV/Pa. The microphone's noise floor is 26 dB(A) re. 20 µPa and, if used properly, it can handle sound pressure levels up to 144 dB SPL before clipping.

Included with the package's mounting accessories is the BLM6000 Boundary Layer Mount, which can be used in conjunction with a 4061 (or any of DPA's miniature mics) to place the mic on a reflective boundary such as a piano lid, floor, wall, ceiling, etc. This enables the microphone to capture the area's ambient sound. The sound captured from this pressure zone miking technique has a higher sensitivity, intelligibility, and clarity compared to the sound obtained from a traditional miking technique. Therefore it acts somewhat as an acoustical "zoom" to the sound source. The included DMM0007 soft rubber holder allows miking of all kinds of instruments, from acoustic guitar to grand piano to drums. This holder allows the mics to be mounted in the best-sounding position using special non-marking adhesive discs. Also included in the kit are two DMM0011-B magnet mounts that can each mount two miniature microphones, two BLM6000 Boundary Layer Mounts, five DUA050 foam windscreens, and two DAD6001-BC adapters with belt clips. These adapters provide power for the DPA miniature microphones via standard 48 volt phantom power.

| IN USE

I initially put the 4061s to work while recording a baby grand at the home of the MuzikMafia's Jerry Navarro. I ran the mics through the Langevin Dual Vocal Combo (which I used to slightly boost the top end). I ended up with a nice piano sound. Since the piano was a baby grand it didn't quite have the bottom end that I'd like, but the imaging was amazing and the DMM0011 Magnet Mounts made mic placement simple and quick. Ultimately, Jerry and I were pleased.

The following week I had the opportunity to use the mics on David Briggs' 9.5-foot Baldwin grand piano while recording artist/pianist Clare Brown at Nashville's House of David studio. Again, I was more than pleased with my results. This time I recorded the mics through the A-Designs Pacifica with no compression or EQ. On another song that didn't require piano, I cranked up the gain, added some compression, and found that the piano created a wonderful stereo drum reverb chamber.

I experimented with the included boundary layer mounts by mounting them inside the completely open piano lid. I found that this pressure zone technique had a bit more midrange and was slightly "nasally" compared to the sound I achieved with the magnetic mounts. This was not what this session called for, but - depending on the song - I could see this being used frequently.

While the SMK4061 Stereo Microphone Kit was designed with acoustic piano in mind, I found that it worked extremely well in a wide variety of situations. I had good results using the microphones along with the DMM0007 Universal Surface Mount to record acoustic guitar. I used one of the included non-marking adhesive discs to attach the mic mount to the body of a Taylor 514-CE acoustic guitar. (In all honesty, I was nervous about using it, but it truly was non-marking.) I experimented with several different placements and found that I ended up with the best results when the microphone capsule was in the center of the sound hole directly behind the strings. I ran the mic through my Pendulum Audio Quartet II with a slight dip at 350 Hz, a slight boost at 15 kHz, a hint of compression, and I ended up with a

FAST FACTS

APPLICATIONS
Studio, broadcast, post production, sound reinforcement

KEY FEATURES
Two DPA 4061-BM omni mics; multiple mounting options; hand selected and sensitivity matched; (also available: SMK4060 mic kit)

PRICE
$995

CONTACT
DPA Microphones | ☎ 303-485-1025
www.dpamicrophones.com


The whole stereo kit (and caboodle).

superb sound that was rich and full without being at all boxy. When needing a stereo sound I used another one of the Universal Surface Mounts to mount the second 4061 on the body closer to the bridge. I ran this mic through my Gordon preamp with no processing and ended up with an amazing sound. Included in the SMK4061 kit is a wide assortment of mounting and windscreen options, which allow the microphone to adapt to virtually any recording or live sound situation.

I had good results using the boundary layer mounts on the floor of my studio to capture drum ambience during a tracking session. Running the mics through a pair of Gordon mic preamps and a pair of Empirical Labs Distressors created a drum sound larger than life.

| SUMMARY

The DPA 4061 is an exceptional microphone and the SMK4061 stereo pair is perfectly suited to record the piano (and practically anything else). If you haven't experienced the high resolution and performance of DPA microphones, you owe it to yourself to give them a listen.

*Over the last decade Russ Long has authored over 100 articles and equipment reviews for **Pro Audio Review**.*

| REVIEW SETUP

Apple Macintosh 2 GHz dual processor G5 w/2 GB RAM; Digidesign Pro Tools 7.1; Lynx Aurora converters; Lucid Gen-X-96 clock; PMC AML-1 monitors.

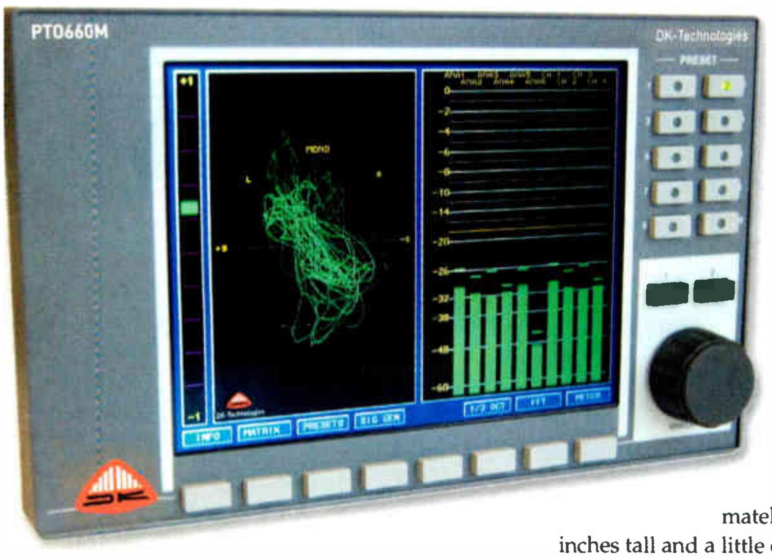
STUDIO | REVIEW

by Stephen Murphy

DK Technologies PT0660M LCD Audio Meter

A near-essential meter for multichannel production, mastering and broadcast.

When Danish manufacturer DK-Audio introduced its first 5.1 surround audio meter in 1998 it was, in effect, a solution looking for a problem. The world of discrete multichannel surround mixing was still a blip on the edge of the radar screen for most mainstream production facilities.



Fast forward eight years and DK's pioneering "JellyFish" multichannel visual display is the de facto standard in surround mixing. With the purchase of the Studio Products Division of ProTelevison Technologies in 2001, the company began video-oriented product development and DK-Audio became DK-Technologies. In addition to its dedicated video sync generators, waveform displays and color analyzers, the video development side of DK has yielded enhancements in its line of LCD audio displays including SDI-HD and SDI-SD input cards for DK's standalone MSD and rackmount PT-series modular meter systems, including the PT0660M reviewed here.

FEATURES

The DK-Technologies PT0660M is a rackmountable stereo and multichannel audio display and measurement device. Like DK's popular standalone MSD600-series meters, the PT0660M features a large, super-bright 640 x 480 LCD panel and a series of eight context-driven soft keys just below the screen. The PT0660M adds a panel of 10 preset buttons and a rotary data entry/output volume knob with two associated option buttons to the front panel. The unit measures approximately 8.5 inches wide, 5.25 inches tall and a little over 3 inches deep.

The "M" in PT0660M stands for modular; to this end, DK offers a wide range of I/O modules so the user can configure the meter to specific needs. The PT0660M can be outfitted with up to four input and four output modules, for a total of 32 input and 16 output channels; a separate module slot is dedicated to utility module that provides a dedicated VGA output for connection to an external monitor plus a serial computer interface, a digital sync input and a DC power input (all on a single 9-pin connector; breakout cable required).

Input options include the Input/1 module (two transformer-balanced analog inputs and one AES3 input), the Input-8A/O (eight analog inputs), the Input-4D/O module (quad AES3 inputs), and the SDI/4 and HDSDI/4 four-channel de-embedding input modules (with reclocked SDI pass-through). The standard output module is the Output/1, which has two analog outputs and one AES3 output. For a complete listing of all I/O options, specifications and restrictions, consult the DK-Technologies website. With the exception of the BNC-equipped SDI modules, specialized 15-pin or 25-pin D-Sub breakout cables are required. Internal sample rate for the analog inputs is 48 kHz. All digital inputs are equipped with sample rate converters for conforming to the internal 48 kHz clock. The sample rate converters can be bypassed when the meter is slaved to an external clock (30 Hz to 100 kHz).

IN USE

The PT0660M meter supplied for review came equipped with four Input/1 modules and a single Output/1 module, for a total of eight analog and eight digital input channels plus two analog and two digital outputs.

The PT0660M provides the full range of metering tools that have made DK's MSD meters the industry standard. In short, these include the tripane "Full-Feature Mode" screen (simultaneous JellyFish/vector oscilloscope, multichannel PPM and phase-correlation meter), a dedicated Peak Program Meter capable of displaying up to 32 channels simultaneously (with seven selectable standards scales), a 1,024-point FFT analyzer, a 1/3-octave spectrum analyzer, a Graphic Loudness Meter and a detailed AES3 bit

FAST FACTS

APPLICATIONS
Studio, broadcast, post

KEY FEATURES
Rackmount stereo and surround master display; modular I/O configuration; variety of I/O modules including analog, AES digital and SDI-SD/HD; JellyFish surround meter; stereo vectorscope, phase correlation meter; 32-channel capable PPM meter; DK-Matrix routing and configuration software

PRICE
starts at $2,995

CONTACT
DK Technologies | ☎ 800-421-0888
www.DK-Technologies.net

stream status display.

In order to concentrate on new features found in the latest MSD software update (V5) and on the PT0660M hardware unit, for rudimentary operation please refer to my earlier review of the MSD-600C-5.1 meter, which can be found in the online archives at www.proaudioreview.com.

On the hardware side, the most significant enhancement found on the PT0660M over other 600 Series models is the addition of 10 instant recall preset buttons and a volume/data entry wheel. While the ability to adjust output gain for a designated audio group (up to eight channels) has been present in earlier versions via the menu system, the addition of the volume "soft knob" provides instant access. This enables the PT0660M to be used as an in-line stereo or surround monitor controller for direct connection to powered monitors or, as I did for this review, a multichannel power amplifier.

One of the things I value most about DK meters is the company's dedication to support and feature enhancement via its operating system software update program. DK has proved exemplary in this regard with past updates and the new version 5 software is no exception. For those interested in potentially purchasing the PT0660M (or other 600-series meters), I strongly suggest downloading the V5 PDF manual from the company's website for complete feature and operational details.

At the core of the meter system is its comprehensive 88 x 96 point (32 x 16 of which are physical inputs and outputs, and the remainder are internal routing points) audio routing matrix. Understanding and using the routing matrix display may appear complex at first but becomes intuitive after a few uses. One of the initial barriers to using the matrix is that, unlike typical audio interface and workstation routing matrices, the meter's matrix is destination-orientated, following the standard found in hardware-based video/audio routers.

Simple point-to-point routing duties can be accomplished in the Compact Matrix screen, and an Extended Matrix screen provides access to a number of other features including channel grouping, channel gain, channel options (e.g., sample rate converter bypass, SDI group settings etc.), and master group gain.

Navigating around the matrix screens using only the soft buttons can be a bit laborious; the addition of the multifunction data/volume knob and option buttons on the PT0660M greatly simplifies navigation and selection (push knob) around the meter in general, and especially in the matrix screens.

Tackling the matrix core and other customizable features of DK meters became even easier with the release of the DK-Matrix Windows application earlier this year. The program presents a more familiar and easy-to-use graphic routing grid, and allows I/O label editing (four characters per channel). Other functions in this useful and most welcome utility include customizing of PPM meter scales and colors, up/downloading and editing of presets, and meter OS updating.

SUMMARY

While it is possible to mix in surround using only the tools provided in a digital workstation, any professional who is seriously involved in multichannel mixing and/or mastering should strongly consider a DK-Technologies meter such as the PT0660M. From its full range of metering and data veracity tools to its integrated surround-capable monitor controller and comprehensive routing matrix, the PT0660M proves to be the best DK audio meter yet and a near-essential for multichannel production, mastering and broadcast facilities.

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.

POST

The latest news and products

NEW PRODUCTS

ALESIS CD Twin

Not normally known for such things, Alesis has launched the CD Twin desktop CD recorder. Not to be confused with rackmounted CD recorders from the Cretaceous period of digital recording, the CD Twin is designed mostly for backup and one-off duplication duties – ideal for post work. The CD Twin uses USB 2.0 as its computer interface. It handles all types of discs (incl. CD-RW) and formats including audio files, video files, photo files and any kind of data/project file. A "One Touch": function eases quick backups.

PRICE: $379.

CONTACT: Alesis | ☎ 401-658-5760 ➲ www.alesis.com

A-DESIGNS Audio ATTY'2D

Need a product with attitude? How about the ATTY'2D from A-Designs Audio? Jokes aside, the ATTY-2D is a multichannel passive line level controller that fits into numerous roles. For the post market it can work as a simple active monitor controller or even to time unruly tracks in mixdowns or post effects. The signal path is purely passive so coloration should not be a problem.

PRICE: $585.

CONTACT: A-Designs Audio | ☎ 818-716-4153 ➲ www.adesignsaudio.com

NEWS

Bob Heil's had an amazing career working with famous acts from The Who and the Grateful Dead through Joe Walsh and Peter Frampton. Now his microphones are being used in big-budget movies. Effects artists for Clint Eastwood's Flags of Our Fathers used **Heil Sound** PR 40 mics to freshly record many of the weapons used in the movie. No sound in a can there! See picture of a Heil PR 40 "miking" a mortar.

And while we're hanging with Hollywood, **JBL** ScreenArray speakers have been installed at the new feature film mix stage at the Pacific Soundwaves post studio in Santa Monica. Also in Hollywood, Elektrofil Digital Studios have added two LSR6328P and nine LSR4326P powered monitor surround systems. Engineer Bruce Sugar has added an LSR4300 surround monitoring system to his studio. His first project was a mix of a live Ringo Starr All Starr Band project. Lastly, Burbank-based Ascent Media Management Services has added several LSR6300 powered monitor systems to its facilities.

by Mel Lambert

AMS Neve 88D Digital Production Console

AMS Neve has been in the digital console business for over 25 years. Its first offering was unveiled to the stunned pro audio industry in 1980, and the firm continues to innovate its range of offerings. But there is one market segment that has eluded not only AMS Neve but its competitors: large-format digital music recording. For reasons too varied to be covered in detail here, the commercial recording industry has been reluctant to move from analog to digital. Sure there are a few studios with heavy silicon investments, but because of familiarly with the analog paradigm and latency issues during tracking and overdub dates - not to mention the advent of powerful DAWs with inboard mixing engines – digital music consoles are thin on the ground.

To meet this challenge AMS Neve recently the 88D Music Production Console. It combines the same basic control surface level as utilized on the 88R analog board, with full assignability and layer control utilized on the firm's range of Capricorn, Logic, DFC, MMC and other digital mixers. So, in a word, the 88D offers analog familiarly with the processing power, assignable control, user flexibility, system integration and full recall we expect from digital architectures. Its modular hardware is fully software configurable and can be expanded to accommodate different projects.

THE SPECS

The 88D combines a 40-bit/floating-point DSP engine providing 1,000 signal paths at a 96 kHz sampling rate, with classic-sounding 1073, 1081 and Air Montserrat mic preamplifiers for acquisition, plus a dedicated 7.1 surround monitor section and Encore Plus mix automation that is fully compatibility with DFC Gemini, 88RS, MMC and Libra systems. And, realizing that if you cannot beat them join them, external DAWs can be accessed and controlled directly from Encore - Pro Tools and Nuendo via HUI and Pyramid via Oasis protocols provide an enhanced mix environment. Multimachine control (MMC) also is included. And for enhanced audio resolution, the recently introduced SHD (Super Hi Definition) I/O modules provide 384 kHz A/D and D/A conversion – ideal choices for high-end music recording and mixing, the 88D's target market.

Of specific note, the 88D includes is a digitally controlled analog monitoring unit that – it can be argued - offers a more familiar sound; either way, the Monitor Facilities Rack offers very flexible stem summing for music for film-style mixing and control of multiple sets of surround loudspeakers. All in all, it's a console with analog sensibilities linked to digital flexibility; I can see its being ideal for music recording and mix down, plus music production and live-to-air.

Usefully, routing buttons and companion displays are offered on each channel strip, in addition to a central position, so that you can work either way. And when you are working in the sweet spot, a bank of assignable faders within the center section make life a lot easier. The basic topology is fixed at four layers of operation, with snapshots of fader Layouts arranged in six banks; each fader can be operated in mono, stereo or surround mode, reducing on-surface clutter.

The 88D comprises five main components: a mixer control surface, a digital processing engine, I/O racks, a monitor rack and a PC to handle system automation. The basic control surface can be supplied with between 24 and 72 faders; the four-layer topology normally comprised two main layers and two sublayers of system inputs. Each channel strip features a conventional motorized fader plus a Logicator that can be set as small fader for second-layer inputs. A flip switch swaps the second input between the main fader. But, as will be obvious, the layout of signal paths is fully customizable, with channels, groups, aux masters and

AMS NEVE Continued On Page 22

AMS NEVE Continued From Page 20

other controls assignable anywhere on the surface. The resultant six user-configurable layouts can be stored and recalled.

The 88D processing system employs the firm's XSP modular DSP cards that handle signal processing, mixing and routing, and connects to one or two analog-digital I/O rack via MADI ports. The system can be configured to accommodate a range of formats, including 24-bit mic and line level converters and AES/EBU I/O. The use of MADI dramatically streamlines connectivity via routers, digital multitracks, etc. During system setup, I/O routing can be configured in any combination of analog and digital audio sources and destinations via a built-in digital patch bay. The digitally controlled monitor rack provides stem summing, control of multiple surround speaker sets and return talkback. Encore Plus uses a standard Windows XP shell for project management, console setup and storage/recall of configuration files and automation data.

While all DSP assignments are, by the nature of the beast, totally soft, the 88D ships with basic configurations that set a four-band EQ, high and low-pass filters plus a DRC compressor/expander for all main input channels. I like the full flexibility that the DSP engine can be reassigned on the fly, without full recompile, meaning that channels and outputs - including groups and aux sends - can be modified from mono thru stereo to surround, dependent upon what assets have been reassigned to the surface and I/O topology.

FLEXIBLE CONTROL SURFACE

The 88D incorporates an extremely user-friendly control surface. Channel functions are easy to locate, with plenty of visual feedback of system status. The layering paradigm allows multiple input channels to be commanded from a compact layout – or from more channels if you need a lot of on-surface controls; it's your choice. The six user-defined desk layouts can each contain up to two main layers and two further sub-layers. In addition to functioning as rotary "small fader" to a second layer, each Logicator – a rotary encoder with integral LED position display within the knob body - also can be assigned to any fader-path control, such as mic gain and aux send.

Each channel strip is divided into an upper and lower section. The upper section handles parameter and routing control via eight touch-sensitive Logicators per fader, arranged as two groups of four assigned to the EQ, filters, dynamics, and other processes plus channel-by-channel gain control of I/O, inserts and aux controls. (Usefully, the LED color changes according to the function being controlled.) The routing array handles assignment of channels to Mains, Aux Busses, Groups and Track outputs, multistem routing during surround mixing. All controls are clearly labeled and easy to locate; after a couple of minutes' use the operational philosophy becomes very obvious, and you pick up speed and agility. (Very usefully, a reverse-routing mode enables the user to determine how the various output busses are being used, and from where.)

> All in all, the 88D Music Production Console offers everything that you'd expect from a manufacturer like AMS Neve: a digital console with an analog sound

The lower fader area contains essential channel and signal path controls plus status displays, ranging from signal present indicators, dynamic metering, cut and solo buttons, pan and a user-definable display strip. Uniquely, the 88D comes equipped with P&G digital faders that feature two scales: a normal level – adjustment scale with 10 dB gain, and a trim scale for creating VCA-style offsets. Companion LEDs indicate the current processing mode being displayed by the channel's bank of Logicators, and also mono or stereo mode plus selection.

Although the 88D's design philosophy enables the operator to access routing and processing controls at the channel strip level – great for setting up I/O assignments methodically as you work your way left-to-right (or right-to-left) across the surface, maybe copying settings across multiple sources – there is no denying that assignabliity from a central position is one of the major appeals for all-digital designs. And here the 88D shines. The AFU – or Assignable Facilities Unit, to use AMS Neve's quaint BritSpeak nomenclature - offers full channel signal processing controls that can be assigned to any I/O signal path. All controls are divided into clearly marked areas, and an alphanumeric display in the middle of the section shows precise control settings and the path name that the section is assigned to. Assignable faders within the center section can be assigned to any signal path, including channels, groups, mains/masters, auxiliaries or track sends/returns. Automation mode switches enable fast access to various options.

SURROUND CONSIDERATIONS

Given the increasing impact of surround technologies, the 88D's multichannel joystick panners and multistem routing system enables the creation of multiple simultaneous mixes, including discrete dialog, music and effects stems, versions for international-language film releases, and in different surround formats - LCRS, Dolby Digital, DTS, Dolby EX, SDDS and IMAX. A separate Joystick Module features a pair of independent controls that enables assigned sources to be moved in a horizontal plane around the assigned multitrack outputs; all dynamic pans and simultaneous object movements are automated to sub-frame accuracy. An optional multimachine transport-control option can command six machines directly via 9-pin ports, and expanded to 33 machines using Synchronet ES/2.

The 88D's output assignments can only be described as comprehensive. As well as conventional output busses, including groups, aux sends and monitors, a total of 48 stem recorder busses are split into six arrays, A thru F, for generating, for example, multiple 5.1-channel dialog, backgrounds, music, hard and supplemental effects submixes. All user controls for setting up stem mixes are easy to develop, and monitoring multiple component M&Es, for example, is a snap.

Talking of the Monitor Matrix, here we see tangible evidence of AMS Neve's concern about using digital technology where it is appropriate – assignability and total automation – but reserving high-resolution analog for the console's 7.1-channel monitor sections, which control in-context playback of three surround stems with independent level adjustment. The section offers routing and control for three sets of independent surround monitor speakers, with level and dim; usefully, a single button controls routing of LCSRs surround elements into the front loudspeakers.

For metering, the 88D is equally well-endowed. In addition to signal-present indicators on each fader that also show dynamics activity, a bank of TFT screens display bargraph meters plus graphic displays of EQ, dynamics, surround pan and AMS Neve's WavTrak - this latter providing visualization of signal status across the entire console. Metering and graphics are provided locally or globally across all channel paths, groups, tracks and mains/masters. The 88D's Ancillary Meter Display offers instant display of all principal signal paths and monitoring points. An operator can access specific paths either independently or linked to the main console monitoring.

DYNAMIC AUTOMATING

As might be expected, the onboard AMS Neve Encore Plus dynamic automation system handles all mix controls, including faders, panners, EQ, dynamics and aux sends. The firm's "Menu-Free Mixing" lets you replay the source materials and just mix; the seamless integration of easy-to-find, touch-sensitive Logicator controls and motorized faders streamlines the creative process. From Flying Faders onwards, AMS Neve has enjoyed an enviable reputation for transparent integration of automation, with a long progeny of analog and digital console topologies. Nice to see that the 88D does not let the team down. User modes include Lock Record - controls replay any existing automation while the transport runs, storing a new version of the mix each time the control is pressed – plus Autoglide and Touch Record, the latter being most useful for making smaller changes and fixes to a mix without overwriting existing automation data.

As I discovered, the 88D is extremely easy to set up and master using Desk Edit and Path Edit applications – maybe an unsurprising conclusion, given the design's previous incarnations and development pedigree. And that is just the beginning. The proprietary Star Command enables automated control of Digidesign Pro Tools and Steinberg Nuendo DAWs (via HUI) plus Merging Technologies Pyramix via Oasis protocol. Under these command protocols, instructions from the 88D's console surface provide fully automated DAW hardware control, with on-surface faders, Logicators and on-surface switches being used to access parameters, including plug-in settings. Neat stuff, indeed.

All in all, the 88D Music Production Console offers everything that you'd expect from a manufacturer like AMS Neve: a digital console with an analog sound. The British Pedigree is self-evident from the hardware build quality, software reliability and sheer "Get-the-job-done-with-minimum-fuss" design philosophy. A console targeted at the heady world of high-end post and music recording, it cannot be denied, but a digital mixing system that represents the very best currently on offer from any company – a rare presence to behold.

My sincere thanks to Ray Gago, AMS Neve Product Manager, for answering my many probing questions during a fascinating demo session at SAE Institute's Los Angeles campus.

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. Now principal of Media&Marketing, a Los Angeles-based consulting service for the professional audio industry, he can be reached at mel.lambert@MEDIAandMARKETING.com; *818-753-9510.*

by **Mike Rivers**

Audio Test and Measuring Equipment - A Survey of What You Might Need

We all need to make measurements on our audio gear at some time or another. It could be as simple as testing for a properly grounded power outlet, checking to see that the sound level at a concert is within the legal limit, or as complex as analyzing a new design or equipment modification. In this overview, I'll look at several tools that cover a wide range of measurements.

WHAT DO YOU NEED TO DO?

For the purpose of this article, I've chosen to group test and measurement gear into some broad categories:

- Utility tools for making quick checks
- Bench or shop equipment for troubleshooting, confirming basic performance characteristics or interface analysis
- Lab or production equipment suitable for design or quality control
- Software that works in conjunction with other hardware, yours or the software vendor's

The golden rule of metrology is that your test equipment must be at least an order of magnitude greater in accuracy and resolution than what you want to measure. If you're checking out a mic preamp that claims 0.001% THD, you need a distortion analyzer that can measure to 0.0001% (that's –120 dB, a formidable task). Accuracy, resolution, and measurement range, as well as the cost associated with those characteristics, are what most often differentiate lab quality equipment from shop grade equipment that makes similar measurements.

Before you choose a piece of test equipment, you must assess your own needs so you don't over or under buy. Do you really need all the accuracy that the manufacturer (presumably) had when he verified the specs? Do you need the capability to supply and measure digital signals as well as analog? Test equipment usually lasts a long time, so as long as it doesn't become obsolete, it's a good investment.

Studio tools such as level and phase meters could be included here, but in the interest of brevity, I'll leave them for another article. So on with the show.

IN THE WORKSHOP AND ON THE ROAD

The Gold Line (formerly Loft) TS1, having been around for 25 years or more, is the great granddaddy of modern audio test sets. It combines a sine wave generator, frequency counter and level meter in a single compact box. While not up to the standards of today's very low distortion gear (the generator is rated at a modest <0.3% THD, though mine clocks in at just over 0.1%) the TS1 is great for routine testing and setup. The level display reads in dB and the reference level can be set with a rear-mounted trim pot. Most will keep it set for 0 dBu (+4 dB on the meter equals +4 dBu) as it comes from the factory. A rackmount version, the TS1RMX packs a bit more output juice than the TS1 (+24 dBu vs. +18 dBu maximum for the tabletop version), sufficient to kick most A/D converters up to full scale.

NTI, the test equipment division of Neutrik (the connector people), offers the MR1 Minirator, ML1 Minilyzer and DL1 Digilyzer, hand-sized Minstruments ideal for general audio testing. The Minirator is a multiple waveform function generator providing low distortion sine wave, square wave, white and pink noise, a polarity test signal, and a stepped sweep over the audio range of 20 Hz to 20 kHz. The Minilyzer measures RMS audio level in dBu, dBV or volts, total harmonic distortion, offers a 1/3 octave spectrum display with numeric readout of the amplitude in each frequency band, an indicator of signal polarity when used with the Minirator's polarity test signal, and an oscilloscope-like waveform display. It also measures the pin 2/3 differential of a balanced output, particularly useful for checking electronically balanced sources. Level measurements can be made either relative to a standard reference (dBu, dBV, or volts) or relative to another measured level (for example, the input source) for convenient gain or frequency response measurement. A combination PPM and VU bar graph display is handy for real time monitoring.

The MR1 and ML1 pair can replace a whole shelf full of traditional test equipment. While they don't measure with sufficient accuracy and resolution to validate specifications of some of today's gear, they're ideal for analyzing problems, showing you why something doesn't sound right. Frequency range is 20 Hz to 20 kHz. The Minirator's maximum output level is +6 dBu which won't quite push most A/D converters to full scale but is useful for general testing. With an optional software package and a calibrated measurement microphone, the ML1 can also make acoustic measurements (SPL, RT60, and speech intelligibility). Alternately, an acoustical measurement version can be purchased as the AL1 Acoustilyzer.

Neutrik Test Instruments AL1

Neutrik Test Instruments DL1 Digilyzer

The NTI Digilyzer makes similar measure-

TESTING Continued On Page 26

TESTING Continued From Page 24

increasingly important in today's working environment. Inputs are AES/EBU, S/PDIF coax and optical, and ADAT optical selectable in channel pairs. A built-in D/A converter provides an audio output so you can hear what you're measuring. Cleverly, the XLR input connector will accept an analog input and pass it to the audio monitor, and let you know that you plugged the wrong cable into the AES/EBU input. A very powerful feature for digital interface troubleshooting is a display of the complete set of channel flag and status bits. The MiniLink option adds a USB port to the NTI analyzers for computer control, uploading of new software, and as a means of storing measurements and transferring stored data to the computer.

The Phonic PAA3 Personal Audio Assistant is similar in form factor to the NTI Minstruments but it's more oriented toward acoustic measurement and analysis. SPL and RT60 measurement, polarity test, and 1/3 octave spectrum analysis can be performed using either the built-in mic or via the line level input. The built-in test generator provides either a 1 kHz sine wave, pink noise, or a polarity test signal. Measurements can be averaged if desired, and stored for later display and analysis. A USB port allows transfer of stored data to a computer.

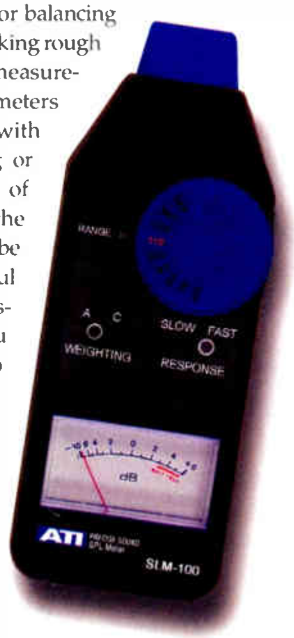
Phonic Pro Audio PAA3 handheld tester

Sencore's DA795 DigiPro Digital Audio Analyzer packs a comprehensive set of digital tests similar to the NTI Digilyzer into a unit that's small enough to take into the field but powerful enough to use on the test bench. Digital I/O includes

Sencore DA795 handheld tester

AES/EBU, S/PDIF optical and coax, and ADAT optical, with dual digital inputs for checking channel synchronization. Tests include level, THD, jitter, latency, channel status and bitstream display, and a unique "transparency" test that measures bit differences between input and output. The DA795 is powered by an internal rechargeable battery, and the signal generator and analyzer functions are controlled by a single turn-and-push knob with menu selection and display on a good sized LCD.

A sound pressure level (SPL) meter is a very basic measuring tool that tells you how loud something is. In addition to keeping your concert sound level legal, the SPL meter is handy for balancing speakers and making rough acoustical measurements. These meters are available with either an analog or digital display, of which I find the analog meter to be the most useful since when measuring SPL, you often have to take an "eyeball average," Inexpensive units such as the Radio Shack model 33-4050 or ATI SLM-100 (analog) or Galaxy CM130 (digital) will keep you out of trouble for under $50. With 50 dB SPL being the minimum value that can be read on these meters, they don't, however, have the sensitivity necessary to measure low-level ambient noise. In order to determine the effectiveness of that new fan that you put in your DAW computer, you'll need a more sensitive meter such as the BK Precision Model 732 ($230) which measures down to 30 dB SPL, or a lab-grade analyzer such as those made by Brüel & Kjær, a different company with similar initials.



ATI SLM-100 SPL meter

A cable tester is a continuity tester that's equipped with common audio connectors so you can simply plug in both ends of a cable, even odd cables such as XLR-to-TRS. The tester indicates continuity of each conductor including the shield, shorts between conductors (which sometimes isn't a fault, for example with a balanced-to-unbalanced cable), and displays wiring errors such as reversed tip and ring. While you can test any cable with a multimeter as long as you know how it's supposed to be wired, a purpose-built cable tester is handy, particularly in the field when you're in a hurry and you suspect that you have a faulty cable.

Top of the heap is probably the Ebtech Swizz Army six-way tester ($180) with connectors for XLR, 1/4-inch, 1/8-inch, RCA, TT (bantam), and MIDI cables. In addition to testing cables, it generates 1 kHz and 440 Hz tones at +4 dBu, -10 dBV, and mic level and it indicates the presence of phantom power. The $30 Behringer CT100 is remarkably close in function to the Swizz Army tester, and for $50, the Inspector from the makers of Cascade Microphones adds Speakon connectors and a pair of test probes for making continuity checks on anything you can reach.

Last but not least, don't underestimate the value of a simple line voltage tester. It will tell you if you have voltage at a socket, and it also displays wiring errors such as a missing ground or hot/neutral reversal that can cause problems ranging from electrical shock hazard to hum. Five dollars at practically any hardware store can help you diagnose ground hum problems and it might even keep you or your clients alive longer.

LAB INSTRUMENTS

Audio Precision has long been the leader in modern audio test systems. Their top of the line 2700 series is found in engineering labs and quality control stations worldwide. The 2700 is a self-contained two-channel signal generator and signal analyzer, available in four basic configurations – analog in and out with analog hardware signal generation and measurement, analog in and out with DSP signal generation and measurement, digital in and out, and the fully dual-domain model with both DSP and analog generation and analysis of analog signals and DSP generation and analysis of digital signals. Other options allow computer control and logging for production QA and long term monitoring, measurement of intermodulation distortion (IM) using alternate standard measurement methods, sine wave burst analysis, tape flutter measurements, and Dolby Digital (AC3) encoding and decoding. Frequency response, THD, noise, spectrum analysis – you name it, the 2700 measures it, with digital I/O up to 192 kHz sample rate.

The new APx585 multichannel analyzer (see review on page 32) is an eight-channel analyzer with similar functions to the 2700. Inputs are analog or digital, with sample rates up to 192 kHz. With applications for this test set leaning more toward the production environment or for field test and setup during installations, Audio Precision has put a major effort into streamlining the process of taking

TESTING Continued On Page 28

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measurements by means of preprogrammed test sequences and computer control and automated logging and report generation.

The Audio Precision Portable One is a self-contained analyzer designed for field measurements or for the lab with limited space and budget. The basic model has analog I/O only,

Audio Precision APx585

but there's a dual domain version adding digital I/O up to 96 kHz. Everything is controlled from the front panel and displayed on a built-in LCD – no computer required. The measurement suite is comprehensive and includes level, noise, THD+Noise, channel balance and crosstalk, wow and flutter, input-to-output phase and SINAD. In the digital domain, measurements include jitter, data integrity, and channel status bits.

While Audio Precision can hardly be accused of building "budget priced" test equipment, the ATS-2 analyzer saves costs by generating and analyzing signals digitally and using high linearity A/D and D/A converters for analog input and output. The ATS-2 connects to a computer through a supplied PCI card or PCMCIA card. All control and display is handled by the computer.

The Prism Sound dScope Series III is a Windows PC application with a dedicated hardware I/O interface that connects to a computer via USB. More than a fancy sound card, the I/O box utilizes internal gain ranging to provide a measurement input range of approximately 140 dB. Certain real-time measurements are performed by onboard DSP in the interface, while control, display, and heavy number-crunching for FFT analysis and multi-tone testing are performed by the PC.

The Prism DSA-1 digital audio signal analyzer appears on the surface to be somewhat similar to the NTI Digilyzer but it's really a whole different instrument. The DSA-1 provides detailed analysis of the digital interface rather than the audio passing through it. It includes two test generators, one to generate standard audio test signals in the digital domain, the other to generate a test signal with known amounts of jitter for analyzing a system's susceptibility to jitter and other cable-based transmission defects. Channel status bits of the test signals are fully programmable, and incoming channel status data can be modified on the fly to verify the response of the receiving device.

Prism dScope 3

TESTING Continued On Page 30


MIDAS

TESTING Continued From Page 28

BASIC TOOLS AND SHOP EQUIPMENT YOU NEED

Nobody in this business should be without a multimeter. The "multi" in multimeter is AC and DC voltage and current and resistance, though most multimeters also provide a beeper for eyes-off continuity testing, and some also offer capacitance, frequency, and temperature measurements. The Fluke model 77 (now discontinued) was the shop standard for over 20 years. Its replacement, the 177, is becoming today's go-to professional digital multimeter. It measures voltage down to 0.1 mV (-78 dBu), frequency up to 100 kHz, and measures capacitance as well as.

There are, of course, less expensive alternatives. Radio Shack offers multimeters for as little as $20 with adequate sensitivity and accuracy to check power supply voltages, resistance, and continuity. Whatever your budget, a multimeter is a tool that you shouldn't be without.

An oscilloscope can be mighty handy for tracking down where a signal has gone astray, seeing at what stage and level clipping occurs, and for looking for oddities such as a stray high frequency oscillation or ringing, or the digital clock riding on an analog signal. There are many inexpensive or free software programs that turn a computer with a sound card into a sort of oscilloscope but I've found that because of latency and bandwidth limited by the sample rate, these are really not very satisfactory for anything other than visualizing a waveform. You don't need a very sophisticated scope for audio work, but if you're inclined to get one, it's worth getting a real one.

Fluke model 177

Even low-end new scopes can cost several hundred dollars. At the risk of having to write a follow-up rticle on how to repair old test equipment, I'll suggest that your best bet is to look at the used market. There are many reputable dealers who offer refurbished and guaranteed Tektronix or Hewlett-Packard scopes for a few hundred dollars. These scopes, originally selling for several thousand dollars, are very well designed and built, and almost never wear out or fail, even after dozens of years. A Tektronix 465 at today's used price is a great buy and a very useful tool in the audio shop.

SOFT WARES

With just about any modern computer having plenty of number crunching horsepower, it's tempting to use it as a signal generator and analyzer. There has been quite a bit of audio test software developed in recent years that utilize a Mac or PC, and a conventional sound card for I/O. When configuring a computer as a test instrument, remember the golden rule – your software-based test equipment will only be as good as the I/O hardware you're using. While there's still room for computer-based test equipment with specialized high quality I/O such as the Prism and Audio Precision products, for routine shop work or making acoustic measurements (which by nature, are fairly low resolution and quite variable) you can get a lot of mileage out of a computer, a respectable sound card, and an application program. This is often a good way to put your last generation's laptop computer to use.

SmAArt Live from SIA Software (a division of LOUD Technologies) is a popular acoustics

AUDIO PRECISION Continued On Page 50


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by **Bascom King**

Using the Audio Precision APx585

AP's latest is their easiest-to-use test system yet

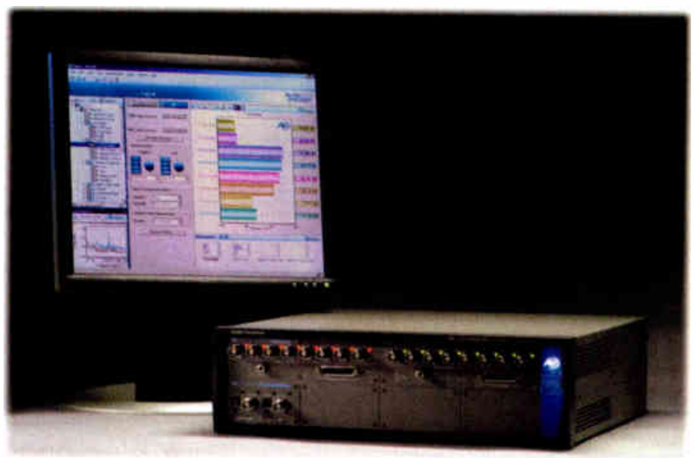
The APx585 is a completely new and different measuring instrument from Audio Precision. To start with, for analog signals, it is an eight-channel unit capable of measuring all eight channels of a unit under test at once. For digital testing, it is a two-channel unit. Notably absent from the front panel are XLR balanced connectors; the basic front panel analog I/O is via unbalanced BNC connectors. Balanced I/O is taken care of via the two DB-25 connectors on the front panel for input and output. Available from Audio Precision in the CAB-585 Cable Kit are accessory breakout cables terminating in eight XLR line connectors for input and output connection to the device under test. Eight channels is a logical number for such an instrument to have: it handles the current surround sound and DVD audio formats and is a nice logical number to measure multichannel studio gear with eight or more channels (which, if it has more than eight, is usually a multiple of eight). [*Audio Precision says that an 8-out /16-in version is in the works - Ed.*]

FEATURES

Aside from these fundamental physical aspects, the software and user interface is completely new and totally different from past Audio Precision machines. The design intent of the 585 was to make the user interface much more intuitive, easier to use, and, as a result, enable much faster testing. This is accomplished through a "Measurement Navigator window" approach that has all the common types of measurements listed. It is a measurement-oriented interface rather than one with an approach of controlling the machine's generator, analyzer, and graphing approach (as was used in prior Audio Precision instruments). In practice, one picks the tests desired by clicking on them and their associated subsets. Each one can be simply set up as to the particular parameters to be used. What is amazing is that one can click on the Run Sequence button at the top of the Measurement Navigator window and the machine does all selected tests in sequence and, if desired, generates a report of the results upon completion. This is far better suited to the production testing of audio gear and is much easier to set up as to limits of pass/fail for the particular tests. Each project is then a complete compilation of all the relevant tests and a setup of them in one project file. In effect, a complete procedure is in a single all-inclusive file — the particular project named and saved file. Note that the measured data itself does not get saved; the idea is to generate and save the report of the measurements. It goes without saying that loading a particular project will repeat all the tests within at some later time, which is a very relevant situation for production testing.

Like the compact Audio Precision ATS-2 instrument, this new machine is all-digital, meaning that the analog inputs are digitized by 24-bit/192 kHz A/D converters and all the output signals are converted to analog with 24-bit/192 kHz D/A converters. All the signal acquisition and generation is done in the digital domain. There are three basic methods of generating the tests: single value using one or two sine waves for stimulus, continuous sweep using a log "chirp" type signal, and stepped frequency or level utilizing a sine wave across a range of frequencies in discreet steps. In all cases, the analysis is done with FFT calculations on the acquired data. Also new in the APx585 is the method of connection to the host or control computer. It is now via a USB interface. Great — no more APIB PC interface card and cable to deal with!

It is not surprising that all this great new capability requires a Herculean powered computer to run it efficiently since processing eight channels takes somewhat more computing power than for two channels. Minimum recommended attributes of the control computer are as follows: a 2 GHz processor or faster, 2 GB of RAM, 2 MB of level 2 Cache, and Windows XP with Microsoft .NET 2.0 Framework (included with the APx500 software). Since I didn't have any machine with quite those capabilities, Audio Precision kindly loaned me a great new IBM Thinkpad laptop with the necessary computing power. As a related difference between the APx585 and prior Audio Precision machines, all of the data processing is in the control computer for the new APx585 with the hardware designated to acquire data only. In contrast, prior machines had a lot of processing done with the unit's machine hardware with much-relaxed requirements for the control computer. For instance, my SYS-2722 is run by a Tiger Direct Sempron 2400 computer with 480 MB of RAM; it runs this machine very nicely and responsively. While not necessarily mentioned by Audio Precision, the demo mode of the APx500 software can be run on a much



FAST FACTS

APPLICATIONS
Test and measurement

KEY FEATURES
Hardware interface; Windows software; 24-bit/192 kHz A/D-D/A; USB port; multiple acquisition modes

PRICE
Starts at $21,000

CONTACT
Audio Precision | ☎ 503-627-0832
www.audioprecision.com

less capable machine. I installed it on my lab computer and it ran just fine. Those interested in exploring the program interface and features should be encouraged to download the software for a try.

IN USE

Turning our attention to the software interface, see **Figure 1** for the basic window layout of the software. This is the default way the program comes up (at least with the high

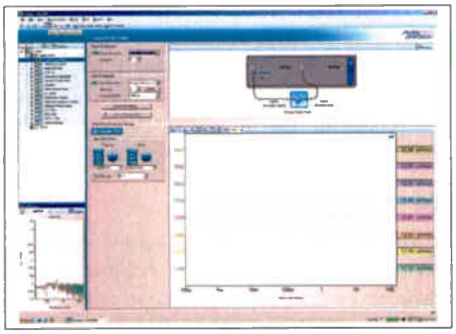


Figure 1

screen resolution of this control laptop). As can be seen, the program window area is divided up into several sections with the program navigation portion being at the upper left. The vertical width of the Measurement Navigator window and the area below it is adjustable; I personally liked the display set with the width maximized. See **Figure 2** for a larger view of the navigation window. Here, some of the tests have had their associated + symbols clicked to open their subsets. The little boxes just to the right of the individual subset test check boxes will show whether or not a test was within set limits or not when run. The signal path setup has been highlighted, resulting in the pictorial diagram in the upper right of the program window showing the particular physical arrangement of input/output selected — in this case, unbalanced analog I/O. Between the Measurement Navigator window and the physical setup path picture are the actual signal path setup variables. The default setup is eight-channel unbalanced I/O. Other choices for the output configuration are analog balanced, digital electrical (BNC connector), digital optical (TOSlink), and none (external). The latter implies input from the DUT (device under test) only. Choices for input configuration are unbalanced, balanced, digital electrical, and digital optical.

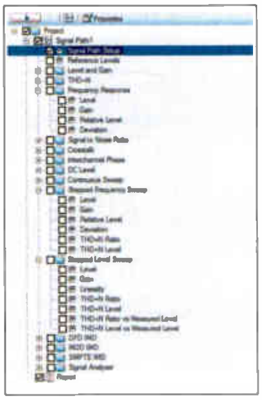


Figure 2

At the bottom of this section are generator controls for generator amplitude, frequency, and the single channel at a time to be driven. The available variables in the section differ according to the particular test highlighted in the navigation window. At the lower left is a window that can be display one of three kinds of view: an oscilloscope for waveforms versus time, a spectral display of amplitude versus frequency, or a set of level meters for all channels. The default spectrum analyzer is shown. Taking up the rest of the area of the program window space is the measurement view window. The results of a particular measurement will show here with the axes and form of presentation being appropriate for the particular

AUDIO PRECISION Continued On Page 34

Audio Precision Continued From Page 33

test run. The default display here assumes an incoming level for the DUT of rms output volts for each of the eight channels. The measurement view can be one of four types: the usual x and y axis plots like amplitude versus frequency, a tabular form of the measured data, horizontal or vertical bar graphs of various things like level versus channel number in horizontal form (or perhaps signal harmonic amplitudes versus harmonic number as vertical bars), and a 3D view that can show a x-y plot of a variable versus channel number like frequency response versus channel number on the z axis.

The loaned control computer was running a version of the newer v1.1 software in beta form, as it has not yet been released [*Editor's note: The v1.1 software is now shipping*]. This version adds support for playback only devices, Dolby/DTS Confidence Testing, and API Programming. After spending several days learning how to run the machine, I made numerous different kinds of measurements with the APx585 on a number of devices including a small digital switching power amp, an old solid state stereo power amplifier, some played back WAV files using Win Amp on the built-in sound card on the control laptop computer, and an eight-channel equalizer.

Figure 3 is a plot of the THD+N ratio versus power output at a 48 kHz sampling frequency for the small switching amp for a 1

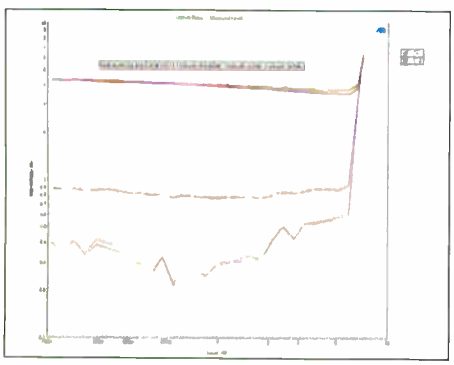


Figure 3

kHz test signal with the 20 kHz measurement filter in (bottom trace) with the full measurement bandwidth of 80+ kHz (middle trace) and for a 5 kHz test frequency with the 80+ kHz measurement bandwidth (top trace). Note that the APx585 doesn't have a data smoothing function (like the earlier Audio Precision instrument software), which would have been nice for the lower curve. The data can be exported to a spreadsheet or other program for such smoothing if necessary.

Figure 4 is a plot of THD+N versus frequency at a power output level of 1W for sampling frequencies of 48 kHz and 96 kHz for the same amplifier. Figures 3 & 4 illustrate

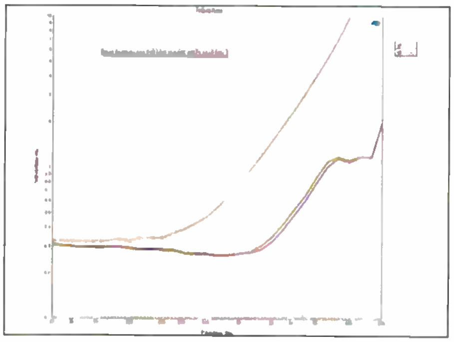


Figure 4

a feature of adding comment text to a measurement graph. It should be noted that this particular little amp is a digital-input-only design with no output digital reconstruction filter as such and has the usual two-pole analog output low-pass filter with a cutoff frequency out of the audio range. These two figures were done using the stepped level and stepped frequency tests, respectively. Going on to the eight-channel equalizer and an eight-channel balanced I/O signal path setup, a different set of EQ curves was set up for each channel and a frequency response was done with the frequency response test mode. A sequence for the checked components was run; the gain versus frequency measurement is shown in **Figure 5**. The ripples in the response curves are artifacts of combining the

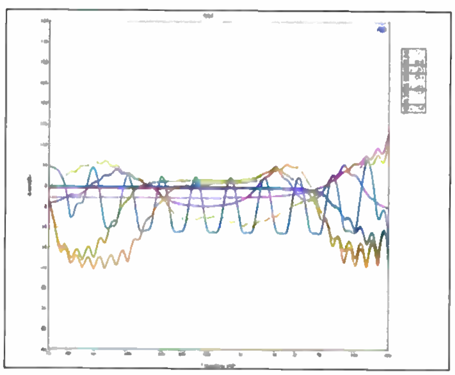


Figure 5

adjacent filter bands in the equalizer. The measurement result reports are exportable in various forms including PDF. A report of this sequence was generated and exported as a PDF file. Page 3 of the report is shown for this measurement in **Figure 6**. The report is first viewable before exporting in the measurement view portion of the program window and is scrollable and steppable through the various pages. The saved page three of five of the report in the figure is typical of the appearance of the other pages of reports.

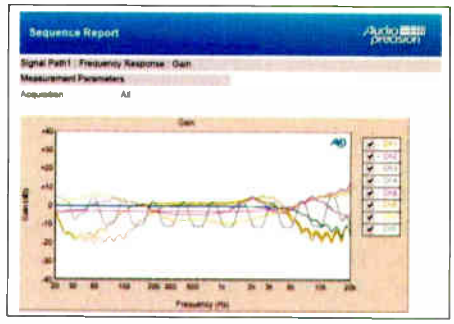


Figure 6

| IN USE: WAV FILES

Measurements from a playback-only device have been greatly enhanced and made easier and more reliable in the APx500 software. Audio Precision has created a series of WAV files of different fixed frequencies, amplitudes and various frequency sweeps at different sample rates and bit densities. Further — and most important — the tests in the playback-only input mode of the APx585 are intelligent in knowing the characteristics of these files such as what the frequencies are and the various timings within, all of which will make these kinds of tests more reliable to run. These test files will be available as test CDs and DVDs for such testing of various devices with the APx585. As mentioned above, the control laptop has a small set of these files loaded into the Win Amp player application. Using the stepped frequency test, a 31-point frequency file was played back and level, relative level, deviation from flat in a 20 Hz to 20 kHz band, THD+N ratio, and THD+N level were all measured as a result of the single pass of the played file. The relative level frequency response of the two channels is plotted in **Figure 7**. With the Measurement Recorder mode, I then measured the amplitude versus

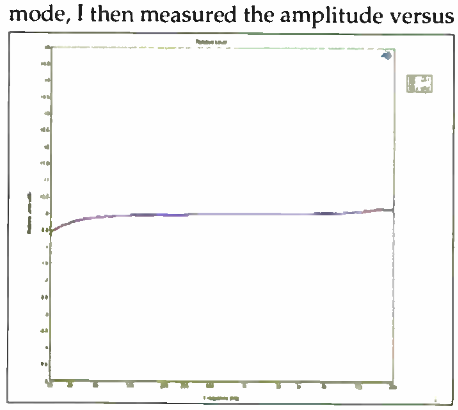


Figure 7

time of a test signal that produced decreasing amplitudes from 0 dBFS down to some very low level in 5 dB steps. The noise level of the sound card in this test environment appears to limit the response to about - 80 dB. This measurement appears in **Figure 8**.

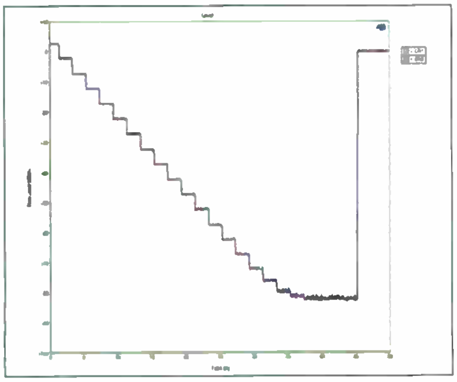


Figure 8

I have a few nits to pick with the machine. First, it does not have any monitor outputs for viewing the measured waveforms and their distortion products on an oscilloscope. Having been taught to always monitor what I was measuring with an oscilloscope (from the first days of my audio learning way back in antiquity by my mentor, Gordon Mercer) and always having done so subsequently as a matter of course, I do miss that. Another thing that is quite a departure from past Audio Precision practice is that the data values are not saved when a project is run and then saved. However, this was a deliberate system design choice with the saved report being the prime means to convey test results. This does make sense, as virtually anyone can read a PDF file. In going through the playback only measurements, I found that the means to measure something versus signal amplitude was not nearly as sophisticated as measuring something versus frequency. These kinds of measurements are done with what is called the Measurement recorder and simply plots the level, THD+N ratio, or THD+N level versus time.

SUMMARY

I could go on and on with more measurements of various IM distortions, crosstalks, phases, and such, but I think I have given some idea of the APx585's capabilities and its program interface. In learning the machine, I quickly grew quite fond of it. If I had lots of extra bucks, I would buy one. While it won't do some things I like and need to do with my SYS-2722, it does a lot, if not most of the things I routinely do in my various testings. Further, it does them in a most expeditious and beautifully graphed manner.

In conclusion, this instrument appears to be very well conceived for its intended purpose. I am sure a lot of them are going to be purchased and put to good use in a great variety of audio testing duties in the very near future.

Bascom King is a consultant to the hi-end audio industry. Recently co-designed with Arnie Nudell most of the new speakers in the Genesis Advanced Technologies line, www.genesisloudspeakers.com. *In addition to measuring digital devices and reviewing test gear for Pro Audio Review, he has been measuring preamps and power amplifiers for the online magazine SoundStage!, writing equipment reviews for The Audiophile Voice, and evaluationg and testing switching power amplifiers for various manufacturers.*

LIVE

The latest news and products

NEW PRODUCTS

CARVIN UX1000 Wireless System

Carvin's latest foray into wireless microphone systems is the UX1000. The UX1000 is a UHF system with a choice of 960 channels. The receiver is a PLL diversity unit with low battery and activity indicators. Lav, headset mic or instrument cable are available for the beltpack systems. The UX1000 is available in handheld and beltpack transmitter packages.

PRICES: $399 (beltpack) and $429 (handheld).

CONTACT: Carvin | ☎ 858-487-1600 ➲ www.carvin.com

WORXAUDIO TECHNOLOGIES V8-PMD1 Powered Touring Line Array

The V8-PMD1 from WorxAudio Technologies packs a lot into one package. The business end of the V8-PMD1 is a pair of 8-inch woofers and a 3-inch compression driver. The woofers are coupled to an Acoustic Intergrading Module to reduce cone filtering. The high-frequency driver is coupled to a FlatWave Former wave-shaping device. Other features include a digital power amp, Baltic birch cabinet with polyurethane finish, powder-coated 14-gauge steel grille and TrueAim rigging.

PRICE: $6,822.

CONTACT: WorxAudio Technologies | ☎ 336-275-7474 ➲ www.worxaudio.com

RADIAL ENGINEERING JX44 Air Control

Not everyone is going to need Radial Engineering's new JX44 Air Control but for those in need of keeping control over multiple instruments on stage the JX44 might be a handy tool. The JX44 offers four input channels and four output channels. However, the inputs and outputs are optimized to handle certain duties such as guitar input or stereo output. Inside electronics are Class A, transformer-isolated with a Radial DI box built in. For emergencies there is even a panic button for overrides.

PRICE: $1,000.

CONTACT: Radial Engineering | ☎ 604-942-1001 ➲ www.radialeng.com

DIGITECH RP250 Guitar Processors

We've come a long way since the Wah-Wah pedal! Digitech's RP250 guitar effects processor/pedal is the latest in multitalented guitar pedals. Sporting 89 models – 54 effects, 21 amp/preamps and 14 speaker cabinets – the RP250 can make a guitar do a lot that it doesn't naturally do. Many of the onboard models will save you from buying those cranky, screechy, broken-down pedals off of eBay in the vain hope of getting "that" sound from 30 years ago. Bonus features include USB connectivity with Mac/Windows editing software, 50 user presets, built-in drum machine and a chromatic tuner.

PRICE: $229.

CONTACT: Digitech | ☎ 801-566-8800 ➲ www.digitech.com

NEWS

KT Tunstall, Alice In Chains and Joan Jett have all been using **Shure** mics at recent concerts. Alice In Chains is using KSM9 wired mics, UHF-R and ULX wireless systems. Joan Jett has been crooning into a Beta 58A. KT Tunstall has been using an SM58 and if you could see under her hair in the picture above you'd see her Shure PSM 700 in-ear monitor system.

The James Gang used **WorxAudio** Max 1.5M and 2.5M stage monitors and 3.5A custom speakers on their recent series of comeback concerts.

Spanning the globe, **Allen & Heath** has provided equipment on several continents recently. Down Under, the Platinum night club on Queensland's Gold Coast has installed four Xone:92 DJ mixers. Crossing the Equator, A7H ML4000 and ML5000 consoles were used at Taiwan's biggest rock festivals and we're not talking geology – the Gung-Liau Ho-Hai-Yan Festival. In Ukraine, the MyTime youth festival used ML5000, GL4000 and GL2200 mixers. A bit closer to Allen & heath's home in the UK, the London School of Sound has added Xone:3D DJ mixers to its DJ Skills course (shouldn't that be Skillz?).

The guitarists backing Ricky Martin on his *Life* tour are using **Lectrosonics** IS400 wireless beltpack systems. See picture of guitarist Dave Cabrera and monitor engineer Raphael Alkins with their Lectro gear.


VCA Power
For The People
Great news for installations and tours on a budget. The new Soundcraft MH2 delivers the critical mixing features of the acclaimed MH Series in a more affordable, smaller footprint, fixed-frame format.
Dual-format FOH/Monitor configuration, 8 Groups or 10 Auxes on fader operation and legendary British EQ – it's all there, along with the most sophisticated VCA grouping of any mixer in its class.
Like all Soundcraft live sound consoles, dependability comes as standard. The MH2 uses an integrated switched mode power supply which can be removed for easy maintenance, with a rear-panel connector provided to fit an external power supply for complete dual-redundancy assurance. And with four frame sizes available, there's sure to be an MH2 that fits your application.
To find out more about the Soundcraft MH2 dual-format live sound consoles, contact Soundcraft or visit the Soundcraft website.
MH
2
REVIEWER'S PICK
Soundcraft
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MIDAS XL8 DIGITAL LIVE PERFORMANCE SYSTEM

by **Dan Wothke**

I recently received the call to visit Westover Church in Greensboro, North Carolina and get a first hand look at the first-ever, fully-digital mixing system from Midas, so I immediately loaded up the iPod and headed east. The Midas XL8 is not a simple dip into the digital pool to check the water; it is the result of a three-and-a-half year project blending Midas' renowned live sound history and pedigree with the latest advancements in digital audio technology. It is a complete system designed to take signal from the source, convert it, split it, route it, and process it without ever having to use one single piece of third-party equipment.

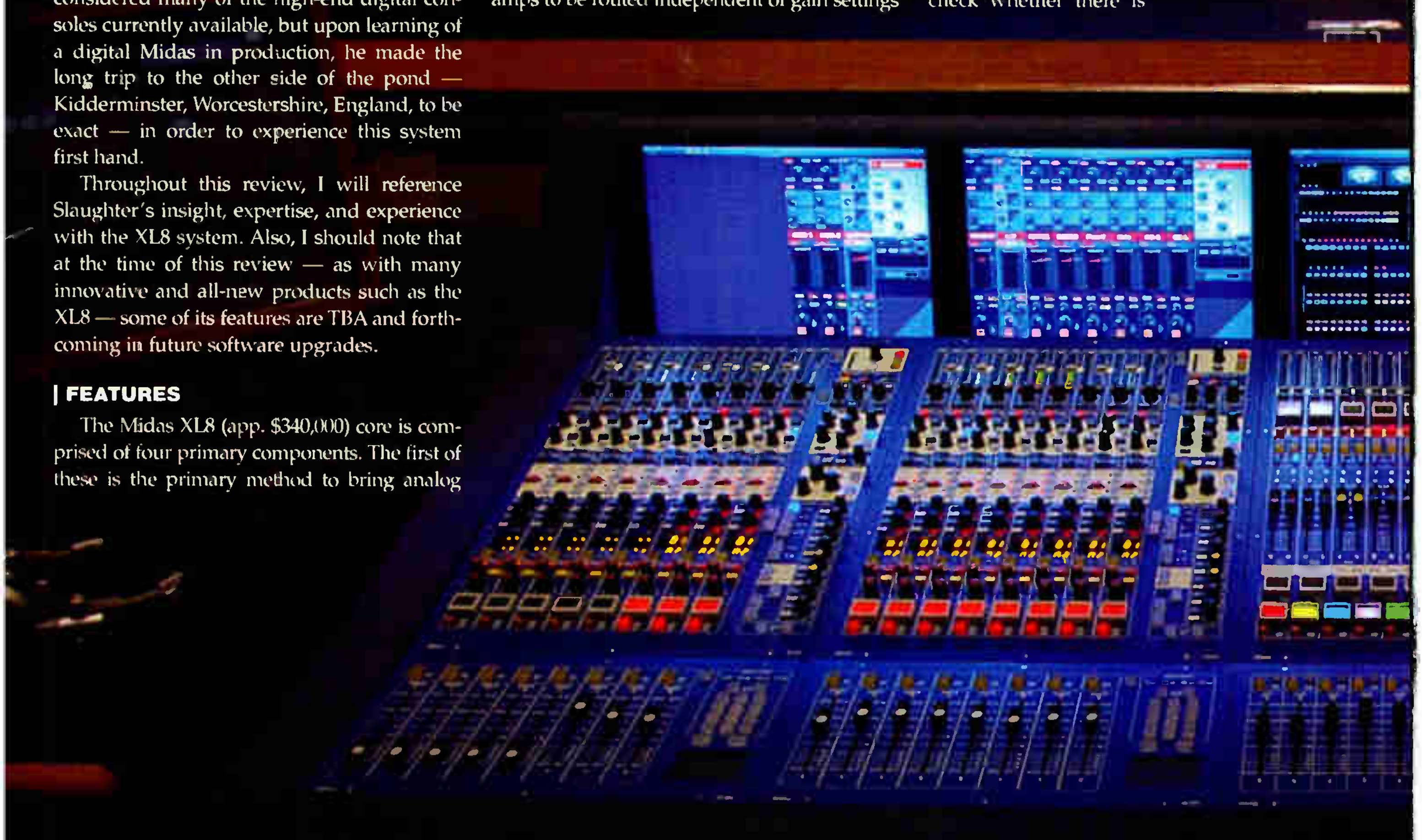
Westover Church's XL8 is the first installed anywhere in the world (serial number 1001). Westover Technical Director Danny Slaughter considered many of the high-end digital consoles currently available, but upon learning of a digital Midas in production, he made the long trip to the other side of the pond — Kidderminster, Worcestershire, England, to be exact — in order to experience this system first hand.

Throughout this review, I will reference Slaughter's insight, expertise, and experience with the XL8 system. Also, I should note that at the time of this review — as with many innovative and all-new products such as the XL8 — some of its features are TBA and forthcoming in future software upgrades.

| FEATURES

The Midas XL8 (app. $340,000) core is comprised of four primary components. The first of these is the primary method to bring analog audio into the system via one of the four 24-input, 96kHz shared mic/line splitters — model number DL431 — totaling 96 available inputs. All inputs and outputs are interfaced with the system via XLR connectors. Each DL431 requires six rack spaces to house the 24 inputs and 96 analog outputs to interface with other analog systems. There are three independent preamps supplied for every input; Preamps 1 and 2 (with analog outputs located on the back of the unit next to the inputs) have separate local and remote gain controls while the third (located on the front – very handy for quick access to the split outputs) is the broadcast transformer-isolated input with a fixed gain. The two preamps with independent gain control also have their own set of ADCs, allowing the two preamps to be routed independent of gain settings and can then be routed digitally via AES50.

The DL431 is a well-conceived, full digital/analog splitter with local metering, power supply indicator lights on the front panel, and an indicator light next to each XLR input. This corresponds to the Check button on the XL8 controller which, when pressed, will automatically light up the corresponding LED located next to the XLR input on the splitter. This is extremely helpful when troubleshooting if an input is not working by having someone at the splitter while the engineer presses the Check button to verify that the cable is plugged into the correct input. If it is, then the engineer can immediately check whether there is



an internal routing issue. With 96 possible inputs, this is a great feature to help expedite troubleshooting any signal issues.

The DL431 is also equipped with a local headphone jack that allows a user to monitor any input chosen on the front panel. A small LCD screen provides local configuration of the DL431 including phantom power, adjust local preamp level reflected by the onboard metering and enable the built-in 30Hz filter. Also on the back of the unit are Ethernet ports for standalone remote control and USB connections for tunneling third-party data; each DL431 is equipped with dual integral power supplies.

Yes, the DL431 will serve any show well for interfacing the system with the stage, but what about the need for interfacing analog or digital gear with the system from an external source? Saloon doors fly open — in walks the five DL451s.

This is the XL8's modular I/O in a 3U rackmount unit. The DL451 can support up to 24 I/O per unit and can be configured in combinations of eight XLR panels in the rear of the unit. Five DL451s are shipped with the system to be placed at different locations, at the same location with different configurations, or a combination of both. Analog mic/line In, Out, and AES/EBU are the three configurations available. The only cable needed to interface the DL451 with the system is a CAT-5/6 cable sent to the router. Multiple DL451s can be used with the system. In this scenario, Westover purchased additional units to use throughout their facility to allow for flexibility and future expansion.

Once analog audio is converted to digital, it is sent to the router (DL461) via AES50 on CAT-5 cable. The DL461 is a 3U rackmountable unit and is the sole link between console, routers, processors, and the control surface and acts as the AES50 traffic cop. All delay compensation for output timing and phase coherency is handled from within the unit. Two of these are shipped with the XL8 configuration and — like many of the XL8's well-thought-out design behaviors — the DL461s are setup as a fully duplicated network for redundancy. A quick glance at the series of LEDs — logically laid out on the front — indicate the system's current status. The rear of the unit consists of a series of AES50 connectors; Ethernet and USB for third parties; and BNC and AES3 word clock interface for external synchronization.

Handling all DSP for the XL8 system is a bank of 10 1U units: the DL471. Nine of the processors are actively in use, while the tenth is on standby and ready to jump in should one of the nine fail. LEDs are on the front of the panel to indicate status and a small screen and quad buttons are available for system diagnostics and configuration. The DL471 rear panel is equipped with AES50 and Ethernet connections.

Let's move on to the control surface. High-end digital systems can all route signal virtually everywhere and anywhere, have built-in DSP, can handle more inputs than the footprint reflects, and so on. But in actuality, the key factor is how well they perform these features, the ease of navigation, and — of course — how the system sounds.

The first element of adapting to a digital console is the virtual knob, where a knob's function changes based on the page or function that is currently chosen. Midas stays close to its analog roots by avoiding this problematic design.

To facilitate the one-knob-per-function design, the work surface is separated into bays with each bay having set areas referred to as "zones." The five main bays are broken into three-input bays — one mix bay and one output bay. Looking at the console from left to right, the bays are laid out as follows: input, input, mix, output, and input, allowing for a total of 24 inputs available. These inputs do not have to be consecutive, but do have to be laid out within standard banks of eight. The control surface itself measures just a hair over 68 inches x 54 inches and offers plenty of knobs and buttons to make even the most adamant analog engineer at least partially happy — the board has a great analog feel. But make no mistake; the board is absolutely digital — the five well-lit, daylight-visible DVI TFT display screens emphasize this fact.

The input bay has a Fast Zone that includes the most common "must-have" controls available to the engineer. The controls are laid out in a logical order and include monitoring input level, gain reduction and gate monitoring, and a link button for stereo inputs. Moving down the Fast Zone is the gain knob, phase, and phantom power followed by the direct output assignments and mute option for the direct output. The Dynamics section is next with the basic control of threshold for compressor and gate as well as an enable button and listen monitoring. (More advanced control for the DSP is explained later, highlighting the Input Channel Strip, as is the need for external inserts, which is dealt with by offering an insert button and corresponding LED indicator.)

EQ LED indicators follow, each of the four bands; treble, hi-mid, lo-mid and bass accompanied by a red light to allow with a quick glance to see what EQ is currently engaged, and a Master EQ On button. A pair of auxiliary knobs — which can be quickly scrolled through to adjust gain, engage the aux, or assign the pair post/pre-

MIDAS Continued On Page 40

A Standard XL8 System

- One XL8 Control Centre
- Ten DL471 DSP Engines
- Two DL461 Audio System Signal Routers
- Five DL451 Audio System Modular I/O
- Four DL431 Audio System Input Splitters
- One DN9331 Klark Teknik Helix Rapide

A Typical Configuration

- 16 Mic/line auxiliary inputs (giving a total of 112 mic inputs as standard)
- 32 aux/group buss outputs
- 16 matrix (main) outputs
- 1 stereo main output
- 1 mono main output
- 2 stereo local monitor outputs

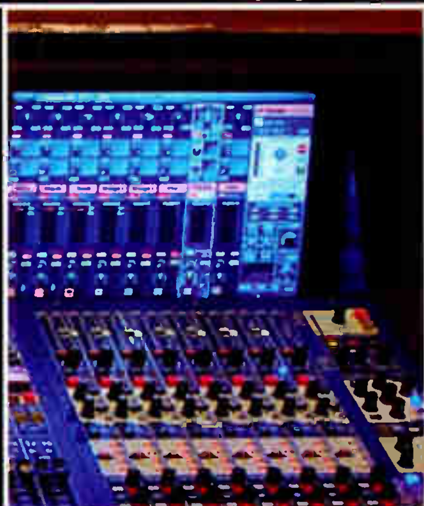
FAST FACTS

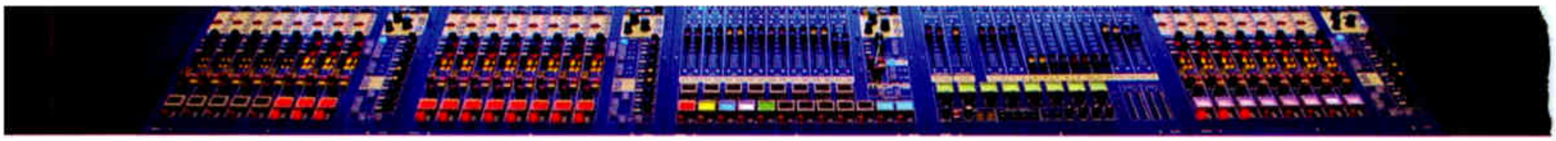
APPLICATIONS
Live sound, sound reinforcement, installation

KEY FEATURES
Modular system; separate I/O box; up to 96 mic inputs; 96 kHz sample rate; Klark Teknik Rapide EQ; onboard DSP; delay; Ethernet; USB; MidasNet

PRICE
app. $340,000

CONTACT
Midas/Telex | ☎ 952-887-7445
www.midasconsoles.com





MIDAS Continued From Page 39

fader — are next in the chain with alphanumeric indication of which pair is currently selected. The section is also accompanied by AFL monitoring and a store button. Next is the mix Buss B section with the standard gain, Stereo/Mono/SIS assignment, and rotary knob level control.

One of the main features that make the board so easy to navigate is the ability to name and color code all of the inputs in the computer for quick identification: basically, this is a next-generation scribble strip. These LED backlit buttons are followed by the red backlit mute button and indicator LEDs for mute and auto safe lights. For engineers who still need a good Sharpie and board tape to jot down notes, there is still adequate space for that before reaching the fader section, which houses the Solo B, main solo, and the newly ergonomically-designed, conductive plastic touch-sensitive moving faders.

For detailed control, each of the three input bays has one Input Channel Strip located at the right of the bay. Here is where the engineer can perform tweaks to the onboard processing. This section includes full control of the direct outputs with level, solo, mute, and pre-assignment. The safety section is a series of switches for EQ, Automation, Mute, Dynamics, Mic, and Fader that will allow any one or combination of these sections to be put into safe mode. When an engineer dials in the perfect EQ and wants to be sure it does not accidentally get tweaked, he should simply depress the safe button for the EQ section.

A series of LPFs and HPFs are next with a controllable frequency and slope. Remember the feature I noted earlier on the DL431 (A/D input section), where the engineer can press a button on the board to light an LED on the splitter to verify that the input is plugged into the correct XLR jack? That button is located right next to the set of filters and the gain control for the main input at the A/D. Also available is a fixed 30 Hz filter, which is directly linked to the input to the splitter. A changeover switch to swap the dynamic/EQ sections follows.

Now let's delve into the meat and potatoes of those sections. In accordance with the one-knob-per-function design of the XL8, the dynamics and EQ sections have fixed knobs in this area, allowing for full control of their respective DSP functions. The ability to side chain from any other input on the board is available, which allows for a de-esser or some other type of side chain compression and gating. Parametric EQing follows with a set of knobs shared between the different frequencies and well-lit LEDs to assist the engineer in identifying what frequency is being modified. One of the four bands can be modified at a time and each band has the ability for different curves and/or shelving EQ.

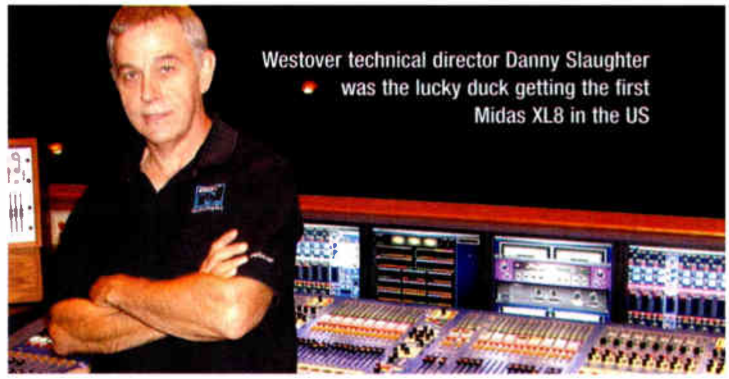
Each XL8 system is also capable of 31-band graphical EQ via a hardware controller, the Klark Teknik Rapide. At the time of the review, the Rapide software was not yet functioning (but should be by the time you read this), but the ability to control the graphic EQ with the use of the mouse was. The auxiliary section follows with four pair of color-coded knobs, each with a designated On and Pre button. Scrolling through banks of eight is accomplished by choosing the Scroll Through 8 function, or by choosing any of the designated bank of eight buttons including any of the 32 auxiliary sends or the 16 available Matrix sends.

Wrapping up the Input Channel Strip is a Scroll By One function to easily call up the next or previous module. The screen located at the top of the bay visually represents anything selected within the bay and does a good job of representing the entire module overview. With one glance, the engineer can see the full layout of the module with well lit color-coded representation of levels, metering, switches, and DSP. Each Input Bay has an Ext Screen selector to bring up an external video source and an onboard mouse pad, which can be used to control the entire bay, if needed. The input pad selection resembles a calculator and allows the engineer to bring up any input to that module by entering the number and pressing Enter. These are always brought up in banks of eight. For instance, if the engineer enters 52, the bank then shows inputs 49 through 56. There is no option to have different inputs show up in the same module using this method, but that is easily remedied in the next bay. Banks can also be locked into a bay so the most critical inputs are always available.

The Mix Bay is the next section in line and takes the worry of technology out of the picture completely. Due to the intuitive design and layout of the XL8, the Mix Bay allows the engineer to focus on the mix. This all starts with the main screen, which — like the other screens — will reflect what is selected in the bay or can be switched to an external source. The home base for this bay is the master status screen. This screen shows every channel's level, gate, dynamics, solo, and mute for quick visual indication of the entire board.

By holding the pointer over any of the channels, a small window pops up with the name of the channel and the input number. One feature that would be handy is the ability to then click on that meter and have that channel appear in one of the input bays. As it stands now, the engineer would have to punch in that number or bring it up in one of the groups and modify from there. Having a "selection-by-click" feature would speed up that process.

The Auxiliary master faders take up the top third of the Output bay, each with their dedicated talkback button and dynamics and EQ available. These also have the expected mute and solo options, safe modes and the well-lit, color-coded scribble strip listing the names of the channels. The flexibility of the XL8 is realized in the Output section via the 12 VCA



Westover technical director Danny Slaughter was the lucky duck getting the first Midas XL8 in the US

(Variable Control Association) and the POP (Population) groups. Each of the 12 VCA groups can be customized by name and color. When a VCA master fader is selected, all of the channels assigned to that group will appear in a bank and reflect the color-coding of the VCA master. For example, if group one consisted of drums on 1 - 10, percussion on 22 - 24, and timpani on 49 - 53, these would all show up in chronological order on the modules to the left of the VCA when that master was selected. The eight POP groups work the same way, except that they do not have a master fader associated with each of them, so when the POP button is selected, all associated inputs appear for ease of individual editing and/or mixing. The two combined allow for up to 20 groups, which is more than enough for even the most intricate shows. A channel can be assigned to multiple groups and the color of that channel will reflect whatever group is currently active.

The output bay houses the L/R/M and 16 matrix output faders, and — as with every output — each have their own dynamic and EQ (graphic and parametric) assignment switches. Eight user-assignable switches are available, but at the time of this review, they have not yet been implemented into the software. I predict these will be great assets for quick modifications to onboard effects, which

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are currently handled by the onboard trackball.

In addition to the pink noise and 1kHz tone generator is a sweepable 50 Hz to 5 kHz tone generator, which can be routed to outputs of the console. Two USB ports are available — a must have for backing up critical data such as shows, presets, or automation. The console has a comprehensive onboard talkback system with variable gain and limiting available for the onboard or external microphone. Closefield monitoring is available via one of the two monitor feeds: Monitor A, which is controlled via fader, and Monitor B, controlled via rotary knob. Solo and the multitude of routing options are housed in this section, all with well-lit dedicated knobs/switches. The main base for controlling the Output and Mix bays is located at the base of the console and houses two custom trackball mice and a series of quick navigation buttons to pull up common views for the main screen and a roll-out keyboard underneath. This screen has a built in three-way KVM switch.

IN USE

The XL8's great layout and navigation is not worth the paper this review is printed on without redundancy. Redundancy is absolutely critical for digital systems and Midas spared no expense in taking every precaution necessary.

For starters, the XL8 control surface has separate power inlets for each bay with their own PowerCon connector - allowing the engineer to get power for the board from two different sources. Each bay is equipped with a dedicated power supply and switch, and the design of the system allows the engineer to take over any bay from any other bay for full control should a bay ever fail. The system DSP is virtually unlimited and is powered by a dual, ultrahigh-speed, contra-rotating data loop for direct processor-to-processor communications. This aggressive design is to allow for headroom in processing and redundancy. Each of the units are linked to the one before, and — after it is in the chain and if any unit should fail — the processing load is immediately handed off to the next processor in line, allowing for a continuous data loop. All of the components incorporate fault detection to immediately notify the user if an error is detected which allows the user to decide whether to switch to the backup system components. Support for the Midas XL8 is 24/7 worldwide; the factory and all offices support the equipment, regardless of where it was purchased.

For me, the highlight of the onboard DSP was the compressors. There are four flavors

ON MidasNET

The XL8 system backbone of data is referred to as MidasNET. MidasNET incorporates digital audio using the AES50 protocol, control data, and standard Ethernet traffic bidirectionally via a single CAT-5 for local (24-channel) connections and a single CAT-6 (or fiber-optic) for a digital snake equal to a 384-channel analog multicore. This enables the XL8 system hardware to interface on RJ45 connectors.

In the background, the system is monitoring data, temperature, or anything that could cause system failure, preemptively communicating with the user prior to any audible problems. The system does all of this with a 96 kHz sample rate and a latency on each network link of only 70uS, with total system latencies at 2mS. Management of all delay points is provided including compensation for any outboard DL451 modular system used for external sources such as insert points. All network connections are duplicated for system-wide dual-redundancy.

of compressors with RMS, Peak, Limiter and Vintage compressor algorithms. Each compressor sports a sleek layout on screen and offers a full palette of sonic characteristics.

Equalization was equally as impressive, with two of the four-band parametrics offering three different shelving types and very

PRODUCTPOINTS

- Complete digital system
- Infinite flexibility combined with ease of use
- Comprehensive redundancy
- Full scene automation
- 24/7 worldwide support

- Price
- Lack of onboard DSP libraries

SCORE

A well thought out modular digital console that has enough features and inputs to fit most jobs.

detailed, accurate control for precise equalization, yet it was still exceptionally easy to dial in the right sound. The graphic EQ is based on proven Klark Teknik parameters. Sixteen different stereo effects are available for the system including reverb (based on the KT DN780), EQ, auto-pan, and variations of delay. The major (but reportedly temporary) downside with all of the DSP is the lack of ability to save presets to a library and assign them to other channels automatically; this is slated for a future software upgrade.

Also, with the entire system built on Linux, it is limited to incorporate any third-party plug-ins because most companies do not have their plug-ins written for the Linux OS. However, I must say that the trade-off for stability is well worth it. It is easy enough to bring in outside DSP processing via AES or analog and the KVM switch allows the XL8 screen, mouse and keyboard to be used to control the third party product. With all of the onboard DSP being proprietary, the need to tie in external processing is likely.

Where this console really shines is when the use of the scenes and automation is fully utilized. Virtually every control can be programmable per act/per scene and only the active controls for that preset are available. For instance, if there are six inputs used for a particular preset, that is all that will be available during the time that preset is active.

For further insight on the XL8 presets in use, I spoke with Wallace Flores, Associate Sound Designer for the "Sister Act" show on Broadway. Flores has been working on the production team and has first-hand experience with the XL8 in a professional theatre environment. His assessment of the presets was that

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ON The Facility

Together for Westover's 3000-seat sanctuary — a theater-like room with traditional church appeal — Audio/Acoustical Consultant Armando Fullwood of Design 2020, Integrator Tim Owens of Audio Ethics, and Danny Slaughter chose to include nearly all Telex-oriented components. The Electro-Voice 108-enclosure/70-amplifier distributed delay system comprised the majority of Westover's product list. This PA features EV X-Array 1183 and 1123 enclosures and EV Xsubs for house mains alongside an all-EV selection of front-fill, balcony, surround, and wedge enclosures, the total of which are fed by EV RL Precision amplifiers. On stage, an Aviom personal monitoring system provides all monitor mixes for Aviom and Shure 600 IEMs.

by **Strother Bullins**

On With The Show

A good friend of mine recently invited me to accompany him to hear Buckcherry and Crossfade, touring mates that — strangely enough — I just so happened to cover for *PAR* within the last six months (in 'Single Slice' and here in 'X/Audio,' respectively). Seeing that I happen to really enjoy Buckcherry, and that Crossfade are officially "local heroes" (being a platinum-selling act from nearby Columbia, South Carolina), and that I had heard that this particular venue recently underwent major (and long overdue) renovations, I said yes and the two of us hit downtown Charlotte for the Tuesday night, $26-a-head rock-club show.

The place (which shall remain nameless to save everyone grief) looked great. I was impressed with the cosmetic updates throughout and both my friend and I agreed that the audience "pit" was more than twice the size that it used to be. This particular venue has been around for quite some time and has had its share of 750 to 1,000-capacity national tours, so it was nice to see its owners and management re-investing for the future and, hopefully, bigger and better events with more in attendance.

In summation, the show was a blast, musicians played their hearts out, and the big Tuesday crowd went home satisfied, sweaty, and hoarse with a significant smattering of official tour merch in hand. In other words, mission accomplished.

However, few in attendance would leave with the realization that the MVPs of the night weren't band members, club management, or even bartenders. As usual, in venues and on tours such as this, the MVPs were most definitely the audio guys. Stephen Shaw — front-of-house (and monitors) engineer for Buckcherry — once again squeezed every bit of fidelity, musicality, and power from a sub-standard house PA with just his ears, his knowledge, a splitter snake, and his trusty microphone package to count on.

AT THE MERCY OF THE HOUSE

"I use house gear and house processing wherever I go," explains Shaw of his current gig with just-gone-Gold Buckcherry, in which he also serves as Tour/Production Manager and Accountant. "I'm at the mercy of the house every night."

And in Charlotte, the venue was seemingly merciless. "I walked in and thought, 'Oh, God!' Two of the EAW mains were hung at a 45-degree angle down, facing in and just off the stage without separate processors. Then I looked at the boxes, which weren't all the same; the outside ones were the 850E Series and the inside ones were 850 Series. They had the 850 subs; mind you, they weren't the SB1000 and were 15 feet back from the hang with no time delay. I had to hit the subs really hard to catch up with the top boxes. So yeah, I was just shaking my head."

For Shaw, each day is a completely new challenge. "After about 21 years of doing this, I've mixed on pretty much everything out there from the crappiest to the best," he explains. "In venues like the one last night — one that's kind of new — they probably got a deal on the six different EAW boxes with mismatched horns. There was one processor control for everything and none of the boxes matched. So, in addition to cutting certain things and throwing vocals out of phase, it was an EQ challenge. When you play a place like last night — what I call a 'roadhouse' — you get what was hip 10 years ago. Then, the very next night, you may play House of Blues where there's a brand new EV line array, or maybe an outdoor gig where a big audio company will bring in a new [JBL] Vertec or [Meyer Sound] MILO rig. Working on a range of gear like that keeps your chops up."

On tour with Buckcherry, a happy Stephen Shaw finds himself with a Midas XL200-fronted rig.

Having one constant — a complete microphone package by Audix — is a nice starting point, insists Shaw. Night after night, the high decibel/high energy sounds of Buckcherry are captured via Audix OM-5 models on vocals; D6, D2, D4, i5, and ADX51 models (with a little help from a Yamaha Sub-kick) on drums; and D3 models on both Buckcherry guitarists' amps.

Photo credit: Doug Rice/Allure

"It provides incredible consistency," summates Shaw of his traveling transducer collection. "With the same microphones night after night, there's minimal tweaking for the in-ear console. It's nice to have my own Whirlwind splitter snake, too, with all the ground lifts on it. Maybe some day I'll have a monitor engineer, but... one day at a time!"

ON BEING SIMPLY PRAGMATIC

In talking with Shaw, it is clear that not a single challenge or obstacle of touring sans rig seems to really bother him. Buckcherry, he explains, along with many others he has mixed in the past, is "simply pragmatic. They have the same attitude as me on this. Their new album just went gold and the band is making a living. They're not ready to take on a truck and extra production. They know that they're at the mercy of what they're handed. They are pros and troupers and do their jobs regardless."

But still, when asked if he has any advice for clubs that are installing new house PA systems, Shaw has more than a few thoughts on the subject. (Club owners: take note.)

"Don't buy something just because it looks big and bad," Shaw begins. "It may not be big and bad, just bad. Have consistency. If you're doing an install, do it right. Have the right processing. Have it time-aligned and crossovered correctly. Buy something good and solid, and I don't care if it's NEXO or JBL or whatever. Just do it right the first time. For the consoles, make sure the op amps are matched. For instance, left and right were completely off yesterday. Have things clean, set up, and ready to go for consistent use. Too often, it turns into audio surgery for most rock club shows when I'd rather just mix."

Nevertheless, regardless of how venues choose to spend money on cosmetic updates and house PA systems, I assure you that professionals such as Stephen Shaw will continue to make the most of the latter.

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

CONTRACTING

The latest news and products

NEW PRODUCTS

STAGE ACCOMPANY DS24 Cinema Speaker

The DS24 is the main speaker in Stage Accompany's new Direct Screen cinema speaker line. Designed to be placed around the screen rather than directly behind it, the DS24 is suitable for large-scale mobile video presentations, quick set changes in multipurpose venues or for use with nonperforated screens. The DS24 has a single 12-inch woofer and a Stage Accompany ribbon mid/high-frequency driver. There is an optional powered version.

PRICE: app. $2,250.

CONTACT: Stage Accompany | ☎ 800-955-7474 ➲ www.stageaccompany.com

STACO ENERGY UniStar IIILA Rack Mount UPS

Though they aren't "audio" products, increasingly audio installers find themselves having to look at "power products" to install along with audio equipment – to protect that equipment and often to enhance performance of that equipment. Staco Energy Products offers the UniStar IILA, a rackmountable modular uninterruptable power supply for consideration. Onboard replaceable batteries offer seven minutes continuous full power after the lights go out. Also onboard is a power conditioner for cleaning up messy current. Windows/Mac/Linux software allow for remote monitoring and operation. The UniStar IIILA is available in 1 kVA, 2kVA and 3 kVA sizes.

PRICE: starts at $648.

CONTACT: Staco Energy Products | ☎ 866-261-1191 ➲ www.stacoenergyproducts.com

ROLLS RM67 Mic/Source Mixer

The RM67 Mic/Source Mixer from Rolls has received some significant upgrades – notably a duck sensitivity function and a 1/8-inch mini input on the front panel for local sourcing such as an iPod. As with the original design, the rackmountable RM67 has three mic inputs and four source/line inputs. Additional features include 12V phantom power and tone controls for the mic inputs along with a two-band EQ on the source inputs.

PRICE: $280.

CONTACT: Rolls | ☎ 801-263-9053 ➲ www.rolls.com

BARIX Annuncicom 100 Intercom

The Barix Annuncicom 100 intercom system is an example of how digital audio and control is rapidly reconstructing even seemingly mundane tasks such as intercoms. The Annuncicom 100 connects to IP or Ethernet networks and provides an input and output for audio. The network can either be an already established standard IT network or purpose-built. Using an IP address any Annuncicom 100 unit can be addressed by any other. Each unit has a microphone and lines inputs and a speaker output. Level and basic EQ can be handled remotely or locally by a web browser. A USB port is a bonus feature.

PRICE: $395.

CONTACT: Barix AG | ☎ 866-815-0866 ➲ www.barix.com

NEWS

The gang at Mac West Audio Group of Long Beach got to fly out to the USS John C. Stennis to do audio for a concert aboard the big aircraft carrier. They brought with them their favorite **Sabine** 2.4 GHz Smart Spectrum wireless system and Graphi-Q2 digital processor. See picture of a Sabine wireless mic receiver onboard.

Keeping on a jet-theme, the JET Nightclub at the Mirage Hotel in Las Vegas has installed **EAW** Avalon DCX, Avalon DCS and JL12 speakers. The Maxine Theater in Valley Center, Calif. has installed EAW AX speakers and SB subwoofers. To the north, Qwest Field in Seattle has installed EAW DSA array systems into several stadium clubs. Also in the Seattle area, EAW VR62 speakers and VRS18 subwoofers were put into the ACME Bowling and Billiard Events Center in Tukwila.

Keeping with the bowling theme – **Furman Sound** has placed PL-8 Pro Series II power conditioners into bowling alleys at the US Air Force base in Yokota, Japan and the GJ Scores Entertainment Center in Grand Junction, Col.

And while we're working out, **Australian Monitor** TX6000 installation mixers were installed into the 28 Equinox Fitness Clubs in the NYC area.

And on the day of rest… Resurrection Life Church in Grandville, Mich. has installed **QSC** AcousticDesign AD-S82 speakers. In Canada, the Quebec City Summer Festival used QSC WideLine arrays and WL218-sw subwoofers. Naturally, QSC provided the amp power – PowerLight2 PL230 amps.

"They Make Me Want to Sing"

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QSC's **New HPR122i** is so musical, powerful and natural sounding it reminds you why you're an entertainer. Driven by a 500 watt module based on the #1 selling RMX series amplifiers, the HPR122i delivers stunning accuracy and legendary QSC reliability.

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QSC

Powered Speaker Simplicity, QSC Reliability by Wayne Becker

Lectrosonics DM1612 Digital Automatic Matrix Mixer + DSP

Great sound and one of the easiest-to-use DSP matrix mixers on the market.

There are many DSP products on the market today that promise to make the installed sound designer, the installer, and end user's life easier. You know the drill: more tricks in the box for less. But you also know that most of the time, you will surely get a steep learning curve and unknown configuration caveats just waiting for that obscure application to say "gotcha." In this review, we'll look at the latest series of DSPs from Lectrosonics — the DM Series Digital Audio Processors. The series offers five configurations: the DM1624, the DM812, the DM84, the DMTH4 Telephone Hybrid and the unit under scrutiny in this review, the DM1612.

Suggested applications for this device are sound reinforcement and conferencing systems for boardrooms, courtrooms, worship centers, distance learning systems, and so on — basically, any environment that could benefit from an automixer with the additional features of a DSP.

| FEATURES

Using a cluster of four SHARC DSP chips and 24-bit A/D converters, the DM1612 architecture consists of 16 mic/line inputs and 12 outputs with a DSP-based crosspoint matrix that allows every input to be routed to any or all outputs. After the A/D conversion of each input, the signal passes through multiple EQs, an automatic digital feedback eliminator, a compressor, and a digital delay. In the matrix core, gain is adjustable from –69 dB to +20 dB in 1 dB steps at each crosspoint. Each of the 12 outputs provides a digital delay, multiple EQs, and a compressor/limiter. The DM Series of products can be linked together using conventional Ethernet cables in a master/slave system configuration.

Housed in a 2RU rack space unit, this unit has a very clean, uncluttered face with a just a few controls. The main interface is the four-line, backlit display flanked by a push for menu select/turn to select knob; two micro switches: one used for turning back in the menu and the other to assist in locking the front panel; and six menu item select buttons to the bottom of the display. There is also a USB interface port and the power switch. The flat black face with white lettering will look nice in most racks.

The back panel of the unit is very neatly laid out as well. Starting to the left is the power cord receptacle, a 1/8-inch RS232 serial port jack, a USB port, programmable I/O ports, Ethernet expansion ports, and the line out and mic/line inputs, which are landed on Phoenix connectors. The unit is compatible with Crestron and AMX control systems, has digital I/O ports to connect to other LecNet2 devices, and comes very neatly packed complete with all connectors, cables, cords, software and manuals.

The input chain is pretty straightforward and contains the basic building blocks one needs to process a signal from mic input to output. Each input channel provides four individual processing stages of gain, delay, filtering and compression. Input gain adjustment is available in a 0 dB - 50 dB range with adjustment in 10 dB steps and a fine adjustment in 1 dB steps. Next, a digital delay of up to 1 second can be applied in .5 ms increments. Following the delay stage is a complement of six different filter sets: low-pass, high-pass, band-pass, parametric, low shelving and high shelving. Additionally, you can select a filter slope with 6 dB or 12 dB per octave Butterworth or Bessel parameters. There are six automatic digital feedback eliminator filters and then a compression stage with adjustment parameters including threshold, attack, release, ratio and make-up gain. The limiter parameters include threshold, attack and release.

In the "Automixer Cell," level control for the automixing algorithm, mixing mode and crosspoint gain is applied to data gathered from other channels and devices. The cell receives data from the master unit in a multiple unit configuration and from the slave units farther down the chain. The automixer does not use a noise gate but rather a "Proportional Gain Algorithm" for a theoretically smoother response.

| IN USE

I decided to see how the unit sets up from the front panel before loading the software. In turning on the unit, the display greets you with the unit's model number and software version. There is no scrounging around and looking for that piece of important info. Turning and pressing the selection knob gets you into the next layer of submenus while the small select buttons at the bottom of the screen allows you to select the parameter of choice. Coming to the main menu, your selections are

FAST FACTS

APPLICATIONS
Installation

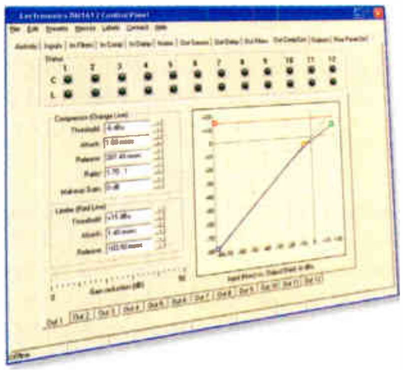
KEY FEATURES
16 x 12 matrix mixing; automixing; delay; low, high, band-pass filters; USB port; LecNet2 software; AMX and Crestron-compatible

PRICE
$3,170

CONTACT
Lectrosonics | ☎ 505-892-4501
www.lectrosonics.com

Setup, Presets, Sysinfo (system information), Lockset, Cmdview, Serial port and Exit. From the Setup menu, you can select inputs, matrix outputs and general.

Drilling down into the Input menu, here is where you choose attributes like phantom power, phase inversion, mute, gain, rear panel gain and input channel. It's probably getting picky, but I thought it would make better sense to have the first parameter be the channel selection, as it is it defaults to phantom power, but the first thing I found myself doing is selecting the channel first. The Matrix menu sets up the crosspoint settings like in and out points, mute, gain and mix mode. Next, the Output menu allows you to select the parameters like mute, source, rear panel gain and output channel.



It is important to point out that none of the DSP functions can be accessed or controlled from the front panel. All DSP control functions are accessed via the LecNet2 application. To do this, first you have to make a connection to the unit via USB or serial port. I chose to go the USB route and, the using the cable supplied with the unit, I made the connection to the front panel, then installed the LecNet2 software and USB drivers onto my laptop. This whole operation took only minutes and was clearly outlined in the accompanying manual. Once the software was loaded, I had to look for it because the installation process did not give me a choice to put a shortcut on the desktop. After locating the "LecNet2" folder, I noticed that control panels for all of the DM series of processors were loaded, and I simply selected the control panel for the DM1612.

When I fired up the application, it loaded up into the "off line" mode. Here, you can see all of the screens and parameters, but no changes will be made to the unit until you "connect" to the unit. Connecting to the unit is done through the "connect" pull down menu, and, once this is done, you are talking to the box with all changes made in real time. I would have liked to see some type of physical confirmation that I am connected to the box — a "connection established" notice or something of the sort.

The control panel is pretty straightforward. Each section of the box has a tab where you make your changes. I especially like the matrix panel, where color-coding makes I/O assignments fast and easy to understand. The tabbed layout makes each input, output and DSP function easy to get to and keep track of. I liked the fact that I didn't have to keep an eye on a DSP meter to make sure I wasn't going to run out. I liked the "nailed up" approach, knowing that any function I need is going to be available at any time without a compromise in quality or function.

Connecting sources to the unit via the Phoenix connectors was quick and easy. Using a mix of dynamic and condenser mic sources and audio from a video source, I used the matrix mixer and automix function to set up some scenarios. Making changes with LecNet2-based control panel, I proceeded to put the rest of the modules through their paces. Delay — which can be set in time, feet or meters — was easy to navigate and the nge in the parameters of the effect was responsive. I would have liked to see a bypass switch so that signal can be compared with or without the delay setting. Having a built-in output source of a 1 kHz tone and pink noise is a nice touch for setting levels and testing connections. Using the macro editor for setting up identical channel parameters can be very useful and was easy to enable, record and apply. Another nice little feature is the ability to label the inputs and outputs. I noticed a slight delay in the response for the compressor and limiter meters, but the operation of the process was good. Overall, moving through the different processes was easy and was met with good results. The unit is clean, responsive to transients, and audio quality is good.

| SUMMARY

The Lectrosonics DM1612 has great sound quality, while an easy user interface and feature set make it a great choice for the intended applications. Our technicians have always grumbled about the complexity and learning curve of DSP boxes. This unit is easy to understand, program and manipulate. Additionally, as with the other DSPs in this series, using this box won't back you into a corner when it is time for expansion.

Wayne Becker is vice president of sales for Communication Systems, Inc. and has worked in the pro audio and systems integration business for 23 years. He also owns Westwires Digital USA, a music production and consulting company based in Allentown, Penn. He can be contacted at wbecker@systemsbycsi.com.

TESTING Continued From Page 30

analysis tool for touring sound engineers, but it's also becoming popular for setting up and tuning an installed sound system or a studio. In addition to the real time spectrum analysis and energy-time curves, SmAArt is also capable of measuring the system transfer function in real time using any source. This allows you to continuously monitor the frequency response of a hall during a concert by using the PA mix as the source. In addition, SmAArt has a built-in interface capable of controlling most of the popular remote-controllable signal processing devices such as equalizers or loudspeaker managers. This allows you to adjust, for example, crossover frequency or speaker delay right from the SmAARt program while watching the display of output vs. input.

ETF is a Windows-based acoustical analysis program which works in conjunction with your own sound card, microphone, and preamp (they recommend using a Radio Shack SPL meter!) providing measurements of frequency response using either a frequency sweep or MLS statistical test signal, RT60, SPL, and it offers a 3D "waterfall" display of frequency response over time (energy-time curve). A demo version to allow you to verify that ETF will run properly on your computer and with your audio hardware is available as a free download (*www.etfacoustic.com*).

The RightMark Audio Analyzer suite (RMAA) is a Windows-based program that turns your computer into a fairly complete test generator and measurement system. Originally designed to test the sound card in a computer by looping the output back to the input, it's useful (once you've established the limitations of your system by measuring your sound card) for measuring the characteristics of outboard equipment.

Operation is simple and results are fairly clear. Select a test and RMAA generates a test signal and applies frequency and level analysis algorithms to display frequency response, TMD, IMD+Noise, usable dynamic range, and provides a spectrum display to evaluate noise content. With a microphone and preamp, you can do some rudimentary speaker testing. There's a non-real-time mode for testing recorders. It also includes a utility which creates a test CD for checking the playback of a single-ended unit such as a CD player. Now, how much would you pay? That's the best part – RightMark Audio Analyzer is free for the downloading from *audio.rightmark.org*.

Over on the Mac side, SpectraFoo from Metric Halo is both a comprehensive metering system for recording, mixing, and mastering and an audio analysis program. SpectraFoo has probably the most comprehensive digital bitstream display and analysis of any of the tools discussed here. As with SmAArt, system transfer function can be measured using any source including music. The test generator offers white and pink noise, bursts, sweeps, and plain ol' sine waves.

TEST CDS

That CD player that you replaced with a DVD player last Christmas can be recycled into a useful test generator. *The Sound Check 2 Test and Demonstration CD* from InnerStudio is an example of one of several commercially produced test CDs. Another is published by the National Association of Broadcasters (NAB). Both contain a wide range of test tones, pink noise, and band-limited pink noise for acoustic measurements, as well as some off-the-wall specialized test signals. For example, the *Sound Check* CD, co-produced by engineer Alan Parsons and acoustician Stephen Court, in addition to test signals, includes a wealth of musical instrument and vocal tracks, sound effects, and five minutes of SMPTE timecode. The NAB disk includes IEEE standard VU and PPM meter response tests.

You can of course roll your own test CD. There are many utilities that generate fixed frequency or swept WAV files and pink noise that you can burn on to your own disk. You won't get the comprehensive tests found on the commercial disks, but you can custom-make whatever you need, from an hour of 1 kHz tone available whenever you want it to a set of alignment tones for analog tape adjustment or reference levels, pink noise for sound system adjustment, or your favorite reference music for auditioning monitors.

The advantage of a test CD is the convenience. The downside is that most CD players have unbalanced outputs, and the maximum output at full scale digital level is usually in the vicinity of around +12 dBu. This may not be hot enough to fully drive a modern digital input, but it's fine for routine bench testing.

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

UpSAMPLER

Capsule Reviews and Product Review Updates

Benchmark Media Systems, Inc.

H1 - Headphone Amplifier
Price: $450 Contact: www.benchmarkmedia.com
Review Rating: ★★★★

Let's face it. Most headphones amplifiers inside of stand-alone players, recorders, mixers, etc., aren't the best. They are added to offer the headphone function, but their parts build is usually not state of the art. A well-built headphone amp that takes the analog signal from a separate component gives you considerable accuracy through a good set of headphones.

I recently discovered a gem of an analog- input headphone amp, the H1, from Benchmark Media — the maker of those fine

digital converters I am always bragging about

This under-$500 1/3 rack unit is an incredibly open and accurate headphone amp. The component layout is simple: it's a small box with a volume control, 1/4-inch plug, rear-mounted balanced inputs, a phone line-terminated DC power cord, and, gulp, the biggest wall wart transformer known to man. Okay, it is not bigger than a bread box, but it does take up all the space on a wall outlet.

Despite the chunky power supply, this headphone amp sounds better than almost every device I have that contains a headphone jack. Using the AKG K701 and Ultrasone HFI-2000UE headphones, I compared the several devices internal amps versus the H1. The Alesis MasterLink's internal headphone output is decent, and the Sony PCM-700 DAT from the late 90s sounds okay. But when you plug in the outputs of those players into the H1, what a difference!

The H1 revealed way more width to the image, had less edginess in the low treble. Instruments, such as drum cymbals and acoustic guitars, were much more real. The higher the resolution, the better it sounded. The only headphone amp I had on hand that came close was the Benchmark DAC1's built-in headphone amp, but that is because the H1 circuit is contained within the DAC1.

If I had wish list it would be a smaller DC transformer and an extra pair of input jacks for unbalanced. But as is, the Benchmark H1 is audiophile/high-end studio grade quality for a paltry $450.

—John Gatski

AKG K701 Stereo Headphones

Price: $449 Contact: www.akgusa.com
Review Rating: ★★★★

Over the years, I have used various headphones from AKG, Sony, Sennheiser, Grado and more recently Ultrasone. The new AKG K701 is by far one of the best sets of phones I have ever heard! Priced at $449 retail, these ultracomfortable headphones have excellent transient response, a balanced low end and stereo image with incredible detail.

Utilizing flat-wire voice coils and dual-layer Varimotion diaphragms with neodymium magnets, this open-back phone has a wide-open detailed presentation— especially in the mid and treble. The headphones are very lightweight and non-fatiguing to the ears. Features include biwired (balanced) drivers, leather headband, oxygen-free cable with mini jack adapter.

In lengthy listening tests using the ultra-accurate Benchmark H1 headphone amp reviewed in this issue, I compared the K701 to the Ultrasone HF-2000UE and the Ultrasonic HP-750 and AKG's sealed K271. Sources included prerecorded DVD-As and SACDs and home-brew acoustic guitar DVD-As at 24-bit/96 kHz. A LavryBlue DAC was used as the digital converter for the home brew recordings.

Compared to the closed back AKG K271, the AKG K701 is more accurate in the midrange and treble with incredible precision on transients. Cymbal whacks and rim shots are very real sounding. Bass is tight without midbass emphasis. And that low treble punchiness of the K271 is not there at all with the K701.

I compared the K701 to my other favorite headphones, the soft dome driver-designed Ultrasone HFI-2000UE. The 2000UE is fairly accurate across the top end, but relays a bit more prominence in the midbass. Imaging is good with the 2000UE, but the K701 is something else when it comes to stereo presentation. The Ultrasone ProLine 2500 had a punchier midrange/low treble emphasis and could not match the wide spacious image of the K701.

If you use headphones regularly and you want accuracy, detail, imaging and comfort, I strongly recommend the AKG K701. The K701 is not cheap at the retail price, but I found a quite a few online retailers that sold it under $350.

—John Gatski

Review Rating Key

★★★★ = Excellent Product. Go Out and Buy It
★★★ = Good performer. Worth a look
★★ = Okay. Worth buying at a discount
★ = Don't Click on the Buy-It-Now button

2006 Reviewer's Pick Awards

Digital Converters/ Clocks/Soundcards
- Lynx Aurora Digital Converter
- RME ADI-192 DD Universal Format Converter
- Weiss Engineering ADC2 Converter

Studio Software/Plug-Ins
- Ableton Live 5
- BIAS Peak Pro XT 5
- Digidesign Pro Tools 7 HD
- Roger Nichols Pro Bundle Plug-Ins
- Sony Oxford/ EQ Dynamics Plug-Ins

Studio Microphones
- AEA R92 Ribbon Microphone
- AKG Perception 200
- DPA SMK4061 Stereo Microphone Kit
- Electro-Harmonix R-1Ribbon Handheld
- M-Audio Sputnik Condenser

Microphone Preamp/Digital Converter Combos
- API A2D
- Lavry Engineering LavryBlue Modular System

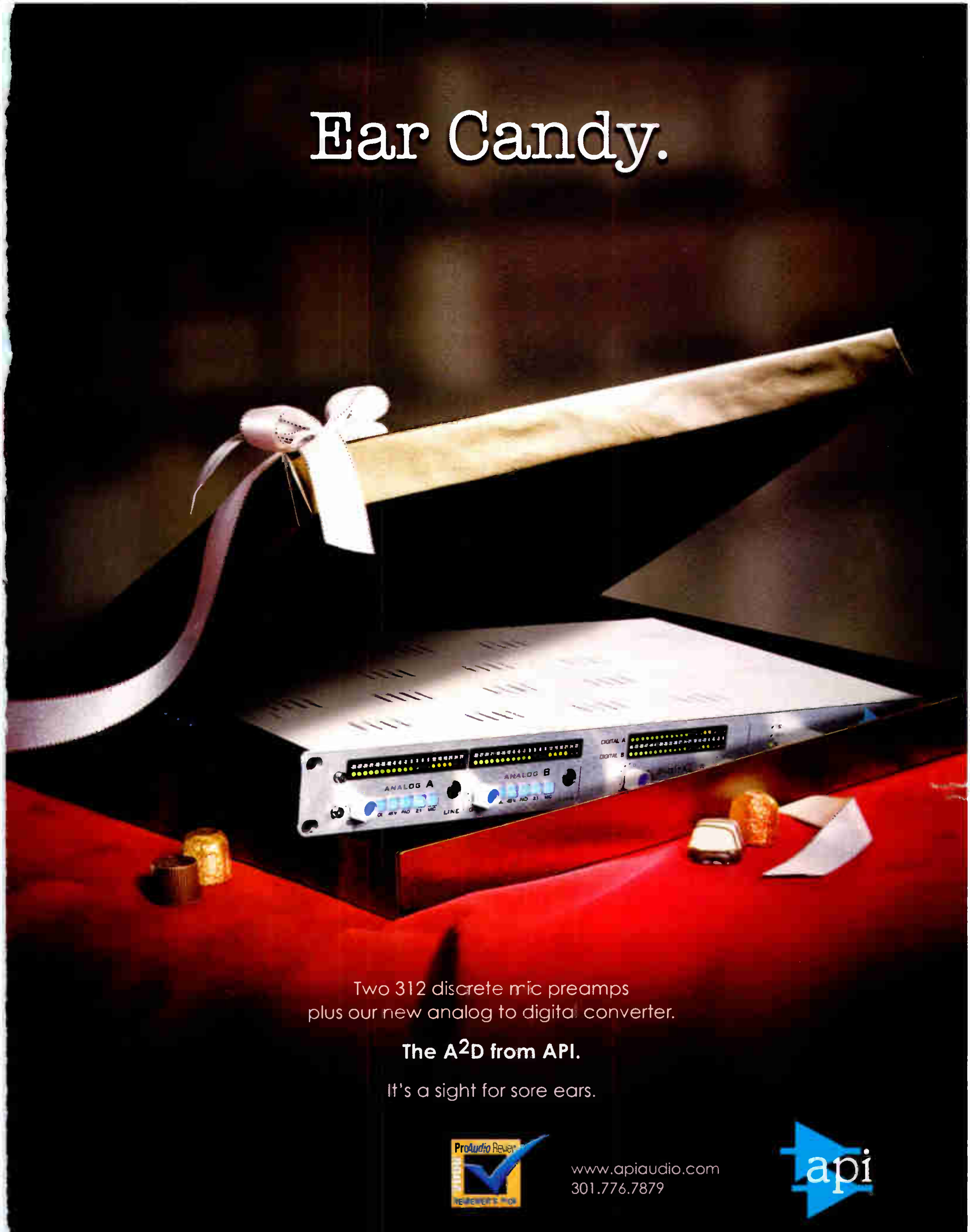
Studio Mixers
- AMS Neve 8816 Summing Mixer
- API DSM-24 Discrete Summing Mixer

Hardware Studio Processor
- Wheatstone Vorsis

Studio Monitors
- Alesis Powered M1 Active 620
- ATC Passive 20
- Dynaudio Acoustics BM5A Powered Monitors
- JBL LSR 4328P
- SLS PS8R Ribbon Monitors

Studio Amplifiers
- Bel Canto e.One Digital Amplifier
- Pass Labs X350.5 MOSFET Studio Amplifier

PAR PICKS Continued On Page 56



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ProAudio Review
REVIEWER'S PICK
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2006 Reviewer's Pick Awards

Continued From Page 54

Live Sound Microphones
- Shure KSM9 Handheld Microphone

Live Sound Speakers
- A-Line AL 10 Line Array
- JBL SRX 700 Series
- SLS RLA/2 Line Array

Live Sound/Contracting/ Installation Processors
- dbx Driverack 4800
- Klark Teknik Square One Dynamics Processor
- Meyer Sound Galileo
- Sabine Navigator

Live Sound Amplifiers
- Crest CC4000
- QSC PLX3602

Live Sound/Contracting/Installation Consoles (Digital)
- Digidesign Venue Live
- Midas XL8 Digital Console
- Yamaha M7CL

Live Sound/Contracting/Installation Consoles (Analog)
- Soundcraft MH2

Contracting/Installation Speakers
- Community R.25

Contracting/Installation Microphones
- Audio-Technica Unipoint Series

Contracting/Installation Amplifiers
- Lab.gruppen C68:4

Digital Audio Distribution (Live Sound/Contracting/Installation)
- RSS Digital Snake

Powered Mixers/PA Products
- Carvin RX1200 Powered Mixer
- Yamaha EMX 5014c Powered Mixer

Field Recorders (Recording/Broadcast/Cinema)
- Edirol R-09 Digital Recorder
- HHB FlashMic DRM85 Microphone/Recorder
- TASCAM HD-P2 Digital Stereo Recorder

PAR PICKS Continued On Page 58


BIG VENUE SOUND ON A SMALL VENUE BUDGET.
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2006 Reviewer's Pick Awards

Continued From Page 56

HeadPhones/In-Ear Monitoring Systems

- AKG K701 Headphones (Studio)
- Benchmark H1Headphone Amplifier
- Ultimate Ears UE-10 Pro In-Ear Monitors (Live Sound)

MI Products

- TC Electronics G-System Processor

Accoustic Products

- sE Reflexion Filter

Computer Hardware

- Apple MacBook Pro

Test/Measurement

- Audio Precision APx585 System
- Lectrosonics TM 400 Wireless Measurement system

Wireless Microphone Systems

- Audix RAD-360
- Shure UHF-R

AURORA INTERFACE OPTIONS

AES16

PCI card offers direct connectivity via PC or Mac to all 16 digital I/O channels with remote control. Includes Aurora software mixer for added routing and 64 channels of metering.

LT-ADAT

Expansion card provides up to 16 channels of ADAT Lightpipe I/O at 48 kHz. Supports higher sample rates using S/MUX. Permits format conversion between ADAT, AES/EBU and analog I/O.

LT-HD

Expansion card provides digital input and output in a format that is recognizable by Digidesign® Pro-Tools | HD®. Operates with all HD-compatible versions of ProTools. Supports up to 32 I/O channels at sample rates up to 192 kHz.

LT-FW

Expansion card available Fall 2006. LT-FW provides a 16 channel cross platform FireWire® interface.

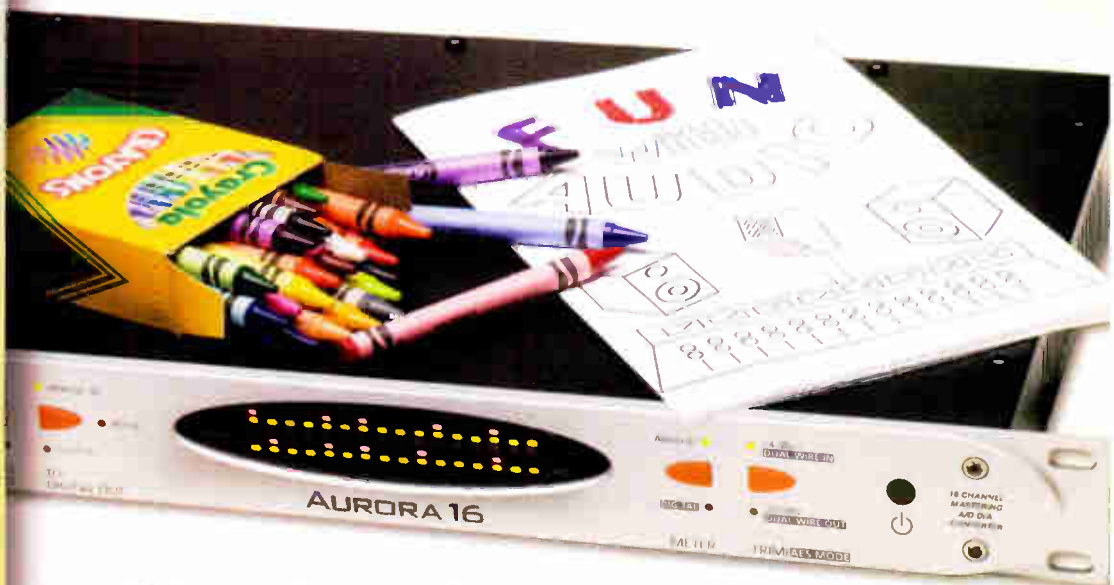
LYNXTWO-AURORA INTERFACE

Cabling kit gives LynxTWO and Lynx L22 owners direct connectivity for up to 16 channels of Aurora I/O.

1824 AURORA TRIM OPTION

Aurora 16/1824 and Aurora 8/1824 models feature +18 dBu and +24 dBu full scale trim settings, which replace the +6 dBV and +20 dBu full scale trim settings of standard models.

If you're looking for coloration from an Aurora Converter



This is the only way you're going to get it.

ProAudio Review
REVIEWER'S PICK

When we designed the Aurora 16 and Aurora 8 AD/DA converters, we had a simple goal. Converters with clear, pristine, open sound and no coloration or artifacts. We wanted you to be able to get the identical audio out of Aurora converters that you put into them. From what we have heard from you and the major magazines, that's what we have accomplished.

Aurora includes no compression, no limiting, no equalization. No coloration. Why?

First, if you want or need coloration, you already have that handled. You have carefully selected your signal processing, which you can add to the signal chain at any point you like, or leave it out altogether.

Second, how would we know what processing would fit your needs and your tastes? We could nail it for our tastes and for a few of our friends, and completely miss what you want.

Third, we wanted to build the best possible AD/DA converter – period, not a converter/signal processor/preamp/exciter. Adding these functions would add the price of Aurora, for features you may not want or need.

Instead we packed in features such as our exclusive SynchroLock™ word clock, LSlot expansion port for optional interfaces, and exclusive remote control options into a single rack space format. And, most importantly, world-class audio quality that rivals converters costing many times the price.

Aurora 8 and Aurora 16 from Lynx Studio Technology. We'll handle the conversion and leave the coloring up to you.

Want more information (like there's not already too much in this ad)? Go to: **www.lynxstudio.com/aurora3**

Lynx STUDIO TECHNOLOGY

LIVE
| Feature

MIDAS Continued From Page 43

the XL8 was unparalleled in the ease of setup, flexibility, and use.

So — what about the sound, you ask? Danny Slaughter explained that Westover had a Verona at front-of-house in their new sanctuary while they awaited the arrival of the XL8. The system sounded great, but when the XL8 arrived, "everything completely opened up," so much so that they had to retune the room and adjust for the clarity of the fully digital system.

I can attest to this as I had already noticed the fine acoustic detail of the seven-piece band and choir during the church's Sunday service. I especially enjoyed the use of the matrix outputs to feed the choir into the surround speakers hung around the perimeter of the sanctuary. This gave a fairly modest-sized choir a huge sound while not taking away from the worship experience. The output timing and phase coherency was spot on with the XL8, automatically doing all of the calculations and adjustments. At Westover, this included the interface of analog feeds from the primary DL431s and DL451s handing additional inputs (both from analog and AES) that included audio from sources sprinkled throughout the sanctuary.

| SUMMARY

The XL8 is a comprehensive system that has been well thought out and was designed to mix the best of analog "feel" with the flexibility and control of a digital system. At $340,000 retail, the XL8 is not for the faint of budget. However, this price may appear higher on the front-end since the XL8 is shipped with all necessary cabling, hardware, and road cases; just add amps, microphones and something to mix, and enjoy creating an audio masterpiece.

Dan Wothke began his journey as a musician and naturally progressed to exploring audio engineering. He has since accrued over 10 years of studio, live sound, and technology experience. Dan currently runs the gauntlet of all things media in his role as Media Director at Belmont Church in Nashville.

Letters

LETTERS Continued From Page 7

blender set to "puree." I keep the original media copies because, #1 this makes my compiled CDs legal - provided that I keep them to myself (which I do), and #2 I like the big LP jacket artwork. Also, the rummaging through yard sales and bulk trash put-outs for old LPs has replaced my desire to hit the now-closed independent record stores for new-to-me music.

The part that gets me is that people in my age-range (mid-40s) are in their peak earning years. Instead of offering us quality music with a classic rock bend, they cater to the 12-year-old market with really bad bubblegum music. Mind you, this stuff they produce isn't Marc Bolan, the 1910 Fruitgum Co., Melanie, or Lemon Pipers caliber; it is just a really poor synthesized music track meant to go along with the video. That's what they've reduced audio to: the bastard son of a music video. Sometimes I think that if the style guys had their way, we'd all be watching silent music videos!

Ben Torre

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Conference: March 13 – 17, 2007 | **Expo:** March 15 – 17, 2007

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LYNX STUDIO TECHNOLOGY Aurora 8/Aurora 16

FEATURES: Combined A/D-D/A unit; eight or 16 channels; 44.1 kHz, 48 kHz, 88.2 kHz, 96 kHz, 192 kHz; onboard 32-channel digital mixer; SynchroLock antijitter technology.
PRICE: Aurora 16 - $3,295; Aurora 8 - $2,195.
CONTACT: Lynx Studio Technology at 949-515-8265, www.lynxstudio.com.

WEISS ENGINEERING ADC2

FEATURES: Analog to digital converter; two-channel; 44.1 kHz, 48 kHz, 88.2 kHz, 96 kHz, 176.4 kHz, 192 kHz sample rates; 24-bit; 48V phantom power; onboard digital peak limiter; POW-R dithering; clock input.
PRICE: $6,500.
CONTACT: Weiss Engineering at 41-44-940-2006, www.weiss.ch.

SONIFEX RB-ADDA2

FEATURES: Combined A/D-D/A unit; 32 kHz, 44.1 kHz, 48 kHz, 88.2 kHz, 96 kHz, 176.4 kHz, 192 kHz sample rates; word clock.
PRICE: $1,289.
CONTACT: Sonifex/Independent Audio at 217-773-2424, www.independentaudio.com.

TC ELECTRONIC Konnekt 8

FEATURES: Analog to digital converter; IMPACT mic preamps; FireWire port; linkable; Jitter Elimination Technology; headphone output; ships with Steinberg Cubase LE.
PRICE: $375.
CONTACT: TC Electronic at 818-665-4900, www.tcelectronic.com.

SM PRO AUDIO ADDA 192-S

FEATURES: Combined A/D-D/A unit; two-channel; 24-bit/192 kHz; 48V phantom power; 20 dB pad; headphone amp. **PRICE:** $499.
CONTACT: SM Pro Audio/Kaysound at 514-633-8877, www.smproaudio.com.

BENCHMARK MEDIA SYSTEMS ADC1 Converter

FEATURES: Analog to digital converter; two-channel; 44.1 kHz, 48 kHz, 88.2 kHz, 96 kHz, 176.4 kHz, 192 kHz sample rates; 24-bit; AES, ADAT outputs; UltraLock anti-jitter technology; clock input.
PRICE: $1,775.
CONTACT: Benchmark Media Systems at 800-262-4675, www.benchmarkmedia.com.

MOTU 8pre

FEATURES: Analog to digital converter; eight-channel; 24-bit; 44.1 kHz, 48 kHz, 88.2 kHz, 96 kHz sample rates; 24-bit; 48V phantom power; 20 dB pad; FireWire port; monitor output.
PRICE: $595.
CONTACT: MOTU at 617-576-2760, www.motu.com.

CEDAR ADA

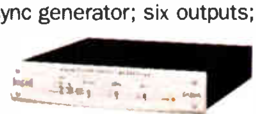
FEATURES: Combined A/D-D/A unit; two-

channel; 24-bit/96 kHz.
PRICE: $1,175.
CONTACT: CEDAR/Independent Audio at 217-773-2424, www.independentaudio.com.

LAVRY ENGINEERING Blue 4496 Converter

FEATURES: Modular A/D-D/A; up to eight channels; 44.1 kHz, 48 kHz, 88.2 kHz, 96 kHz sample rates; 32 – 100 kHz Varispeed; double or single wire configurations.
PRICES: two-channel A/D module - $868, two-channel D/A module - $840, basic chassis - $485.
CONTACT: Lavry Engineering at 206-381-5891, www.lavryengineering.com.

LUCID GENx6-96 Studio Sync Generator



FEATURES: Clock/sync generator; six outputs; 44.1 kHz, 48 kHz, 88.2 kHz, 96 kHz sample rates; word clock; Superclock.
PRICE: $499.
CONTACT: Lucid at 425-742-1518, www.lucidaudio.com.

DRAWMER M-Clock Master Clock

FEATURES: Master clock generator; up to eight outputs; 44.1 kHz, 48 kHz, 88.2 kHz, 96 kHz, 176.4 kHz, 192 kHz; word clock, Superclock; onboard sample rate converter.
PRICE: $1,500.
CONTACT: Drawmer/TransAudio Group at 702-365-5155, www.transaudiogroup.com.

PRISM SOUND ADA-8XR

FEATURES: Combined A/D-D/A unit; eight-channel; 44.1 kHz, 48 kHz, 88.2 kHz, 96 kHz, 192 kHz; 2.8224 MHz, 5.6448 MHz sample rates; 16, 24-bit; peak limiter; noise shaping; word clock; Superclock; DSD-compatible.
PRICE: starts at $11,200.
CONTACT: Prism Sound at 973-983-9577, www.prismsound.com.

API A2D

FEATURES: Analog-to-digital converter; two-channel; 24-bit; up to 96 kHz sample rate; API 312 mic preamps.
PRICE: $1,995.
CONTACT: API Audio at 301-776-7879, www.apiaudio.com.

DIGITAL AUDIO DENMARK AX24

FEATURES: Modular combined A/D-D/A unit; up to eight channels; 44.1 kHz, 48 kHz, 88.2 kHz, 96 kHz, 176.4 kHz, 192 kHz, 352.8 kHz; DSD, DXD-compatible.
PRICE: $6,000.
CONTACT: Digital Audio Denmark/ Las Vegas Pro Audio at 702-273-0731, www.lasvegasproaudio.com.

VIDEOQUIP DAC2

FEATURES: Digital to analog converter; two-channel; 24-bit; 44.1 kHz, 48 kHz, 88.2 kHz, 96 kHz sample rates; word clock.
PRICE: $895.
CONTACT: Videoquip at 888-293-1071, www.videoquip.com.

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by Strother Bullins

"Close To You" | Dionne Warwick (with Mya)

SINGLE: "Close To You"
ALBUM: *My Friends & Me* (Concord Music Group)
RECORDING DATE: Summer 2005 at the Beach House in Marina del Rey, California and Mya's private studio in Washington, DC
SINGLE PRODUCER: Damon Elliott
SINGLE ENGINEERS: Damon Elliott and Ariel "A Train" Levine
SINGLE MIXERS: Ariel Chobaz and Dave Pensado
MASTERING: Brian "Big Bass" Gardner at Bernie Grundman Mastering in Los Angeles
OTHER PROJECTS: Elliott has worked with superstars including Pink, Whitney Houston, Destiny's Child, Kelis, Jessica Simpson, and Ziggy Marley and the Melody Makers, among others.
SINGLE SONGWRITERS: Burt Bacharach and Hal David
RECORDING CONSOLE/CONTROLLER: Digidesign ICON
RECORDER: Digidesign Pro Tools|HD3
MONITORS: TAG/Augsperger custom mains, Yamaha NS-10
MICROPHONES: Neumann U 67, Neumann U 87, and AKG C12
MICROPHONE PREAMPS: Neve 1073, Avalon VT-737SP tube preamp
VOCAL CHAIN (WARWICK): Neumann U 67 and Avalon 737 straight to PT with a Waves L1 Ultramaximizer

ENGINEER'S DIARY

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

Currently celebrating her 45th year as a vocal superstar, Dionne Warwick has no dearth of hit singles. Proof is her latest album entitled *My Friends & Me*, a re-recorded collection of original number one hits by Dionne (mostly penned by the powerhouse songwriting duo of Burt Bacharach and Hal David), but, this time, appearing as duets with both old and new vocal acquaintances alike. The collection, produced and engineered by the musically prolific Damon Elliott, is notable, not only for its grand scope, bevy of comprising vocal talent, and balanced blend of both old and new tonalities, but for its being an R&B royal family affair (Dionne is Damon's mom).


Producer/Engineer Damon Elliott

How was it to produce both mom and one of golden voices of pop? "Challenging," states Damon while explaining the challenges were the fun, invigorating kind. "It was all about dealing with major, super, humongous hit records that the whole world has heard a few hundred times in a few hundred different ways," he qualifies. "So it was my goal to flip the songs — to update them — while keeping the integrity of the originals."

In blending the voices of Warwick and young R&B songbird Mya on the first single, "Close To You," Elliott found the two vocal styles to mesh even better than he had originally envisioned. "My mom has such a thick tone and voice, she doesn't require anything effects-wise," tells Elliott. "With Mya, I wanted it to sound like Mya, but a bit more of a 'grown-up' Mya. Because I know both voices and their patterns, I knew how it would work — and it worked even better than that!"

A busy Mya tracked her vocals at her own private DC-area facility, which were then combined by Elliott with the lush soulfulness of Dionne, captured with a chain comprised of a Neumann U 67, Avalon VT-737SP, and a Waves L1 limiter within PT|HD. "She would have her parts done in 15 minutes — with backgrounds and harmonies — and I never flew any vocals," tells Elliott with an equal blend of awe and family pride. "Every time the hook comes around, it's a new hook. It's easy to fly hooks in Pro Tools, yet she wanted to sing it every time."

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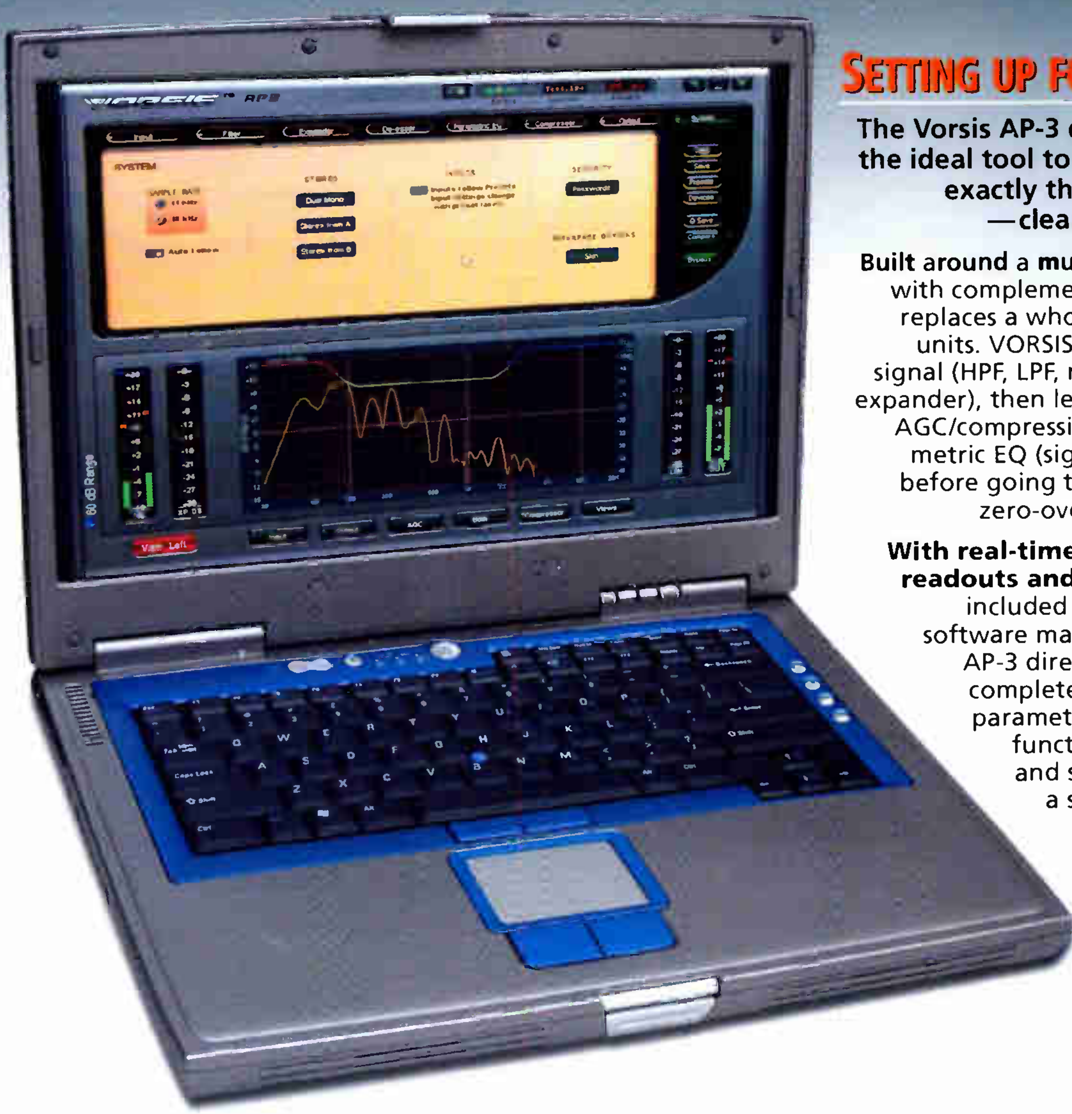

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