

AES Show Review
San Francisco, California
October 28 – 31 2004

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After a too-long vacation in Los Angeles, the Left Coast US Audio Engineering Society convention returned to San Francisco, much to the joy of the attendees and exhibitors. The weather was great, the air clear, the food great, the Halloween costumes much in the keeping of the spirit of the city, oh, and the audio stuff was pretty cool, too. Generally I fill these reports with short reviews of neat new products, but with web sites and magazine ads, much of what I saw for the first time at this show has already been pretty well publicized. I'll be writing about some product highlights that support trends and directions in which the industry is going as well as my usual reports of gotta-have goodies.

Maybe it's because I've been to so many of these shows, but it seemed like this one was more social than usual, meaning more stopping at a booth and talking for half an hour than just getting a quick run-through on a product and moving on. It took me all of the four days to get through the sixteen aisles of the show with only a few diversions to the demo rooms and a couple of technical sessions. So, the usual disclaimer applies. If I didn't write about something you were hoping to read about, either it simply didn't interest me or I decided that it was already well enough publicized that I didn't need to spend time with it.

Now, on with the show!

Transducers – In and Out

There was the usual collection of new microphones, inexpensive and top-shelf, factory and hand-made, tube and solid state, big and small. The highlight for me was finally seeing the Stephen Paul microphone in a presentable state. Mic guru Stephen Paul had always wanted to build a microphone the way he thought it should be built, and a couple of years ago, Alan Hyatt of PMI (Studio Projects) offered to fund the project. Stephen, until his death in 2003, Tony Merrill (Stephen's shop sidekick), and Brent Casey of Studio Projects worked together to come up with the design and then figure out how to manufacture it affordably. While they had hopes of using PMI's Chinese facilities to build some of the parts, the precision just wasn't available and all the machining and assembly will be done in the US, with the capsule hand built in the Stephen Paul Audio shop. They're waiting for quotes for the metal parts to come in before any pricing and delivery is announced, but the good news is that the design is essentially complete and that the microphone will be a reality, and importantly, will carry Stephen's ideas of microphone design on to another generation.

<http://www.spaudio.com/>

Latest in the “How Low Can They Go?” category is from Audio Technica, the original condenser mic price buster. The new AT2020 is a side address cardioid condenser mic for \$99 list. The medium-sized (16 mm) electret element capsule takes an SPL of 144 dB so you can put it in front of a loud amplifier or kick drum without fear of overloading. It looks like a good choice for the ever-reduced-budget project studio user.

<http://www.audio-technica.com/prodpro/profiles/AT2020.html>

Two new mics for the casual recordist are the Audio Technica PRO24 stereo condenser microphone and the VideoMic from RØDE. A step down from the popular AT-822, the PRO24 has a captive cable terminating in a mini phone plug. It contains a pair of cardioid elements in X-Y configuration and is a suitable size and weight for camera mounting as well as using on a table stand for recording a club gig or a jam session. It takes an internal 1.5V battery or will work with a plug-in powered (not phantom) mic input. The RØDE VideoMic is a modestly priced short shotgun suitable for mounting directly on a digital video camera. It's powered by an internal 9V battery and has a built-in windscreen as well as a switchable high pass filter.

<http://www.audio-technica.com/prodpro/profiles/PRO24.html>

<http://www.rodemic.com>

Sanken introduced the COS-22 dual capsule lavalier mic. With two omni capsules in a tie-tack sized housing, it's good for redundancy in live broadcast applications. The new CO-100K gets its name from its suitability for use with 96 kHz sample rate recording. It's the first really wide frequency response mic designed for recording rather than measurement and actually has some measurable frequency response up to 100 kHz and has usable response in that “super-audible” range that we know is present with some musical instruments. According to the frequency response plot, it's pretty flat up to around 10 kHz, then rises about 10 dB up to about 80 kHz (kind of mountainous in this region), dropping a few dB at 100 kHz. For stereo applications, the Sanken CUW-180 is a single body containing two swivel-mounted cardioid elements for X-Y recording with adjustable width. The construction is such that two can be conveniently mounted back to back for four channel surround recording.

<http://www.plus24.net/sanken/default.asp>

Never a company to be shy about form, as long as they can retain function, Blue has a new condenser mic in a similar housing to the Blue Ball dynamic which they introduced last year. The 8-Ball is a phantom powered cardioid that they recommend for vocals as well as reed instruments. For fans of the vintage mic appearance, they now offer the optional Ringer dual ring shock mount reminiscent of the early broadcast mics. It would be just the ticket for the singer in your big swing band. In other Blue news, the Blue Bird, a bottle-style mic which was previously sold only as part of a Digidesign bundle, is now available on its

own. It's similar to the Baby Bottle, but has a bit flatter and more even frequency response.

<http://www.bluemic.com/>

Last but not least in the microphone stack of literature is the Earthworks drum kit system. Many of the mic manufacturers that have product lines oriented to project studios have offered sets of mics appropriate for drum recording. Earthworks is the latest. The Earthworks drum kit comes in two flavors. Both feature the cardioid SR25 cardioid mic for kick drum. The studio version includes two TC25 omni mics while the live version bundles two more SR25 cardioids for overheads. Both packages include the KickPad (also available separately), an in-line (XLR in and out) attenuator for the kick drum mic which also provides a shaped frequency response. The Drum Kit comes packed in a nifty wooden case that resembles a drum and includes the three mics, KickPad, a windscreen, and mounting clips.

<http://www.earthworksaudio.com/>

The usual speaker manufacturers were well represented, from A to at least L. Lipinski Sound, a company whose speakers have been getting quite a bit of attention from the mastering community exhibited at the show this year, but their speakers were out on the main show floor rather than a demo room, so there wasn't much point in trying to hear what they sound like. Turns out that designer Andrew Lipinski is right in my neighborhood, so one of these days I'll make an appointment to visit and listen.

<http://www.lipinskisound.com>

NHTPro showed a new pair of monitors that are related to their previous powered line only to the extent that it's an integrated system but the power amplifier chassis isn't integral with the speaker cabinets. They've taken two big steps with their new models, the M60Xd and M80Xd. The M60 is a 2-way speaker with a 6.5" magnesium cone bass driver and the same tweeter that was used in the company's A20 system. The cabinet is an unusual shape, wide at the top around the tweeter to provide a solid baffle for high frequencies, and narrower around the bottom with well rounded corners to reduce low frequency phasing effects due to reflections from the cabinet face and sides. The M80 has a more conventionally styled cabinet with two 8" magnesium cone woofers, two 2" aluminum dome midrange speakers, and a single 1" aluminum dome tweeter.

The thing that does the trick with these speakers is the XdA processor/amplifier unit which both share. This contains four (two per channel) Class D power amplifiers designed by Power Physics plus a DSP equalizer and crossover designed by DEQX. The DEQX allows very precise correction of the driver frequency and phase response and allows a much smoother crossover than with conventional designs. Since all the processing is digital, frequency response and phase shift of the crossover can be independent, eliminating the need for other means of phase correction in the crossover region. The ability to correct

frequency response anomalies of the drivers allows the designers to work with essentially perfect drivers and optimize crossover and cabinet design. In the M80, the XdA unit powers the mid and high speakers, and a separate power amplifier is provided for the woofers. The XdA unit is equipped with a USB port and a mic input for “future proofing” which I suspect means uploading new DSP code and perhaps some real time active compensation.

All of this doesn't come cheap, at least not yet. While the drivers aren't terribly costly, the DSP hardware and code still is. The M60 clocks in at \$4,500 while the M80 is \$7,000. Conditions at the show were of course not adequate to do a listening justice but early users have reported great sound, wide sweet spot, and low listener fatigue.

<http://www.nhthifi.com/2004/>

Mic Preamps

After a flurry of mic preamps for a few years, things seemed to slack off for a while. I guess they became too much alike, but this year it seemed that the really good got better, and some new concepts and alliances have emerged. In this day and age, we think of a “plug-in” as a piece of software, ADK, known for their microphones, showed the AP-1 mic preamp with plug-ins, which in this case means plug-in “990 style” op amp modules and input transformers. To change the characteristic sound of the preamp, just pop open the cover and change the transformer, the op amp, or both. The op amp pinout is that used by the Jensen, Hardy, and API modules as well as one made for ADK. The input transformer (Jensen, Lundhahl, and mysterious were at the booth) is mounted on a circuit board with a similar plug-in arrangement. I asked if one option was “transformerless” and they rep said they never thought of that. I wonder why not.

<http://www.adkmic.com/>

Buzz Audio Ltd., a boutique electronics manufacturer from New Zealand, has been around for close to 20 years and has had some presence among newsgroup posters primary from Down Under, but it's only recently that “the buzz” has reached the pro audio community at large. This show they introduced the MA2.2 discrete (class A of course) dual channel mic preamp with transformerless input and an optional transformer output. A good companion to the Sanken 100 kHz mic, the MA2.2 frequency response goes out to 1 MHz, and is only 3 dB down at 250 kHz. Up to 65 dB of gain is provided, with a maximum output level of +24 dBu from the transformerless (unbalanced) output.

<http://www.buzzaudio.com/products/ma2.2.htm>

Last year I was told that PreSonus was working on a preamp in conjunction with designer Anthony DeMaria. who has his own ADL line of high quality studio products. It's come to light as the Presonus model adl 600 two channel tube preamp. This one is built in the US and it feels like quality all over. Three tubes per channel, adjustable input impedance (including a high impedance instrument

pickup), and a stepped gain control with a continuous trim overlapping the steps. Near the lower end of the Presonus line is the Bluetube dp two channel mic preamp which offers both a tube and solid state signal path. Basic gain control is through the solid state amplifier, with adjustable (or defeatable, for us purists) tube “drive.”

<http://www.presonus.com/>

Lipinski Sound showed two new solid state preamps with their main difference being the input transformer design. The model L-408's transformer has a mu metal core, providing a 200 kHz bandwidth and maximum input level of +16 dBu. The model L-409 has an amorphous core providing an 80 kHz bandwidth and maximum input level of +29 dBu. They sound different, the later recalling the “vintage” sound, while the former one is cleaner. Both are powered from an external power supply (the model L-410). Unique to the design is the use of two cascaded gain stages, each with its own level control and metering. The first stage is discrete while the second stage uses high speed op amps. Since the transformers have selectable taps for variable impedance loading of the microphone (and hence affect the gain through the transformer), the impedance switch plus the combination of the setting of the two gain controls give a range of tonal quality from really clean to crunchy. Construction is top notch, with extensive RFI shielding.

<http://www.lipinskisound.com/products/electronics.html>

In last year's report, I gushed over the attention to detail and design and component selection taken to the extreme in the Gordon mic preamp. It was hard to believe that they could top this, but sure enough, last year's dual channel model 3 has been superseded by the Model 5, with the Model 4 single channel version added to the line. These new models use basically the same circuitry and design philosophy as the previous model but the grounding system has been totally revamped, lowering the noise floor by a couple of dB. Designer Grant Carpenter said “Since the AES now has a recommended standard for ground design, I might as well follow it.” The dual channel model, as in the previous model, is remote-controlled, with the remote panel handling up to four channels. The single channel model has integral controls.

<http://www.gordonaudio.com>

A new offering from Manley Labs is the TNT (tube and transistor), a dual preamp that's truly dual – the two channels are different, one being a tube circuit, the other being solid state. No, they don't sound the same. While they may be applicable to using both channels on a single instrument, it would be pretty strange with a stereo mic pair, but then that's not the purpose. Input impedance on both channels is variable, though with a different range for each, both including a high impedance instrument setting. Curious sound modifying controls are an “Iron” control that enhances the transformer sound, and a switch labeled 60s and 70s (there's an off position) which attempts to add the characteristics of preamp designs of the 1960s and 1970s.

<http://www.manleylabs.com>

Lavry Engineering is another “If it isn’t the best, you didn’t get it from us” company who’s well known for high quality A/D and D/A converters. New this show is the Lavry dual channel mic preamp that fits the Blue series chassis. This is a modular system which allows custom configuration of A/D, D/A, and now mic amplification in a single rack space chassis. The mic preamp fills two slots of the chassis for two channels (there isn’t a single-slot single-channel version) so a typical Lavry mic preamp with digital output would be comprised of a Blue rack, a preamp, a clock generator/sync module, and an A/D converter. Alternately two preamp modules can be fitted to a single rack for a four channel analog in/out unit. Gain is set digitally and displayed on a numerical readout on the front panel. The two channels can be set to track or to be individually adjustable. Gain is saved in non-volatile memory so it returns to its previous setting after being powered down. The input is transformerless but with common mode rejection exceeding that of a transformer.

<http://www.lavryengineering.com>

One thing that these preamps share, even down to the low priced PreSonus, is careful attention to detail in both design and construction. I like that. One more note - The model numbers and transformer characteristics for the Lipinski preamps came off the data sheet I brought home with me. I might be out to lunch here (my notes and memory are a little fuzzy) but I seem to remember this being the preamp I saw which had two input transformers and a front panel switch to select which one was in use. Perhaps there is a third model with both transformers. If not Lipinski, then somebody. A very cool idea.

Signal Processors and Channel Strips

Rupert Neve never gets a vacation. After moving from Focusrite, he’s now designing circuitry for a new company, Legendary Audio. Masterpiece is a new analog mastering suite, a modular system with modules that are a combination of new circuitry and classic Neve designs. Designed to augment a mastering system (as opposed to a recording chain) many of the controls have stepped resolution as small as 0.25 dB. The main module is a stereo I/O control with input and output gain controls, metering, and selectable high and low pass filters. A mysterious Image Control section with knobs for “Depth” and “Ambience” enhances or reduces the ambient level in a stereo recording. I suspect it’s some sort of M-S (sum-and-difference) manipulation.

There’s a multi-function module with a Tape Texture section which employs a magnetic circuit (is this marketing-speak for easily saturated transformer?) to control tape-like saturation. There’s also an all-pass filter section to provide a continuously variable phase shift similar to the much touted Little Labs IBP. An Auxiliary section of this module provides gozintas and gozoutas for inserting outboard equipment into the signal path as well as inserting an outboard

equalizer into the sidechain of the dynamics module. A selectable filter for each of these three sections allows the process or insert to work broadband, low frequencies or high frequencies only, or mid-band. The dynamics module has the typical compressor controls with the ratio adjustable from 1.1:1 to hard limiting. Multiple dynamic modules can be linked for multi-channel tracking, and a soft knee characteristic is switchable.

There are two equalizer modules intended to work individually or in tandem. One is a three-band parametric peaking equalizer, the other a high and low frequency shelving equalizer. A "Classic" button engages a transformer-coupled discrete circuit for added analog warmth. A Zoom button on each of the equalizer sections switches gain adjustment resolution to ¼ dB steps from the un-zoomed setting. <http://www.legendaryaudio.com>

New from Empirical Labs (the Distressor and FATSO folks) is the Lil FrEQ parametric equalizer with a dynamic EQ section. The high and low shelving, and four parametric sections are pretty much as expected with the bonus of lower THD (0.0007% claimed) in the parametric section than you'll find anywhere else. The high pass filter has a sharper cutoff than typical, with a little bump just above the cutoff frequency to prevent excessive thinning out of the low end when it's engaged. The dynamic section can be tuned for de-essing or can function as a soft-knee high frequency limiter to soften up a harsh high end. Big Distressor-style input and output level knobs let you know to which family it belongs. <http://www.empiricallabs.com/>

The ARC1.1 is a new and highly flexible analog channel strip from Buzz Audio. The input stage is similar to their earlier SSA-1.1 mic preamp design with mic, line and instrument inputs, variable loading on the mic input, and a switch that adds 15 dB of gain for a maximum mic gain of 65 dB. The output section has a "Tranny" switch for extra coloration. The equalizer section consists of a high pass filter, variable low and high shelving, and two fully parametric sections. Each section can be individually bypassed, or can be switched into the sidechain of the compressor. Compressor gain control can be selected between an optical soft-knee or fast acting FET element. The compressor can be switched to be either before or after the EQ section. The ARC1.1 is intended primarily as a vocal input chain, but of course can work for anything including processing line level signals. <http://www.buzzaudio.com/products/arc1.1.htm>

New from Weiss Engineering is the DNA1 Denoiser and ambience processor. It incorporates PureNotes proprietary real time noise removal technology for automatic removal of noise such as analog tape hiss without adding digital artifacts. The DNA1 also includes Bob Katz' K-Stereo processor which recovers and enhances (it doesn't create what isn't there) ambience in a stereo recording, and even does a good job of adding extracted ambience in stereo to a mono recording. Weiss is known for extremely high quality and really solid digital

design. This is not a casual “clean up your old cassettes to put on your MP3 recorder” device, but is a serious tool for the mastering engineer.

<http://www.weiss.ch/dna1/dna1.html>

A/D and D/A Converters and Interfaces

Whether or not you believe the pro/con 192 kHz sample rate arguments, it looks like it's here to stay, until something else is invented. So naturally there's a new crop of converters that accommodate the higher sample rate reasonably gracefully. First on the block is the Aurora 8 and Aurora 16 single rack space 8 and 16 channel (respectively) A/D-D/A converters from Lynx Studio Technology – analog to single or dual wire AES digital and back. Apparently people have been asking Lynx, known for their high quality sound cards, to build an external converter. Now that they have the AES16 16-channel AES/EBU I/O card for a computer, the Aurora boxes are a good companion, but can work with any other AES/EBU interface, and, in the future, Firewire, ADAT, and maybe other interfaces.

The Aurora looks pretty innocent from the front panel, but it's packed with interesting features that aren't immediately obvious. While all functions are accessible from the front panel (including metering of all the channels) it can be remote-controlled by the infra-red protocol used in popular pocket PC units. It's also aware of the Lynx AES16, and when it detects that it's connected to one, makes clever use of the AES user bits to remotely control the Aurora from the AES16 control panel.

In addition to controlling the basic settings such as sample rate, digital format, and word clock source the Aurora includes an internal router allows you to send any input to any output (analog or digital) on a channel-by channel basis. An analog input can be routed directly to an analog output, bypassing the converters and DSP for true no-latency monitoring. In addition, an internal mixer allows you to mix up to four channels to any output, providing means to monitor a mix coming from a DAW while overdubbing. When used with the AES16 card, complete setups can be saved and recalled on the computer. All I/O is on 25-pin D-subminiature connectors with exception of the word clock in and out (BNCs) and MIDI in and out (5-pin DIN). The Aurora incorporates the low jitter clock technology introduced with the Lynx AES16 card and has an L-Stream expansion slot to accommodate future interfaces.

Prism Sound has been at the top of the converter game for many years, and they've released their most adaptable converter yet, the ADA-8XR. It's available in three basic configurations differing in the configuration of the I/O cards. There's the “plain vanilla” AES/EBU version, a Pro Tools HD version which directly replaces the 192 HD converters, and a version that provides DSD I/O. It's a modular unit, so as your requirements change, you can swap I/O modules.

Also new from Prism at the show is a Firewire I/O module for the ADA-8 series of converters.

<http://www.lynxstudio.com/aurora/index.html>

Mytek, another top shelf converter manufacturer, introduced the new 8X192-ADDA 8 channel 192 kHz A/D and D/A converter in a single rack space. Standard digital I/O is AES/EBU, with ADAT, TDIF, ProTools HD, or Firewire interfaces optional. More than a straight-ahead converter, the 8X192-ADDA includes a high quality headphone amplifier for monitoring either stereo pairs or a unity gain mix of all eight channels. Rather than just a single word clock output, it has several (sorry, I neglected to count and the literature doesn't say), so it can serve as a master clock generator for your system – and the Mytek clock technology is nothing to sneeze at. Another cool feature for those who still don't believe in mixing "in the box" is a built-in eight-channel analog summing bus on the D/A converter outputs.

Two other new related products from Mytek are a multi-output word clock generator, providing outputs from either the internal clock generator or an external source and a sample rate converter that goes up or down, either synchronously based on the incoming sample rate or re-clocking to the internal or an external clock source.

<http://www.mytekdigital.com>

TASCAM introduced the FW-1804, a rack-mounted version of their FW-1884 analog-Firewire interface, but without the 1884's control surface. The FW-1804 provides eight analog inputs and two analog outputs, 8 channels of ADAT Lightpipe I/O, 2 channels of coax S/PDIF I/O, two MIDI inputs and four MIDI outputs. Four of the analog inputs are mic or line on combo XLR/TRS connectors with the remaining four analog inputs being line level only. The mic inputs have phantom power. The clean front panel has input gain controls for all eight analog channels, a 1/4" jack for a guitar input, a headphone output, and switches to send the direct analog signal, the computer playback or a mix of both to the headphone jack.

<http://www.tascam.com/Press/Releases/fw1804.html>

Mackie showed a similar unit, the Onyx 400F Firewire interface incorporating four mic/line inputs on combo XLR-1/4" jacks, incorporating their new Onyx mic preamps. Four additional 1/4" jacks offer line-only inputs, analog outputs are on eight 1/4" jacks, and there are two headphone jacks. The first two mic/line inputs have rear panel insert jacks. A DSP-based internal mixer allows low-latency monitoring of a combination of DAW playback through the Firewire interface and direct inputs. There's also a MIDI in and out, as well as a coax S/PDIF digital input and output and word clock in and out. While I couldn't find it on the blurb and I didn't make a note of it, there must be a high impedance instrument input on there too. The single rack space unit comes bundled with Mackie's Traction recording software.

http://www.mackie.com/press/2004/aes/10282004_400F.html

Consoles, Mixers, and Control Surfaces

A few shows back, I was talking with John (“Father of British EQ™”) Oram of Trident Audio, telling him that people looking for their first decent analog console were going after old Trident 65 models and that he should consider making a console that would fit that segment of the market. It should sell for a price that would allow the buyer to seriously consider a new console rather than one in possibly unknown condition that needed a lot of help before it could become functional. I guess he must have listened, because at this show he introduced the Dream Series of 8 to 40 input consoles. Based on the channel strip from the compact S100 console (3 stereo bus or 6 mono bus outputs) with an upgraded preamp and upgraded components, the Dream Series starts at around \$10K. Decent mic inputs, musical EQ, 5 aux sends per channel, and a clean mixing bus makes it applicable for tracking and mixing whether you’re using a DAW or multitrack recorder. Buy one. Prove that I was right and that there really are customers for this sort of console.

<http://www.oram.co.uk/>

Mackie got into the large format live sound console pretty early but got a bad rap due to some reliability problems. They’re back with a new design, the Onyx 80 series, based on their latest Onyx compact console designs (Onyx preamps and EQ design by Cal Perkins). The Only 80 series is available in 24, 32, 40, and 48 input models with eight aux sends per channel, 8 groups, 4 mute groups, a 10x2 output matrix and built-in power supply.

<http://www.mackie.com>

Speck Electronics has been building rack mount line level mixers for quite some time, and now they have the LiLo (Line In, Line Out) console format mixer designed for the DAW crowd who already has mic preamps, equalizers, and compressors, but wants the comfort and routing conveniences of a real console work surface. The LiLo connects to virtually any analog line level source through fully balanced XLR or 1/4” jacks. Main stereo outputs are transformer balanced, and transformers are optional on other outputs. Maximum level is +28 dBu for plenty of headroom. There are sixteen channel strips with dual inputs for a total of 32 inputs. Those can be assigned to two group busses or the main stereo bus. There are four auxiliary sends and two stereo aux returns. There’s a fully balanced insert send and return on each channel. Monitoring is selectable between the mix, the return from a DAW, the mix plus DAW return, or playback from a 2-track source such as a mixdown recorder. Given the dual inputs (that can go other places than just to the channel input) and the groups, the routing is pretty complex and is worth a peek at the web site for a block diagram.

<http://www.speck.com>

While not brand new, I got my first peek at the AWS-900 console from Solid State Logic. This is a small (though at \$80K+ not small-budget) SSL console with real SSL preamps, EQ, mix bus, and dynamics that's designed as a front end and control surface for a DAW. While you can patch analog DAW outputs into the line inputs and mix through the 900's analog path, it's not really designed to work that way. The fader and pan controls are controllers for a DAW (all the common protocols are supported) so that you record through the SSL analog path and monitor a stereo return coming back from the DAW, mixing on the DAW using the SSL as a control surface. There's no classic SSL automation, but you do get the classic SSL analog preamps and channel processing. Total Recall is a new introduction for the show, but it's not an automation package for the mixer, it's an aid to (manually) resetting the console controls. This is obviously a special purpose mixer for special users, and for sure, it's a real SSL.

<http://www.solid-state-logic.com/music/products/aws900/>

On the other end of the scale, the Studio Projects SP828 is a single rack space eight channel mic preamp with an 8x2 mixer built in. Showed in prototype form at last Winter's NAMM show and promised for mid-year, it's now shipping. The mic inputs have individual direct outputs to go to a DAW or multitrack hard disk recorder, with a separate level and pan control on each channel going to the stereo mix. A solo button allows you to monitor any individual channel. Up to four units can be cascaded for 32 inputs. Three of these racked up with your favorite stand-alone 24-track hard disk recorder would make a great remote recording package. I only wish the inputs of the mixer section could be switched to monitor recorder returns rather than the preamp outputs but for the price, I can't quibble.

<http://www.studioprojects.com/sp828.html>

One of the most talked-about products at the show was the Smartray large format control surface with Arc Technology, the "arc" being a bow-shaped overbridge that represents 96 channels. This is where the individual channels are metered, where solos are selected, and where the range of channels represented by the "fully knobbed" lower control surface is selected. The lower section looks pretty much like a conventional console with concentric controls for the equalizer section, motorized faders, and plenty of knobs and buttons for controlling selected DAW functions. Lotsa channels are controlled by fewer channel strips by dragging a finger across a touch-sensitive strip on the arc. Essentially the controlled channels slide along through a window (24 channel strips wide on the model displayed at the show) so that if you're working on the drums, you can put the drum channels right in front of you. When you're working on the vocals, you can put the vocal channels in front of you. You can subgroup things and lock in those channel strips, as well as bring any individual channel on to the main work surface, adjust it, then tuck it away. It's a pretty cool concept and a text description can't begin to do it justice.

The Smartray is currently configured for several DAWs, curiously not including ProTools. My intrepid reporter question to the rep was "How many people today

have asked you when it was going to support ProTools?" It's a control protocol issue of course, something that Digidesign keeps pretty close to home. They'll be happy to build a ProTools version as soon as Digidesign is sufficiently interested. <http://www.smartav.net>

Recorders

Recorders at an AES show? What a concept! Perhaps the most useful product of the show was the DV-RA1000 stand-alone DVD recorder from TASCAM. The RA1000 records on standard CD or DVD+RW recordable blanks and records standard CD format all the way up to 192 kHz 24-bit resolution in PCM (Broadcast Wave) and DSDIFF formats ready to deliver for high resolution mastering. It's not clear to me that it will make playable DVD-A disks (give it to the client as a high resolution reference copy) but it might. AES/EBU single and double wire inputs as well as balanced XLR and unbalanced RCA analog inputs. While a serious mastering engineer will probably have his own preferred tools, equalization and multi-band compression are available. The really exciting part is that street price should come in at around \$1600, about what a stand-alone CD recorder cost just a few years ago.

<http://www.tascam.com/Press/Releases/dvra1000.html>

For the past several years, TASCAM has offered digital versions of their groundbreaking cassette multitrack Portastudio™ with various degrees of complexity and success, but I think they've finally done it right. The DP-01 and DP-01FX are 8-track integrated recorder/mixers with a built-in 40 GB hard drive with a jog wheel, an LCD for metering and a few other things, and a straightforward analog-style channel strip with real fader, pan, effect send, high, and low EQ controls for each channel. There's even a Record button. Simple enough not to scare away the reluctant cassette user away. There's a dedicated stereo track for internal mixdown and a USB connector provides means for backup to a computer and CD burning. The FX version has a built-in reverb and effect processor as well as two XLR mic inputs with phantom power. The standard version doesn't have the reverb and effects, and has two 1/4" jacks for mic or line inputs. MIDI time code output provides synchronization for DAWs or MIDI sequencers. If you just can't give up your cassette Portastudio, give this one a look.

<http://www.tascam.com/Press/Releases/dp01.html>

Ever since my portable DAT (gee, I hate to say it, but it's a TASCAM) recorder died, I've been looking for a more suitable replacement than the Nomad Jukebox 3 that I've been using in the interim, and I think the Edirol R-4 might be the answer to my prayers. It's a portable 4-channel recorder that runs up to 96 kHz, has real mic preamps with real phantom power XLR inputs, and while not small enough to hide in your underwear for concert stealth recording, it's large and heavy enough not to fall off the table while still being light enough to be easily portable. Audio is stored as WAV files at 16 or 24-bit resolution. Recording modes are mono, stereo, two stereo pairs, or 4 discrete (mono) channels. Inputs

are selectable as mic or line in pairs, appropriate for a pair of mics for ambience plus a stereo board mix.

Built-in DSP provides noise reduction, speech enhancement, compression/de-essing, and equalization either on record or playback. An “oops” buffer captures what happened a few seconds before you dove for the Record button in a panic. Coax S/PDIF digital I/O is provided as well as an analog output for a preset mix of the four channels. A USB port and Compact Flash card slot provide means to transfer audio files in and out. There’s even rudimentary wave file editing (to the resolution of the LCD) for on-site program assembly. And if that isn’t enough, it even has a pair of microphones and speakers built in, and there’s not a mini-jack in sight. Powering is from eight AA batteries or an external power supply.

<http://www.edirol.com/products/info/r4.html>

DSD News

The real interesting news, aside from new gear and software to support the Super Audio CD format is that as of the AES convention, Sony has passed the baton to Gus Skinas, formerly head of Field Operations for the Super Audio Project at Sony Corporation of America. During his tenure at Sony, Gus launched the first production center in the U.S. dedicated to providing a comprehensive array of support services for SA-CD clientele, including recording and tape transfer, mixing, editing, mastering and authoring for SA-CD projects. His new company, The Super Audio Center, LLC will take over support and development of the Sony Sonoma DSD workstation as well as providing production support for SACD products. As if the responsibility to make the SA-CD format survive and flourish, the company introduced a new 24-track version of the Sonoma SACD recorder and editor. While more expensive than your music store DAW, it includes EMM Labs converters, touted as best in the world by some, and a straightforward multitrack user interface for recording and playback including gapless punches, and powerful music-oriented editing. While I’m not ready to trade my Mackie HDR24/96 in on one, I heard that the show special was an *extremely* attractive price.

<http://www.sa-cd.net/shownews.php?news=32>

Weiss introduced the P2D PCM-to-DSD program for Windows that converts PCM files to the native DSDIFF or DSD RAW formats. The input source can be any WAV, AIFF or SND file from 44.1 to 192 kHz, up to 32-bit fixed or floating point format. One to six channels can be specified for input, and input files can be specified as to how they’re treated as surround output channels. Gain and dithering are selectable on a file-by-file basis.

<http://www.weiss.ch/p2d/p2d.html>

Neat Stuff for the Studio

Radial has made innovative direct boxes (DI) for some time now. New to the line this show is the JDX Mic Combiner-Eliminator. It has a phantom powered Class A buffer for the instrument input and incorporates a 0-180° phase control to align the direct signal with a microphone input. This is the second product in the show that I've mentioned which incorporates this feature, so it might be catching on.

<http://www.cabletek.ca/>

Some of our new products are sufficiently complex due to many functions integrated into a single unit that an instruction manual just can't cover it all. TASCAM's 2488 multitrack workstation is one example, and they've solved the problem with a new instructional DVD. The tutorial begins with beginners the fundamental concepts of multitrack recording and mixing from the microphone through the mixer to the recorder. The next section is specific to the 2488 and covers all aspects of recording, editing, and mixing.

<http://www.tascam.com>

DK-Technologies, maker of a set of unique (as well as conventional) metering products has published a wonderful book that's all about metering. Check this one out for more than you ever thought there was to know about the subject. It's not the web page, but for info, e-mail Dave Daniels, the sales manager at dad@dk-technologies.com

Speaking of books, while I didn't have a booth of my own at the AES show, I was carrying around my own new book, The Last Mackie Hard Disk Recorder Manual. It takes over where Mackie's manuals (to which I was a major contributor) left off, describing features of software updates released after publication of the last Mackie manuals for the HDR24/96 and MDR24/96, as well as troubleshooting aids, upgrades, and modifications. Every Mackie HDR or MDR user needs a copy.

<http://www.cafepress.com/mikerivers>

In their continued support of DAW-based recording, Mackie introduced the HMX-54 and -56 headphone mixer systems. Similar to the Oz Audio headphone mixer that's been around for a while, the Mackie headphone mixers have inputs for a main stereo mix plus four additional mono inputs. The musician can make his own custom mix, for instance more vocals, more drums, more bass, or more me (assuming the engineer sends "me" to one of the inputs). The -54 is a rack mount unit with four individually controllable headphone outputs, the -56 is a tabletop unit with six outputs.

http://www.mackie.com/press/2004/aes/10282004_headphone.html

My vote for the coolest gadget of the show goes to Frontier Design's Tranzport. It's a palm-sized controller for a DAW, providing transport control including a jog wheel, track selection and arming, auto punching (there's a footswitch jack for kicking yourself in) and it's wireless, using the WiFi technology. A "base station"

plugs into a USB port on the computer, and the controller can roam around over about a 30 foot range. For DAWs that can support it, the Tranzport data interface is bi-directional so it can display time code position, track names, and even metering. Power is from 4 AA batteries (included!) and price is about \$200. It even has a socket to mount on a mic stand, or take it out to your drum kit and bang away. Frontier Design led the pack a few years back with some innovative audio interface cards but in recent years has been sidetracked developing products for other companies such as TASCAM. It's good to see them back with a product of their own.

<http://www.frontierdesign.com/products/tranzmain.html>

My award for Best Graphics of the Show goes to Yamaha for the Open Deck analog tape simulator add-on effect for the 0-series digital consoles. This is just so cool! You can select among two Studer and one Ampex decks, you can have different tape decks for recording and playback, and the graphics for the plug-in changes to resemble the selected tape deck. Switch between 15 and 30 ips tape speed and the rotating reels change speed. Switch between tape types and the color of the backing of the tape running between the reels changes between black and brown. You can set the EQ and reference fluxivity (called VU calibration here). I had to ask where the head alignment adjustment was, and how to play the calibration tape (but no dice on those). I have no idea what it sounds like, but it's so much fun to play with. Yamaha took the same approach to modeling the functions of a tape deck as Universal Audio did when building the software models of their compressors. Rather than model the device from input to output, they model the individual components, put them in a circuit simulation, and use that to derive the transfer function of the process.

http://www.yamahaproaudio.com/products/mixer/d_mixer/add/master/master.htm

Best brownies of the show go to the Tannoy demo room. Sorry, those aren't on the web site.

Trends and Wrapup

This was one of the best shows in a long time. The venue was good, the weather was nice, and the professional audio industry was well represented. In past years, there was kind of a blur between the pro audio and music store products and neither branch of the industry wanted to miss a promotional opportunity. At this show, some of the former exhibitors that primarily sell through music stores (Alesis, Behringer, and Roland come to mind) were absent, though I saw folks from those companies hanging out, kicking tires, and learning what was going on. I think this a good thing.

192 kHz sampling is definitely with us. Despite some valid technical arguments that it's not necessary and may even be detrimental to the performance of certain present-day components, the demand has been created and products are filling the marketplace. It's a good sign, though, that the serious pro audio

manufacturers are paying attention to the sound and not compromising. While not ready for the show, John La Grou of Millennia Media told me that he was working on a 192 kHz A/D converter option for his mic preamps. His feeling is that while there may not be any audible benefit from the higher sample rate, he's hearing an improvement over his present converter technology most likely because of better this-generation components, and that's the basis for his decision to use them in upcoming products. As long as the designers keep a careful ear to what's happening with their products and don't just go with what the market is asking for, we can feel comfortable with their decisions.

While there will continue to be more people recording for the first time because the buy-in cost for the basic hardware continues to drop, I'm seeing more relatively high priced speakers and microphones and more project-level users showing interest in them, perhaps because of the availability of more cash to spend on the front and back end due to the low cost of the basic recording hardware. This is a good thing as it makes room in the budget for some improvement in places where it counts. Also, with exhibitors of acoustics-in-a-box plying their wares, we may see more attention paid to the recording environment.

Pro audio seems to be in good health. Let's keep it that way.