Audio Engineering Society 143rd Convention Report

October 18-21, 2017
New York, NY

This year's US AES convention was back in New York for, with a few new twists. NAB's (National Association of Broadcasters) main show is in Las Vegas in April, but for many years, they've held a smaller show in New York in the Fall. This year, and for at least next year as well, AES and NAB, while maintaining separate shows and identities, operated concurrently in the Jacob Javits Convention Center, sharing registration to the extent that exhibit badges for one show, which include workshops conducted on the exhibit floor as well as some special events, were honored for the other show. I believe that the intent was to encourage cross-pollenization between the audio and video communities, giving the NAB folks better exposure to pro audio products and techniques, and giving the AES folks better exposure to new opportunities in audio-for-video (primarily, it seems, in immersive video). Neither I, nor the few exhibitors I spoke with on the subject, saw very many NAB badges in the AES exhibit area, and I only did a quick run-through of the NAB exhibit area myself so I didn't see too many AES badges there. But I think the concept has merit, and they'll have another go at it next year.

AES conventions have always been about education, and this year, the show floor presentations were expanded substantially. The Project Studio Expo stage had three full days of programming while a second stage had three full-day Expo programs - Broadcast Audio, Broadway Sound, and Live Sound. The Mix With The Masters booth had hourly mixing talks, demos, and Q&A with more than a dozen well know producers and engineers. Finally, one large room wired for surround sound hosted the Special Events program covering a wide variety of topics throughout the show. So in addition to the technical sessions for the paying customers, Exhibits Plus registrants (free up until nearly opening day) had plenty of opportunities for some schooling when they managed to pry their attention away from the exhibits.

Although the exhibit area was fairly compact (booths, like studios, continue to get smaller and smaller) things were hopping every day. There was plenty to see and, unlike some trade shows, there were knowledgeable people to talk to wherever you stopped to look. Official count was about 15,000 attendees, up 25% from last year's show in Los Angeles.

So, getting down to the nuts and bolts, here are some of the things from the show floor that I found particularly interesting. The usual disclaimer: I'm not writing about everything at the show – I'll leave that to the (dwindling, it seems) publications that have a real staff and budget. I will mention, before you ask, that there were a few regular exhibitors conspicuous by their absence, Royer, Universal Audio, Sound Devices, and Manley Laboratories (though their gear was on display at another booth), and the Harman companies were represented only with a JBL demo room. I suspect that the MIAs may be waiting for the upcoming "AES @ NAMM" to-do coming in January to show off their new products. Hopefully I'll catch up with them there.

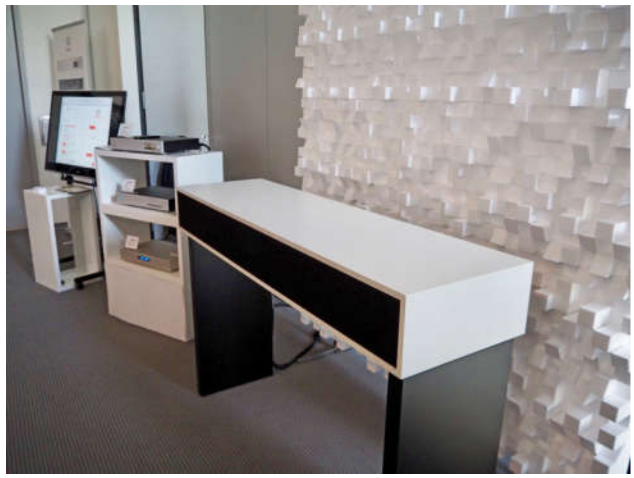
So here's my booth tour, in no particular order.

Speakers – I don't usually pay much attention to speaker exhibits, primarily because there are so many similar small and medium sized monitors that are all about the same but different, and they're best evaluated in a better listening environment than on a show floor. This year, though, there were a couple of things out of the ordinary that drew my attention - a couple of unique designs, and new models from a couple of top-tier speaker manufacturers that are priced for the smaller budgets while maintaining the level of quality for which they earned their fine reputation.

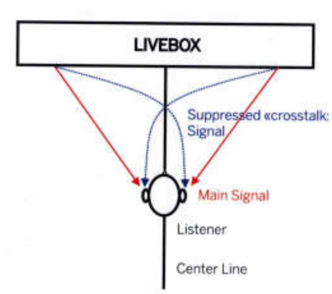
The Kii THREE, in addition to being a fabulous sounding speaker, is unique in that it has very little low frequency radiation from the rear. There's a woofer and tweeter in front, plus a single radiator on the left and right sides, and a pair of radiators in the rear. Extensive DSP is used to steer all of those speakers into something pretty close to a cardioid radiation pattern. The "audiophile pitch" is that you can put them where your domestic partner wants them (out of the way, in a corner or close to a wall) without destroying what you hear. The "studio pitch" is similar in that they can work well in smaller rooms with less fuss about placement and surface treatment.

Bruno Putzeys, a name familiar to those of us who hang out in pro audio forums, is the chief designer. Bruno's most noteworthy work recently has been in the development of the Hypex N-Core Class-D amplifier found in many of today's high-end speakers. He also designed the Grimm Audio LS1 monitor, and in addition to his DSP experience, he had something to do with the design of the Kii drivers as well. The Kii THREE is smaller than I expected it to be, 8 inches wide by 16 inches high and deep, but it sounds much bigger. Input is on an XLR connector and can be either analog or digital, selected by a switch. When using the digital input, a Cat5 cable connects the left and right speakers, with a switch to determine which speaker gets which channel of audio. There's also a remote controller that adds three additional digital inputs including USB and has the typical monitor controller functions – volume, mute, and dim. At about $11,000 for a pair, it's not in the budget for most of us, but considering that it might save you

about half that amount in acoustical treatment if you're starting with a bare room, it's not that bad of a deal.
http://kiiaudio.com
http://www.gracedesign.com/ (US distributor)



PSI Audio, along with digital design whiz Daniel Weiss (another old friend of the pro audio community) and Illusonic (a 3D algorithm designer, among other things) displayed the Livebox. Described by some as "the world's biggest sound bar," it's a loudspeaker system designed to reproduce binaural recordings. If you've had any listening experience with binaural recordings made with two omni mics placed on either side of a dummy (or real) head where the ears go, you know that positioning and depth is very accurate and realistic when listening on headphones, but it sounds odd when played through conventional left/right speakers. This is because there's a lot of mixing of audio going on in the room, whereas with headphones, each ear gets only what's intended for that ear.



Perhaps you've heard a specialized headphone controller such as the SPL Phonitor, which blends some of the left signal into the right and vice versa, simulating in headphones what loudspeakers do in a room. The Livebox does the inverse of this. By using some clever digital signal processing to suppress the crosstalk in the room, each ear hears almost entirely what its corresponding ear-mounted microphone records. It sounds very realistic, and when listening on-center, the soundstage extends well outside the speaker box. This isn't something you'd mix on, at least not yet, but it's an interesting proof-of-concept.

Although there're no details on anyone's web site yet, I suspect that when it becomes a real product, it'll be PSI Audio, since, of the partners on the development team, they're the speaker builders. Those are Weiss A/D and D/A converters in the picture.
http://www.psiaudio.com

Genelec's contribution to the lower cost, high performance small monitor is the "Ones" series, 8331, 8341, and 8351, smallest to largest size respectively. Design concept and construction is similar across the line, all using dual oval cone woofers that radiate from upper and lower ports, and a coaxial mid-range and tweeter that radiates directly into a molded waveguide. They differ in bass extension and maximum SPL but are similar in performance in other respects. Based on published frequency response graphs, the 8341 appears to have high-mid frequency response flatness 30° off axis bettering the others by a hair, but in the demo space at the show, it was pretty hard to tell them apart. Inputs are both analog and AES/EBU digital on XLR connectors, including an AES Thru output for interconnecting speakers. These are Genelec's SAM (Smart Active Monitor) speakers and fit right into a multi-speaker surround setup that can be configured and calibrated using their GLM (Genelec Loudspeaker Manager) computer software.

https://www.genelec.com/studio-monitors/sam-coaxial-studio-monitors
https://www.genelec.com/theones

PMC, another speaker manufacturer known for their high end monitors, introduced the Result 6, a small, two-way active monitor with the nominal 6" woofer rear-loaded by a folded chamber that PMC called the Advanced Transmission Line (ATL) which, in principle, is what we used to call a bass reflex. The back wave passes through a serpentine duct and exits the cabinet at the bottom front, delayed by an amount that puts it pretty closely in phase with the front wave. The tweeter is bracketed by a distinctive finned diffracting surface to widen the high frequency radiation and reduce diffraction from the cabinet edge. Although the crossover was designed using computer modeling, it's analog, with the only controls being a ±10 dB input level trim and the power switch.
http://pmc-speakers.com/products/professional/active/result6

Microphones – There are always a few new mics at every show, and here are a few of the few.

DPA has a new capsule technology for their d:screet and d:fine series of miniature microphones. While the original models are still available, the CORE versions offer lower distortion and greater dynamic range, or more accurately (as

I interpret the specs), 12 to 14 dB more headroom when working in a high SPL environment. The capsule is a tiny square chip that looks a lot like a surface mount resistor. I asked if it was MEMS (Micro Electro Mechanical System) technology, and the guy at the booth said: “I don’t know. We just got it.” There’s nothing really radical here, just a little better mic in the same package for those who can use more headroom, and everybody can use lower distortion. CORE models are physically identical to their standard counterparts but can be identified by the text “core” printed on the case.
http://www.dpamicrophones.com/core

And now for a completely different Core: Core Sound, who for some years has been making the TetraMic, a miniature 4-capsule mic for Ambisonic recording, introduced the OctoMic, an 8-capsule 2nd order mic intended for virtual reality recording projects. While 4-channel Ambisonic recording has been around for quite some time, it’s had limited applications. Now that VR is a hot ticket item, there may be more call for mics like this. You’ll need an 8-channel recorder (the Zoom F8 is a good choice) and DAW plug-ins (Core supplies ‘em) to manipulate the sound image and assign outputs to individual speaker channels.

Sony showed a new line of “high resolution” condenser mics, the C100 multi-pattern large format side address mic and two smaller front-address mics, one cardioid (ECM-100U) and one omni (ECM-100M). The C100 is particularly interesting in that it has two capsules, a large one that covers the full audio range and a smaller one mounted above it that’s high-pass filtered at about 20 kHz to capture sound that we can’t hear directly but that contributes to what we do hear. The two “pencil” mics both use the same capsule as the small one in the C100, but without the high-pass filter, allowing it to cover the full audio range. The C100 is the first one to be released; the others will follow. No release date or price (other than “moderately priced”) but they’ll be worth a listen. Sony hasn’t made a new mic in a while, but they have made some excellent ones in the past.
http://www.sony.com/proaudio (no info there yet)

Samar is known for their excellent ribbon mics, and now, designer Mark Fouxman has gone and done it – made a condenser mic. Actually, it’s the start of a new product line to be called Omni8 Audio. His first product is a three-pattern large diaphragm mic with a unique backplate design. After studying conventional condenser mic construction, Mark concluded that by making a stepped backplate

chamber assembly, he could flatten out the high frequency rise that's characteristic of the standard design without the use of equalization. It was truly a cut-and-try process by machining two different size chambers, varying their spacing with shims, and testing the frequency response until he got the hang of the design. The finished capsule has a single CNC machined piece and he says it sounds wonderful. There was a one hooked up at the show, but it's darn near impossible to be critical of a mic in that environment. He hopes to have some available early next year.
http://samaraudiodesign.com (nothing about the condenser mic yet, but check out the ribbon mics)

Signal Processors (hardware only, thank you) – There are too many software plug-ins for a DAW-avoider like me to recognize what's new, what's important, and what about three people will find really cool, so here are a few hardware signal processing devices that at least four people will find cool.

In 1969, Rupert Neve, at the request of ABC Television in Great Britain, designed a replacement compressor for their consoles, naming it the 2254. With the goal of higher reliability and cooler operation than the original ABC compressors, he came up with the idea of using a germanium diode as the gain control element. In order to improve linearity of the audio passing through the diode, he used four diodes in a bridge configuration, which functioned as a balanced pair of voltage-controlled voltage dividers across both legs of a differential signal path. It's a simple circuit, but one that's tricky to build, requiring the diodes to be accurately matched, and for the input source and output load of the bridge circuit to be accurately balanced.

Balancing was achieved with transformers, which, while expensive, also contributed to the characteristic Neve sound. Studios caught on to the 2254 when compressor distortion was in vogue (the diodes do indeed distort rather badly when driven harder than normal), and the 2254 was modernized a couple of times since the original ABC-TV model to make it more studio-friendly. Rupert Neve Designs recently introduced the model 535 Diode Bridge Compressor, a 500-series module using the diode bridge configuration with improved stability, stepped controls for ratio and response time, a 150 Hz side chain filter, multi-channel linking, a compressed/uncompressed control for "parallel compression" operation, and the classic distortion imparted by the diodes and transformers.
http://rupertneve.com/products/535/

For many years, “Hutch” Hutchison was the principal designer at Manley Labs, but has been free-lancing for the past few years. At this show, he displayed the prototype of his new design for an equalizer called the Finesse, coming soon from A-Designs. It’s a stereo unit intended primarily for mastering, with analog guts and digital controls.

It offers four parametric bands plus high and low frequency filters. All four bands can be either peak/dip or shelving, with conventional modern, vintage, and some unusual passband curve shapes available for both the parametric and high and low pass sections. There’s an assortment of “color” choices – typically transformers and soft-clippers, with a front panel switch to select among three colors that can be chosen as presets for quick comparison. An M/S Mode switch takes stereo in and reconstructs stereo out so you can monitor what you’re doing, but internally, the two EQ channels can be applied independently to the mid (sum) and side (difference) components of the stereo signal.

With a computer connected, you get a graphic user interface plus the ability to control the equalizer directly from a DAW via a plug-in interface. Zoom in on my crummy camera photo above and you’ll get a hint as to what you’ll get when it comes out. Watch for it.
http://www.adesignsaudio.com/

Scott Dorsey, who many of you probably know from on-line forums or his articles in Recording and Audio Express magazines, wanted eight equalizers of a pet vintage design for his remote recording rack, and in order to get the custom inductors that he needed, had to order a hundred of them, so now he has 92 of his Model 506 500-series equalizers for sale. He refers to it the equalizer that we all wanted in 1975, but that nobody could build one quiet enough back then because the components weren’t available. There are three bands with switch-selectable frequencies – a mid-range peak and high and low shelves. Hand-built in the US, with some help from Justin Morse of Roll Music

Also from Scott is the Model 510 Transwormer, a 500-series module that’s simply a transformer, a modern custom-manufactured one that matches the sound of a

vintage one that he likes. There's nothing like the real thing to get that transformer warmth.
http://www.kludgeaudio.com/500/

Digital Converters and Computer Audio Interfaces –Finally, a few manufacturers are getting around to building something that's been requested for some time, a multi-channel computer audio interface that's line level only, no mic preamps, for the folks who want to insert some analog processing into a DAW session. Focusrite has one, TASCAM has one coming soon, and I'm sure I saw at least one other but I can't remember from whom – darn these manufacturers who come to a show with new products and no literature to remind me of what I saw!

But first up is Crane Song, with the HEDD Quantum 2-channel A/D and D/A converter, which I think is going to replace the HEDD 192. In the pursuit of lower clock jitter, which is a big deal with designer Dave Hill, the Quantum uses the same clock as in his Avocet IIA D/A converter, applying it to both the A/D and D/A converters. The Quantum has six word clock outputs so it can serve as the master clock for a system with multiple digital devices. Other new features are TosLink optical digital inputs and outputs in addition to the AES/EBU and S/PDIF digital I/O. The new Quantum retains the analog tape, triode, and pentode simulation that makes it a HEDD (Harmonically Enhanced Digital Device).
http://cranesong.com/Hedd_Quantum.html

Antelope Audio showed their new Discrete 8 and Discrete 4 audio interfaces with both Thunderbolt and, for the rest of us, USB computer connectivity. They're functionally similar, different only in the I/O port count. The model numbers refer to the number of discrete solid state mic preamps (two connectors on the front, the rest on the rear) and analog I/O ports. The 8 counts 26 inputs and 32 outputs including S/PDIF and 2 ADAT optical inputs and outputs, plus 2 dedicated monitor outputs, 2 headphone outputs and 2 re-amp outputs. The 4 has one ADAT optical I/O pair and 4 headphone outputs.

Both models are equipped with Antelope's suite of effects that can be assigned either to the recording path or to only the monitor path. The Discretes are the first on Antelope's interface products to include microphone modeling. This can be applied to any mic to make it sound like something else, but it's more predictable

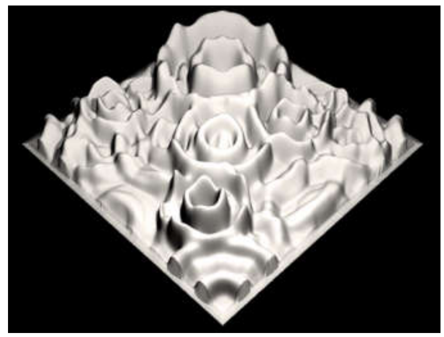
when used with their optional Verge (small diaphragm) and Edge (large diaphragm) mics. In addition to computer control for setup, routing, and mixing, there's an iOS and Android mobile app coming. There are interface + mic bundling deals on the books now including a free mic for pre-orders before October 31.
http://en.antelopeaudio.com/products/discrete-8/
http://en.antelopeaudio.com/products/discrete-4/

The Focusrite RedNet X2P is a desktop form factor 2-in by 2-out audio interface with two Red Evolution mic preamps. Mic/Line inputs are on XLR-TRS combo jacks, main analog output with a hefty +24 dBu maximum output level is on XLRs, and a front panel ¼" TRS jack with an independent volume control provides a headphone output. Computer connection is via Dante, with power supplied either through the Ethernet cable from a PoE (Power over Ethernet) router or from an external 12v power supply. While everything can be controlled remotely over the Dante network, real knobs and buttons are provided for input gains, phantom power, polarity reverse, low-cut filter, and to engage Focusrite's "Air" character. A single knob adjusts the balance between the input signal and DAW return for no-latency input monitoring when overdubbing. Normally this input/playback mix feeds only the headphone jack, but there's a button to switch the main outputs from DAW return to the Input mix. An OLED panel provides metering, input gain indication, and status of various things.

Also from Focusrite comes the Red 16LINE with a multitude of analog and digital inputs and outputs. There are 16 analog inputs and outputs for your outboard gear plus two ADAT optical inputs and outputs for another 16 channels of audio and – wait! There's More! – 32 channels of audio coming and going over Dante. There's a pair of Pro Tools HD Digilink connectors plus a pair of Thunderbolt connectors for DAW connectivity. All of the outputs are hot all the time, so you can have it wired up to two different DAW systems as well as have its inputs and outputs appear on a Dante network. And if that's not enough, there's a pair of Red mic preamps just for good measure.

Focusrite recently announced the establishment of the Focusrite Pro division to handle marketing and support for the Dante RedNet products, Red interfaces, and the classic ISA preamps. The Scarlett, Clarett, and iOS product lines are still alive and well and are being handled by the main division.
http://pro.focusrite.com/category/audiooverip/item/rednet-x2p
http://pro.focusrite.com/category/audio-interfaces/item/red-16line

Other Stuff – Things that are interesting enough to report on but you don't have to plug in or turn on.



Acoustics First makes a full line of acoustic treatments and this year they had a new diffuser panel that looks so cool I just had to mention it. The ArtDiffusor D looks a bit like the surface of another planet, but actually, it's a quadratic diffuser that follows the same equations as the conventional style made up of blocks of different lengths. Those disburse reflections in one plane, usually horizontal, while this one diffuses in two planes for near-hemispherical diffusion.
http://www.acousticsfirst.com/diffuser-art-diffusor-model-d.htm

Audio resistors might or might not fall into the realm of pyramid speakers stands and hardwood control knobs, but Susumu was displaying thin film resistors that they claim have lower noise than other resistors, which may (their "may") improve audio quality. I don't recommend that you change out all your resistors, but some credible folks in the audio business have reported positive results. IC maker THAT Corp uses Susumu thin film surface mount resistors in one of their development boards since they tested better than other brands. Also, Dave Hill of Crane Song reported that he found, after considerable testing of surface mount resistors, that Susumu's were the best performers among half a dozen brands he tried.

Any resistor of a given value will have the same thermal (Johnson) noise, but there's also modulation noise, sometimes called "excess noise" that's related to the granularity of the resistance material that varies with current flow. This noise is generally low enough to be masked by audio, but when Dave Hill says he can measure and hear a difference, maybe they've got something here. If you're a do-it-yourselfer, Digi-Key and Mouser stock the brand.
http://www.susumu.co.jp/usa/product/category.php?cid=23?cid=23

The Technical Sessions -

I never seem to get to as many of these as I want to, and more often than not, I've put two or even three on my to-do list that are in different places at the same time. The technical program is laid out in tracks, and I found myself on the Archiving and Preservation track most of the time I was “downstairs.” There’s really some fascinating stuff going on among people who are trying to pull the best quality audio off of obsolete formats. Most grooved media (cylinders and phonograph records) is still played mechanically, but lots of R & D has gone into improving the mechanics of pulling the grooves past a stylus.

A (maybe *the*) chemist with the Library of Congress Audio Visual Conservation group described his work in learning why cylinders in the Library’s collection that were whole when they went into storage were cracked when the came out to be played. He analyzed scrapings from cylinders to create new samples of the original material, since nobody has the original formulas. He made some mock-up cylinders based on formulas that he derived so he could test the effects of moisture, heat, cold, and cleaning procedures that they used. They store cylinders in a cold room, and he found that some of his formulas were prone to cracking when brought up to room temperature. Looking at what went into the soup, he discovered that one component of the compound had an unusually high coefficient of thermal expansion compared to the other compounds, and it was this difference in thermal expansion that caused them to crack. The solution? Put the cold cylinders in a cooler and let them come up to temperature slowly.

Steve Lampen of Belden told of the research that went into designing what they believe to be the ultimate audio cable, and of the engineer who they let take on the project. First they had to figure out what makes a cable better for audio – basically the right combination of resistance, capacitance, and inductance, and then figure out how to make a cable that has the best combination of those parameters that’s robust enough to put into real world service. It turned out to be a star quad configuration of relatively thin uninsulated conductors (24 gauge, I believe) supported by a custom Teflon molded core and shielded with a double braid. Steve’s very generous with samples, but because of the construction of this cable, attaching XLR connectors is difficult enough so that they want to assemble them in house so as to get it right. He’ll lend you one or two if you think you’re audiophile enough to hear if this is really a better cable than what you’re using, but you’ll have to give it back.

A couple of interesting presentations I attended were in the historical track. Tom Fine’s talk about the history of stereophony and Dan Mortensten’s history of the CBS studios on 30th Street in New York City were great reminders of how we’ve evolved. Paul Blakemore, an old friend from NPR and now mastering engineer for Concord records demonstrated the Soundstream recorder that he used to play Telarc masters for new releases. The 1812 Overture played directly off the digital tape over large PMC speakers never sounded so good. Since the Soundstream had response down to DC, if you were quick, you could count the approximate 6 Hz of the mortar firing by watching the woofer cone.

There were dozens of tutorials, both generic and product specific. I wish the show were two weeks long so I could get to more of those sessions, but I couldn't afford the hotel bill.

There's no NAB show report because I really only did one quick walk-through of the show floor and didn't even look at their tutorial program. I was actually looking for one of the tool distributors that usually exhibits at that show, but he wasn't there.

Well, that's all for this year. The AES and NAB shows will be in New York again next year. Mark your calendar: October 17-20, 2018.