

Audio Engineering Society 129th Convention
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I really enjoy San Francisco in the Fall. It's not too cold, it's not too rainy, the food is great, and the SF AES chapter is still hanging on to the US West Coast show. Every show is a little different, but this one was a little different in a couple of ways. For one thing, the exhibit floor was only open for three of the four days. Thursday was devoted entirely to the other side of the convention, the technical program, which was great for me as I usually spend all day every day kicking tires and rarely have time to squeeze in one paper or workshop.

More about the technical sessions later on, but since you're probably reading this article to learn about the new and cool gear, here goes:

Recorders

TASCAM HS-2000 & 4000 – TASCAM introduced a pair of rack mount flash memory recorders, two and four tracks respectively. These are targeted to the broadcast market, but would be a good choice for a church install where you need to record the service and would rather not fool with a computer or portable recorder. Up to 96 kHz 24-bit recording in broadcast wave format, SMPTE time code, balanced XLR inputs and outputs. A color touch screen provides menu navigation, metering, and time scale. Two memory card slots provide your choice of redundant recording, sequential recording (starts on the second card when the first one is full), or making a backup copy. For playback during a show, files can be arranged in a playlist, and the playlist can be edited during playback to accommodate changes. The 4-track HS-4000 can operate as two independent 2-track recorders, though I neglected to ask if the setup parameters could be different for the two recordings, for example recording an entire church service in low resolution while making a high resolution recording of the band to sync up to video.



The four tracks can be mixed to stereo for headphone monitoring. These are quite industrial looking, and appear to be quite

bulletproof. The RC-HS32PD is an optional remote for controlling up to two recorders. http://tascam.com/products/hs_series/

TASCAM BD-R2000 – This high resolution video recorder records to an internal 500 GB hard drive, SD/HC flash memory card, or direct to a standard recordable DVD or Blu-Ray disk. Recording formats are BDAV, BDMV (HDMV), DVD-Video, DVD-VR, and JPEG. Encoding modes are MPEG-2 TS, H.264 (MPEG-4 AVC), MPEG-2 PS (video) and Dolby Digital), and Linear PCM (audio). Inputs are S-Video or composite video, iLink (4-pin Firewire) DV or HDV, and can load files via USB2.0 or from a memory card. Video outputs are HDMI, composite, component, and S-Video analog. Analog audio I/O is on (bleah!) RCA jacks with digital audio output provided on an optical Toslink connector.

Simple editing (split, delete, name the disk, create a playlist) and basic DVD and Blu-Ray menu authoring can be performed on program material recorded on the internal hard drive prior to



burning to optical media. This looks like a good bet for a corporate installation, meeting room, church, or theater, or for live concert recording.

<http://tascam.com/product/bd-r2000/>

JoeCo Blackbox Player – The JoeCo 24-track hard disk recorder, a 1 rack space unit that records to an external USB disk drive, has been around for about a year now, and has become quite popular on tours for a no-hassle, reliable means of recording shows for future use. Apparently some tours and shows have been using one for playback of prerecorded material, so JoeCo is now offering a playback-only version to meet this need. With 24 tracks available, you can have stems for the live PA mix plus custom cue tracks for individual monitor mixes, for example with and without a count-off, with a click, different blends. WAV files can be pre-loaded on to a hard drive or USB flash drive and can be organized with a playlist. Playback can be triggered from the front panel, foot switch, or by MIDI. A just-in-case feature allows a second unit to run in sync with the first, and with the output of the primary player looped through the backup player, playback will switch seamlessly over to the backup unit. I expect that people have been doing multitrack show playback with a computer for a while now, but for my dollar, any day you can eliminate a computer in a live sound situation is a better day.

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



TASCAM DR-03 – TASCAM’s entry into the hundred buck stereo hand held recorder market. Recording is up to 48 kHz, 24-bit stereo and down to 32 kbps mono MP3. A pair of omni mics is built in, and a mini phone jack is available for input from external mic or line sources. A low cut filter, limiter and automatic or manual record level can be selected from a fairly simple menu. Recording is to a micro SD card. Like TASCAM’s other handheld recorders, the DR-03 offers a few musician-oriented features: variable speed at constant pitch, loop play, and a built-in speaker. <http://tascam.com/product/dr-03/overview/>

In other recorder news from TASCAM – The DR680 8-track portable recorder has a firmware update. Version 1.20 offers M-S decoding on a pair of inputs, with the decoding selectable for monitoring (recording the mid and side mics directly) or for decoded left/right stereo recording. The X-48 48-track hard disk recorder has a firmware update that provides several user interface improvements, better file management, SMPTE time code midnight crossover.

Korg MR-2 – This hand held recorder has been out for several months now, but this is was my first opportunity to see one in the flesh. It’s an update to their original MR-1, which had the distinction of being the first sub-\$1,000 device capable of recording DSD formats as well as the more common PCM and MP3. Even though the future of the SACD is kind of shaky, Korg is still pushing DSD (Direct Stream Digital) as a “future proof” (can I say “future resistant?”) audio data storage format and they maintain a broad line of DSD-capable recorders, including the new MR-2. When the MR-1 was introduced, 1 GB memory cards were still pretty expensive and, for the wealthy, the capacity topped out at 4 GB. For this reason, Korg chose an internal 20 GB hard drive as the recording medium in the interest of providing ample recording time before it was necessary to unload. Now that 4 GB SDHD cards are about as cheap as cassettes (and record a lot more time), Korg has switched to solid state media, improving the reliability and battery life, as well as reducing the size and weight. The MR-2 has built-in mics (the MR-1 had an external mic, which I liked, but it was on a short leash, which I didn’t care for) which can be rotated to point



toward the front, the rear, or endwise. There are 40 presets (10 user-configurable and storable) which include such things as input sensitivity, limiter, and low-cut filter. Power is from removable AA cells rather than the MR-1's fixed internal battery, but if you're using rechargeable cells, they have to be recharged outside the recorder, they can't be recharged via the USB port. Wait! There's more! A chromatic instrument tuner, a Korg signature product for many years.

<http://www.korg.com/product.aspx?pd=575>

But the nice guy at the Korg booth snuck one past me. Though we were talking about DSD and its applications and limitations, he neglected to tell me that across the street in a hotel suite, they were demonstrating a PC interface box and software for multitrack DSD recording. I only heard about it after the show so I can't tell you all about it (maybe I'm not supposed to) but I picked up a few details. Most important is that it's not a product yet, so don't go knocking on your dealer's door, it's more like a proof of concept or capability, but I didn't hear that it crashed, and that's a good sign. The number of tracks that can be simultaneously recorded is of course limited by the hardware (I think right now it's only two), but the software is like what you'd expect from any DAW, with the number of tracks that can be played and mixed limited only by the computer's horsepower. There's a console-like display with faders and knobs, even an EQ section, and recorded tracks can be edited, presumably while remaining in the DSD format. The only other system that can do this is Sonoma, developed originally by Sony and now sold and maintained by Super Audio Center.

I own a Korg MR-1000 (the grown-up version of the MR-1 and I can tell you that DSD really sounds great. I can also tell you that I rarely use it in that mode because unless I convert it to a PCM format I can only listen to playback from the recorder itself. There's not yet a suitable authoring program that can make a playable SACD disk (it's an anti-piracy thing). I know a couple of mastering engineers who are now capable of taking in projects on disks made with TASCAM's DSD recorder, so maybe it'll come around. It's one of those things that, to its fans, sounds "more analog" in a good way.

Another piece of news from Korg is that their AudioGate program, a player/editor and converter between just about every audio file format including DSD, will become openly available soon. It's currently included with the Korg DSD recorders, but when first installed, it requires you to connect your recorder to the computer's USB port to prove that you have one. This requirement will go away. Official release date is November 15, so check the web site: <http://korg.com/>

Microphones

Not a lot of cheap mics at this show other than those at the bottom end of a broad line. One thing that I have to comment on though, is that we're starting to see some pretty garishly ugly mics. Satin off-silver is traditional, as is matte black. Then we started to see chrome and gold-plated bodies. Now we have finishes like car bodies of the 60s – metalflake red, iridescent green, big, bold white lettering running the whole length of the body. Big honkin' badges! What's the deal here? Are they making these for bling-encrusted hip-hop singers? I'm sure that none of this affects the sound, but gimme my basic black and gray, please.

BeesNeez – This is a newcomer from Australia. The range includes both solid state and tube head amplifiers, spanning the range from a small diaphragm transformerless FET mic through a large diaphragm tube design. All capsules seem to be kin to the Neumann K47 capsule and are made in house. These aren't inexpensive, ranging from just under \$1,000 for the transformerless Lulu FET to close to \$5,000 for the T1 U47 tribute. They all have odd (for a microphone) names like Arabella, James, Mahalia, or Phelicity.



Despite the big ugly nameplate that reminds me a bit of the trophy in an announcing contest, the buzz is that these are very good sounding mics. High quality parts all around, Cinemag transformers, old stock tubes, good finish and build quality. These are replicas of originals, but some models are tailored to have the characteristics of well known mics such as the Neumann U47 and Telefunken C12. From listening to people talk into them at the show, I could tell differences in the high end and proximity effect. If I had to choose one as an all-around mic, it would probably be the multi-pattern Arabella at a bit south of \$3,000.

<http://www.beesneezmicrophones.com.au/>

AEA KU4 – AEA's Wes Dooley has been studying and repairing ribbon microphones long before he actually took on the project of building them, but with half a dozen new mics in the catalog now, that project has proven to be a real success. The latest is a modern realization of one of the rarest and the most expensive of the RCA condenser mics, the KU3A, also known by its manufacturing number M-10001, of which there were fewer than 600 made. Unlike the bi-directional polar pattern of a conventional ribbon mic, the KU3A has a super cardioid polar pattern, with a smooth and broad lobe in the front similar to the RCA 44, but with good rejection to the rear. Bi-directional mics by nature (and hence ribbons) have the most pronounced proximity effect, but the AEA KU4 has



similar proximity effect to that of a well behaved cardioid dynamic mic. While there's a bit of a dip between 10 and 20 kHz, it has usable frequency response beyond 20 kHz, with very smooth response below 10 kHz. Merging old and new technology, the KU4 uses a ribbon made from new old-stock RCA material with a new custom designed transformer and neodymium magnets for a higher output sensitivity (looking at the numbers, I'd guess it's about 3 dB hotter than an SM57). <http://www.ribbonmics.com/>

Mojave Audio was showing two new mics. The MA-101fet is a small diaphragm solid state condenser mic which combines features of two popular Mojave mics, the interchangeable cardioid and omni capsules from the MA-100 with electronics from the MA-201fet. A new feature is a switchable 15 dB pad, making it suitable for recording loud electric guitars or drums as well as quiet acoustic instruments. Also new is the MA-300, a three pattern version of the MA-200 cardioid tube condenser mic.



<http://www.mojaveaudio.com>

Shure introduced the Beta 181, a compact pencil-style body with interchangeable side-address heads. Available in omni, cardioid, super cardioid, and bi-directional, Shure anticipates that this will be popular for snaking into tight locations. Priced at \$499 with one capsule, additional capsules are available for \$249 each. They swap out quickly with a turn of a positive locking ring.



Half-new is the Beta 98AMP, a Beta 98 miniature cardioid head with an integral short gooseneck extension. It will be supplied with a nifty universal drum mount.



In other news, Shure has re-introduced the A27M stereo mount, a favorite of mine, nicknamed "the Shure billy club." It seems that someone dug one out of a closet and, with the rising interest in stereo miking, decided its time had come again. The one on display looked pretty worn, but they promised they'd be making new ones again.

Computer Interfaces and Converters

TASCAM US-1800 TASCAM has produced a number of multi-channel USB audio interfaces over the years and this is the latest in the series. It's a 6 in by 4 out USB2 interface with 8 XLR mic inputs with phantom power (switchable in two groups of four channels), 6 TRS line inputs, stereo coax S/PDIF in and out, plus



MIDI in/out. Two of the 1/4" line input jacks are on the front panel and can be switched to be instrument direct inputs. A stereo mix returned from the computer can be mixed with what appears to be a fixed mix of inputs using an Input/Computer blend knob on the front panel. This is true zero-latency input monitoring, a good thing in my book, and something that TASCAM has stuck with since their earliest USB interfaces. Outputs 1-2 feed both the front panel headphone jack and the rear panel Monitor jacks, with independent volume controls for each. Outputs 3-4 are independent. I didn't think to ask at the booth, but it's not clear from the block diagram just what comes out the S/PDIF output. Possibly a mix of the odd and even outputs. Sample rates run from 44.1 to 96 kHz, resolution is 16- or 24-bit. At \$300, TASCAM claims that this is more I/O than any device in its class. Could be. That's a lot of utility for three hundred bucks. <http://tascam.com/product/us-1800/overview/>

Sound Devices USBPre 2 – Sound Devices is well known for their high quality gear that's designed for tough field use. When it came time to update their USB stereo preamp, they didn't let a good thing go to waste and borrowed the mic



preamps and A/D converters from their well respected 700 series field recorders. The USBPre 2 puts a stereo pair of mic preamps or line inputs as well as S/PDIF input I/O (both

coax and Toslink) into a compact package and spits out USB to a computer. It's a USB class compliant device, so when plugged into a Windows system for the first time, it will install the appropriate driver. In this mode the sample rate is limited to 48 kHz (Core Audio on a Mac OS 10/5.8 and higher allows sample rates up to 192 kHz), however there's a Windows ASIO driver which I assume will allow the full range of sample rates.

There's a large front panel meter with sufficient resolution near full scale so that even I won't complain about it (inadequate metering is one of my biggest complaints about digital input devices). Monitoring is quite flexible, as is

mono/stereo configuration – plug a mic into one channel and you can monitor it centered in mono at the press of a button). There's a stereo return from the computer so you can monitor playback or mix the playback with the input for overdubbing with zero latency input monitoring. With many configuration options set by DIP switches on the side panel, the clear diagram on the bottom of the case showing each switch setting is a blessing. Sample rates run from 8 to 192 kHz, 8 to 24 bits. Power is from the USB port (no batteries or wall warts) with a soft start, thoughtfully limiting the inrush current when hot-plugged to 100 mA. This is a USB specification that's sometimes ignored, but it's adherence is an example of how Sound Devices keeps their hardware bulletproof.

<http://www.sounddevices.com/products/usbpre2.htm>

Lavry Black AD-11 A/D converter – A/D conversion is really important since your recording will never be any better than your converter, and Dan Lavry's modular designs have been long been



considered among the best. The Black series are half-rack sized converters that retain the high quality engineering and clean sound at lower cost than the modular Blue or full rack width Gold series. The AD-11 is an update to the AD-10, offering a pair of clean mic preamps and A/D conversion with USB, AES/EBU (XLR) and S/PDIF coax outputs. The mic preamp offers switchable phantom power on individual channels and the line input sensitivity is switchable to accommodate sources with a nominal output level of +4 dBu or -10 dBV. A "soft saturation" features offers Lavry's interpretation of a tape-like sound, and the meters have wide range and good resolution near full scale. The unit on display was a pre-production model, and I asked why I couldn't find a headphone output or any analog output at all. When I'm recording, I want to be able to hear what's going into the mic, and it seems the only way you can do that with this converter is to monitor the digital output through a D/A converter. It's no problem if you also have the companion DA-11 or another high quality D/A converter, but if you're putting together a portable recording setup, needing another box, or settling for the headphone output of the computer's sound card for monitoring, seems like a significant sacrifice. To Lavry's credit, he indeed recognized the need and considered including a headphone jack, but didn't want one that wasn't up to the quality of the rest of the signal path. Because of a crowded box and the power requirements of a high quality amplifier with enough power to drive headphones, he just couldn't fit it in.

http://www.lavryengineering.com/productspage_pro_ad11.html

JCF AD8 8-Channel A/D Converter – JCF (Josh Florian) has a few interesting products and a clever motto for his company: "Ruining audio less since 2004."

He seems to be obsessed with the sound of analog tape decks and has, for example, adapted the Ampex 351 output stage circuitry to one of his D/A converters. At this show, he introduced the AD8 (not to be confused with his DA8-V 8-channel tube D/A converter), an 8-channel A/D converter with some DSP juju inside that he claims makes it sound just like analog tape without adding any noise, 2nd or 3rd harmonic distortion. It definitely does something, though I couldn't say what. I'm not one who misses the sound of analog tape so I'm not sure I want something in my signal chain that intentionally changes the sound of what's going into it, but then there are those who believe that a "normal" A/D converter changes the sound. Anyway, it got my attention, and the attention of several others I spoke with at the show. Stay tuned for further information on the web site:

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

Millenia Media AD-596. This is an 8-channel A/D converter in the API 500 series modular format, and, as far as we know at this point, the first digital 500-series module. Millenia has always offered top quality products designed by both engineering and listening, so there's no reason to suspect that this one won't live up to the company's reputation. It's 24-bit and supports sample rates from 44.1 to 48 kHz. Clocking is internal or external with front panel BNC word clock input and output connectors.

Since a 500-series rack could sensibly include one of these converter modules and a few mic preamps, how they co-exist in the same case, sharing the same power supply, is of legitimate concern. John LaGrou said that he took great care and plenty of measurements to assure that radiated EMI from the converter is adequately low, and he didn't hear any evidence of contamination in his own 500-series mic preamp adjacent to a converter in his own API brand rack, definitely a good start. But the 500 series rack was designed for API modules, which are all analog. The rack design spec (for the genuine article) calls for jumpers to connect (or not) audio ground, power ground, and chassis ground, but that may or may not be adequate for keeping analog and digital ground currents separate. Now that there are many manufacturers of a "standard" rack, we'll need some time and experience to know how universal this concept is, or if the rack design needs further work and more detailed specification.

<http://www.mil-media.com/AD-596.html#ph>



Consoles – In again!

I'm really pleased to report the upswing in mixing consoles and DAW control surfaces that include console functions such as monitor routing and level control, talkback, and even some mic preamps for good measure. My studio is still based around an analog console and I sorely miss it when forced to work with nothing but a computer and audio I/O interface. Since so many studios today are set up by people who have never used a console before, I'm happy to see that they're beginning to realize the workflow benefits.

Allen & Heath GS-R24 – Not long ago, A&H released their first recording console in several years, the ZED-R16. I had one in for review and really liked it, but decided that it wasn't quite ready to replace my ancient (but not really "vintage") Soundcraft 600. It was missing a few features that I considered essential. I spent some time during the review and afterward e-mailing designer Mike Griffin, pitching some things that I thought were essential. Last NAMM show, he showed me some photos of his new creation, and now the GS-R24 is here and it looks mighty good. 24 mic/line input channels with analog direct



outputs, 4-band EQ with sweepable mid bands, 6 auxiliary sends, 4 subgroup buses, and optional digital I/O. The first digital I/O card available now is 32 streams in and out on both Firewire and ADAT optical ports. The card includes word clock in and out, something lacking on the ZED-R16. There's also a meter bridge, another thing that I consider essential. The mic preamps and

EQ on the ZED-R16 that I used for a few months sounded excellent so there's no reason to question them here. I suspect that they're essentially the same.

Routing is very similar to the ZED-R16. Digital outputs are always post-Insert, with switches that select whether the signal feeding the A/D converter is picked off before or after the EQ section. Similarly the digital return to the channels can be either through the full channel path or directly to the fader. With this setup, whether using a computer with a Firewire connection or a stand-alone multitrack recorder fed from the ADAT optical or analog direct outputs, it's possible to operate just like an in-line console with track playback always available to the monitor path, as well as direct analog monitoring of the live inputs.

The faders can be switched to become MIDI controllers. In this mode, the analog signal path from channel input to main output has fixed gain so that channels can still be summed through the analog channel strips (with or without EQ) but levels can be controlled in the DAW via from the faders. With the optional motorized faders, in addition to writing volume control tracks, the faders will follow an automation track in the sequencer, allowing full analog mixing including

automated fader movements. If you choose to use the DAW to control levels, the faders will still follow changes so you'll know what's going on in the mix. There's a transport control section with dedicated conventional "tape deck" buttons, a jog wheel, two dedicated MIDI faders, and a handful of buttons and rotary encoders for whatever functions you choose to assign to them.

Mike Griffin is really proud of the tube stage that he designed for a couple of other A&H consoles, so two of them are included with the GS-R24. These can be inserted in the mic input path, or used as a high impedance instrument input. They can also go in line with the main mix bus for some final tube warming, and their inputs and outputs are accessible through the digital I/O, allowing you to pass a track through a tube and back out to the DAW.

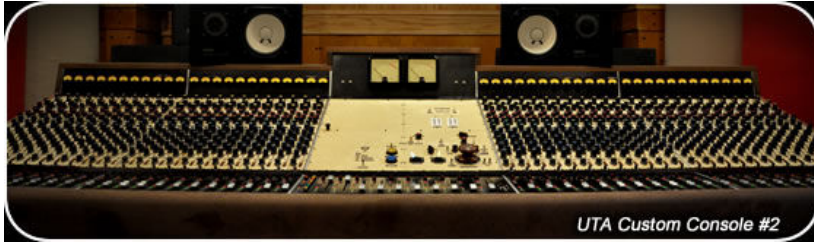
This isn't a console to go with your tabletop studio, but it would be right at home in a medium sized facility, whether it's based around a DAW or analog recorder. With the Firewire/ADAT card and moving faders, it'll probably come in at around \$9,000. <http://www.allen-heath.co.uk/US/DisplayProduct.asp?pview=115>

Trident 82 – This is another new serious studio console, the first large format Trident console designed by Malcom Toft since he sold Trident Studio about 25 years ago. Sparked by the success of Malcom's ATB console, a modernized version of the Trident 80B introduced a couple of years ago, PMI, the company who markets Toft's designs (and also the popular Studio Projects microphones)



encouraged him to build a large format modular console just like they used to make, and the Trident 82 is the result. It's all analog, though there may eventually be a digital I/O option, but really, when putting together a system based on a console of this size, you're probably going to want to choose your own A/D and D/A converters rather than take what the console manufacturer offers, if nothing but on general principles. There's far more than I can describe in this summary report so go to the web site and drool. There was a fully loaded 32 channel frame on display with 16 subgroup buses, 8 auxiliary sends (two pairs are stereo) and both in-line and split monitoring sections for an additional 24 line inputs available on mixdown. Visit the web site a little later (nothing there about this console yet) for all the details. In the mean time, you can drool over this photo from Harmony Central, snapped at the show. <http://pmiaudio.com>

UnderToneAudio (UTA) – I suspect that nobody reading this report will be buying a UTA console soon, but it's interesting enough to deserve mention. This is a custom built consoles, designed meet all the requirements of the designer/engineer who hadn't been completely happy with any console he's used. The complete signal path, including the EQ section, is based on Class A amplifiers, with a tube in the mix bus amplifier. The design philosophy here was



that the console's equalizers should be flexible enough so that it shouldn't be necessary to go to outboards to get the desired sound on any

channel. The UTA has the look of a classic studio-built console, but it's a nice job, with a mix of modern features and some retro knobs and meters. Particularly noteworthy is the top panel material, a foamed aluminum that's acoustically transparent for minimizing reflections from meter-bridge mounted monitors. He says that it really really works, it has a patent pending, and even a name – ATWS (Acoustically Transparent Work Surface).



<http://undertoneaudio.com/>

Solid State Logic showed a major update to their AWS-900 series. To refresh your memory, the original AWS-900 was a combination of a DAW control surface that looks like three HUIs, and a 24-channel Ultra Analog tracking console with all the standard studio functions and SSL's Total Recall for quick recall of a previous mix. The new AWS-948 accommodates 48 channels and offers the functional equivalent of a 24-channel in-line monitor console when using the second channel path as a DAW return. In addition to the moving faders being simply part of the setup recall, they'll follow a MIDI automation track from the DAW, providing an automated analog mix or a positive indication of the current DAW fader positions. The AWS-924 has similar updates but without the dual-input channels. <http://www.solid-state-logic.com/music/aws/>

Also new from SSL is the Nucleus, a DAW control surface with two mic/line inputs and a variety of monitoring outputs. The Nucleus uses both HUI and MCU control protocols as well as MIDI continuous controller data, which should cover just about any DAW. Two banks of eight faders each plus a section of dedicated hardware controls (buttons and rotary encoders) can be configured to do whatever the user wants. A 2 channel USB I/O interface with mic/line inputs and stereo returns to the Nucleus monitor section. These can be mixed with the inputs for overdubbing. The mic preamps are the same SuperAnalogue design used in the AWS, Duality, and X-Rack lines. There's a plethora of other analog and digital inputs and outputs which should take care of most outboard devices.

The faders can be switched among three layers for controlling lots of tracks, or the whole shebang can be switched to control three different DAWs.



MIDI connection is through Ethernet (Euphonix pioneered this with their Artist series controls, to which the Nucleus, at least from a distance, is similar). There's a set of V-Pots which can be assigned to frequently used DAW editing functions. A digital scribble strip below each pot indicates

its function. Now that Pro Tools 9 is capable of working with non-Avid audio hardware, combining it with a Nucleus could make for a very flexible production workstation as long as you didn't need multiple inputs and could work with its two SuperAnalogue inputs.

<http://www.solid-state-logic.com/music/nucleus/index.asp>

The 500 Club

The craze for modules that fit the API 500 series rack or "lunchbox" hasn't slacked off a bit. Time, space and, frankly, lack of interest prohibits me from reporting on every new module introduced for this format. If you can dream it, you can probably find that someone has made it. I've already mentioned the Millennia A/D converter and I have one more that's too cool not to mention, but first:

API - One of the most important contributions this year comes from API themselves with an update to their Lunchbox. The rack and internal bus structure remains the same as always for compatibility but the power supply has been beefed up to provide 215 mA per slot (sorry, I don't know what that's up from). In



addition to the six pairs of XLR input and output connectors, a pair of DB-25 connectors (TASCAM pinout) provides balanced analog input and output connections for each slot. Since the lunchbox has slots for six modules

and the DB-25 handles eight pairs, two extra pairs of XLR in/out are wired to channels 7-8 of the DB-25s for auxiliary connections. Each slot's power bus is individually fused so a single shorted module won't bring down the whole rack.
<http://www.apiaudio.com>

AnaMod XF Tube – AnaMod just seems to come up with the cleverest stuff, and in the cleverest way. Their design approach is to create a digital model of the analog function that they want to perform, then realize that digital model with analog components. One of the things that people discovered early on when putting a piece of vintage gear in an analog signal path is that even with it set to have no effect, just patching it in made things sound better. The reason was that the input and output transformers were still in the signal path, as well as perhaps a tube output stage. AnaMod's latest creation puts an analog modeled input and output transformer and two modeled tube stages in a 500-series module to replicate the components that surround just about every vintage outboard processor that adds that mysterious "analog warmth." Both the input and output transformers can be switched between an iron core or nickel core model transformer, the tubes can be switched between triode or pentode and a single ended or push-pull output, and everything can be individually bypassed. It doesn't glow, and it doesn't have any throughput latency of a digital tube simulator.
<http://www.anamodaudio.com/xftube.html>



Really Nicely Built Stuff

There are hundreds of good sounding and useful studio products that work fine, but there are a few that deserve acknowledgment for very high build quality and attention to special details that just make you feel extra good when you're using them. This of course adds expense and not everyone can justify ownership, but it's nice to know that some builders are putting out the extra effort because they can, and the results show. With all the new really high end signal processors and accessories, it was hard to pick out a few examples of just how well built and thoughtfully designed some of these units can be. There were about 20 ties here. These are a couple that came to the surface as I sorted through the literature pile.

Great River Mix Master 20 – I didn't include this with the mixers and consoles because, while, like a conventional console, it's suitable as the hub of a DAW-based recording system, this the Mix Master 20 is more about analog sound and less about DAW control. It's also about careful and



thoughtful design and top flight construction. There are four mic/line inputs with transformer-input mic preamps, each with phantom power, polarity switching, and balanced inserts. There are also four transformer coupled line inputs with a mic/line switch (not simply an attenuator ahead of the mic input). The outputs of these four channels can be routed to the stereo mix bus, sent directly to a DAW or recorder, and sent to four auxiliary buses.

In addition, there are 16 transformerless line input channels routed to the stereo bus with individual level and pan settings. All signal paths are purely analog, however levels and panning are digitally controlled. The four mic/line channels have their own level and pan knobs, while level and panning for the 16 line level channels are controlled with a single horizontal slide fader, a button to select the active channel, and buttons to select level, pan, and solo to the monitor output. The digital control subsystem allows the mix settings to be saved to a USB flash drive, and taking advantage of the digital control path, a GUI application is in the works to allow mix control, including dynamic automation, from most DAWs.

Mixing is passive, with make-up gain provided by a transformer coupled output amplifier. This goes to a master insert point with transformer coupled return, through the master fader, and finally out through another transformer coupled stage. Control room functions include a control room monitor output to two switch selectable outputs (big and little monitors, for example), channel solos, a stereo input for mix playback, and talkback to the aux (cue) outputs which also dims the control room outputs.

<http://www.greatriverelectronics.com/product.cfm?ID=32&type=0>

Studio Blade Electronics MA-10A Microphone Preamp – When I looked behind the front panel of the MA-10A single channel tube mic preamp, I immediately was carried back to my high school days when I was building my own ham radio gear and I had a part time job in a machine shop. I overbuilt a lot of stuff because I could, and this hand built tube preamp, while pretty simple in design, is built like a tank and I'm sure performs as well. A front panel Input switch selects taps on the mic input transformer to accommodate a wide range of microphone loading preferences, plus additional input transformers for line level and a high impedance instrument direct input. A separate input is provided for 130v powered instrumentation grade mics from DPA and such. Designed for high headroom, the MA-10 can pass +30 dBu end to end without clipping. Gain is supplied by two triode stages, the first one switchable in 3 dB steps, the second continuously variable. VU meter response is selectable between standard VU and peak reading.

<http://www.redwoodcoastmusic.com/studioblade.html>

Got An App For That?

IPads are showing up all over, from set lists to lyric sheets, to chord charts to remote controllers. The fact that all it takes is a WiFi access point and an Ethernet connection to the device to be controlled makes this application ideal. Just to name a couple of apps,

Yamaha StageMix remote controller for the M7CL series consoles allows adjustment of practically anything, but its primary purpose is to be able to set up monitor mixes from the stage. It can display channel names and meters, and the headphone jack on the iPad can carry the mix being adjusted so you can hear what some on in-ear monitors is hearing. I managed to stump the guy demonstrating it when I asked if he could show me a stereo mix with the pots on two auxiliary sends becoming a level and pan, something that can be done on the console. He couldn't make it show that, but it's probably there somewhere, or will be, now.



<http://itunes.apple.com/en/app/m7cl-stagemix/id397288883?mt=8>

Neyrinck makes a lot of high end software applications, mostly for surround mixing and encoding, but he was showing a very nice looking iPad remote for Pro Tools. It comes in two versions, a \$10 version and a Pro version which is capable of displaying more control functions. When I told a friend about this and pointed him to the booth, he got out his iPhone and showed me about a dozen Pro Tools controller apps, but agreed that this one looked nicer than the bunch he already knew about.



<http://www.neyrinck.com/en/products/v-control-pro>

IK Multimedia iKlip - Need a place for your iPad while you're on stage? IK, who has a vested interest in this since they make several iApps including their Amplitube amplifier simulator and iRig guitar input adapter, has just introduced a mic stand adapter that clamps to the back of your pad and attaches it to a mic stand. What'll they think of next?



This wasn't actually at the AES show, but the announcement just popped up in my e-mail as I was writing this section. Think of it as a bonus track, or a preview from the NAMM show report in January.
<http://www.ikmultimedia.com/iklip/features/>

Speakers

For the most part, the big iron was absent from this show but I wanted to give a nod to a couple of speakers.

Focal SM9 – I'm continually impressed by this French loudspeaker company. Every time I see one of their little CMS40 powered monitor I look for the subwoofer behind the curtain. The news this show was the new CS9 with an interesting twist. It's a three-way powered speaker with a top-firing passive radiator. The inside of the box is constructed so that the bass driver is acoustically isolated from the tweeter and midrange drivers. A switch turns off the bass driver and, I believe also changes some crossover settings, allowing it to be used either as a wide range three-way or more "home-typical" two-way monitor. In the "Focus" (2-way) mode, the bottom of the low end response moves from 30 Hz to 90 Hz and the top end (probably because the midrange driver is working a bit harder) is a little less smooth, but this is not an accident. Their goal was to eliminate the need for two sets of monitors for mixing and checking for good mix transportability, and according to some golden ears, they achieved it. About \$7,000 per pair. <http://www.focalprofessional.com/en/sm9-line/sm9.php>

Neumann KH-120 – A year or so ago, Sennheiser (parent company of Neumann) took over distribution of the Klein + Hummel line of loudspeakers, and the KH-120 is their first entry in the studio monitor market under the Neumann brand. This is a small near-field powered monitor that's slightly modified from the original K+H design. Controls are a bit different, and a digital (AES3) input is optional (that's the model KH 120D). I've always liked the sound of K+H speakers, and it's nice to know that they now have solid distribution to the studio marketplace. <http://www.kh120.neumann.com/index.htm>

Useful Stuff

As one who isn't a big user of specialized software, I've lumped software here, as well as other handy little things that can make your work easier or are just cool to have around.

IZotope Nectar – This is more of a packaging job than a new software development. Oriented toward vocal processing, there some new tricks and new user interfaces on top of a collection of IZotope processing modules - EQ, dynamics, de-esser/breath control, pitch correction, double-tracking, distortion, delay, and reverb. The user can select a genre such as blues, country, jazz, rock, voiceover, and the package comes up with a suggested set up presets and the appropriate processing chain. Everything can be tweaked, though an Advanced mode allows you to get deeper into the adjustments.

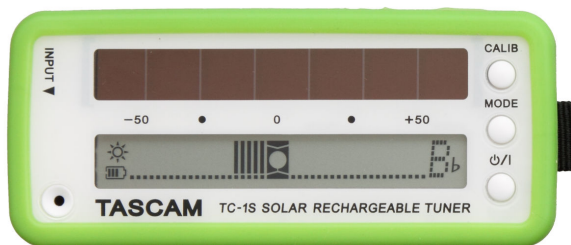
<http://www.izotope.com/products/audio/nectar/>



E.A.R. SoundChecker – This is a keychain sized sound level meter from EAR, Inc makers of custom molded ear plugs, so they care. Simple to operate, you just press the button and the red, yellow, and green LEDs tell you if the sound level is safe, if it's getting to the harmful level, or if you need hearing protection. I made a quick comparison with my trusty first generation Radio Shack sound level meter and the SoundChecker tracked it pretty closely. If you own a music venue, operate live sound, or work in a noisy environment, you might want one to keep an eye on how much damage your ears are being subjected to. SoundChecker - \$25. Hearing – Pricele\$\$.

<http://www.earinc.com/earsoundchecker.php>

TASCAM TC-1S – Everybody wants to be green nowadays, so here's a solar powered instrument tuner. Dark stage? No problem. There's an internal rechargeable battery which will charge from the solar cells or through a USB port. It comes with a silicon protective cover in your choice of colors, has a built-in microphone as well as a 1/4" phone jack to plug in an instrument. It's chromatic so it can be used with any instrument, there's a fine pitch resolution mode that reads offset down to 1 cent for those non-tempered instruments, and you can calibrate it to a hard-to-tune instrument so the whole band can tune up to the Autoharp. There are a couple of display modes. The default display



is a horizontal bar with “on pitch” in the center. There’s also an animated mode that emulates a strobe disk which I sort of like. I find that type of display easiest to get me close. Under \$40.

<http://tascam.com/product/tc-1s/overview/>

Soundproof Windows, Inc. - They do Windows – Studio Window Max and Studio Door Max are build from heavy glass in a pre-fab sealed mounting frame. Both are custom-sized. The window assembly fits over an existing window, and the sliding door is built to replace an existing one. Typical STC with 5/8” glass is of 60 to 63, which is a good upgrade for a bedroom studio.

http://www.soundproofwindows.com/recording_studio.html

Pro Tools 9 – What’s a show without Pro Tools? The big news, in case you’ve been living under a rock, is that, unlike previous versions, with Pro Tools 9, you can use “any” (quotes are mine, because “any” covers a lot of untested waters) audio I/O hardware that has a standard ASIO driver or is supported by Apple Core Audio. Of course Avid has their own hardware that they’d like you to purchase, and Pro Tools HD (with the hardware DSP card) still requires their hardware, but this opens the door to a lot of potential customers who either had good hardware that they wanted to continue using, or those who simply rejected the idea of buying into a proprietary system. For existing users, there are upgrade deals and crossgrade deal for LE and M-Powered users.

The way thing seem to be lining up, Pro Tools LE and M-Powered versions will go away. PT9 is the upgrade path for both, and it’s promised that your present LE or M-Audio hardware will work fine with it. Although I didn’t get a definite “yes,” I got a definite “I assume so” when I inquired about the Mackie i series mixers with Firewire I/O, since they have a driver add-on that makes them look like hardware that’s supported by the M-Powered version. Can you use ASIO4ALL with your vintage 1997 Soundblaster? Try it and see.

I had a chat with one of the tech guys at the booth because I was curious as to how they planned to handle that “any.” He said they didn’t yet have a plan, but they would probably start building a list of devices as they get around to testing them, either of hardware that worked or didn’t work. At the moment, they’re just deferring that issue to the Pro Tools Users forum, so most of the compatibility info will be coming from users’ experience.

While the hardware is probably the biggest new thing with Pro Tools 9, there are plenty of other new features such as upping the track count to 96, automatic delay compensation, and compatibility with the Avid (formerly Euphonix) Artist series controllers.

There was also some new Pro Tools hardware on display. The Omni, from a distance, looks like an updated version of the rack mount version of the Digidesign 003, but it's a whole lot more (and a whole lot more money, too). It's compatible with Pro Tools HD, but also with PT9. It features two mic preamps, and a plethora of both analog and digital inputs and outputs, though you can only send 8 simultaneous streams to the computer. Unlike the older hardware which depended on the software for mixing, the Omni includes an on-board DSP mixer which provides near zero latency input monitoring, performance pretty much on par with a full out HD system. When you set up monitoring in Pro Tools with an Omni attached, you're actually talking to the Omni mixer rather than the Pro Tools mixer.

There's also a new HD I/O box to replace the 96 and 192 HD interfaces. The basic unit has stereo AES3, S/PDIF, and ADAT optical I/O. Additional I/O is handled through up to four optional I/O cards. There's one card with 8 channels of analog input on a DB-25 connector, a companion card with 8 channels of analog output, and the third card that provides 8 channels of digital I/O in ADAT optical, TDIF, and AES3 formats. Maximum I/O fully loads is 16 channels in and out.

The best ad copy relative to Pro Tools 9 that I've seen so far is from Focusrite: "Pro Tools 9 – Now Focusrite Ready!" touting their Saffire range of I/O boxes. Newsworthy that Focusrite has an RTAS plug-in in the works to bring up the Saffire mixer/setup control panel from within Pro Tools rather than have it as a separately running program.

<http://www.avid.com/US/products/Pro-Tools-Software> (the software)

<http://duc.avid.com/forumdisplay.php?f=50&cmpid=AV-V-PT9-2> (users forum)

Technical Program

There are far too many technical sessions to take them all in, so one must choose prudently. Here's where I spent my off-the-show-floor time.

In addition to the myriad of highly theoretical papers, this year's technical program included a new track devoted to product design. Although I don't consider myself a product designer, I particularly enjoyed a session of war stories and lessons learned working with offshore manufacturers.

Since moving to digital TV, there are more places where audio and video can get out of sync, and a round-table of video designers presented the whys, wherefores, and possible implementations of embedded time relational data in the MPEG stream to allow a sufficiently smart TV set to put sound and picture together accurately regardless of how they went astray in the processing and transmission chain.

A session on orchestral recording in 5.1 surround turned out not to be what I expected, but rather how sound designers for games record orchestras so that parts can be placed dramatically to go along with the action. It's no secret that these days much orchestral recording is done with dozens of spot microphones in addition to a general stereo or surround pickup. These game folks are taking a big step further because they need to be able to isolate sections and solos. Part of the production involves having sections and soloists play by themselves, yielding tracks essentially identical to a modern pop multitrack session. I was sharing this with an orchestra conductor friend over dinner, and he was suitably both impressed and appalled that a producer might want to have, say, the string section coming from a moving car outside the concert hall.

Two top experts in the field of grounding and shielding, Bill Whitlock and Ralph Morrison presented a series of sessions on equipment design flaws which almost guarantee hums and buzzes, and how to avoid them. In addition to his regular talks on shielding and balanced interconnections, Bill presented a poster session about hum problems which result from building wiring. His postulate is that the small but measurable difference in voltage between grounds is a result of magnetic induction between the line and ground wires acting as a one-turn transformer. Case history demonstrates that simply twisting the line and neutral wires together (this of course calls for conduit, not Romex) can profoundly reduce system noise problems.

Of course with the upcoming regulations about loudness differences in broadcast program material, loudness measurement and management both for US and European standards was a hot topic. The Loudness and Dynamics session began with a good overview of what's commonly known as "the loudness wars," exploring the history of "louder is better" and the consequences such as aesthetically unpleasant sound and rapid listener fatigue. The speaker presented evidence contrary to the common assumption that perceived loudness is correlated to listener preference and sales. Follow-on presenters offered fairly technical discourses on loudness measurement algorithms and methods. The bottom line is that broadcasters will still strive to grab our attention to commercials, but hopefully in a more polite manner than pushing the audio level to the max.

Trends

Everyone was predicting that this would be a small show, and, at least physically, they were right. About a quarter of the exhibit floor in the Moscone Center North Hall was curtained off so it wouldn't look empty, but there were still plenty of exhibitors, many of whom had smaller booths than in previous years. Live sound products (speaker systems, large consoles, power amps) were conspicuously absent from this show, likely because InfoComm and NAMM have become their

major venues. The trend at this show was toward higher end products. There were a few hundred dollar mics on display, but these were the low end of a full product line, not the company's primary focus. Ribbon mics and 500-series modules are still strong, and there were a number of high end (and high priced) mic preamps. I was particularly happy to see consoles making their way back, too.

Yup, it felt like an AES show.