SHOW REPORT

AES Nuggets from Amsterdam

Audio Engineering 124th Convention, May 17-20, 2008.

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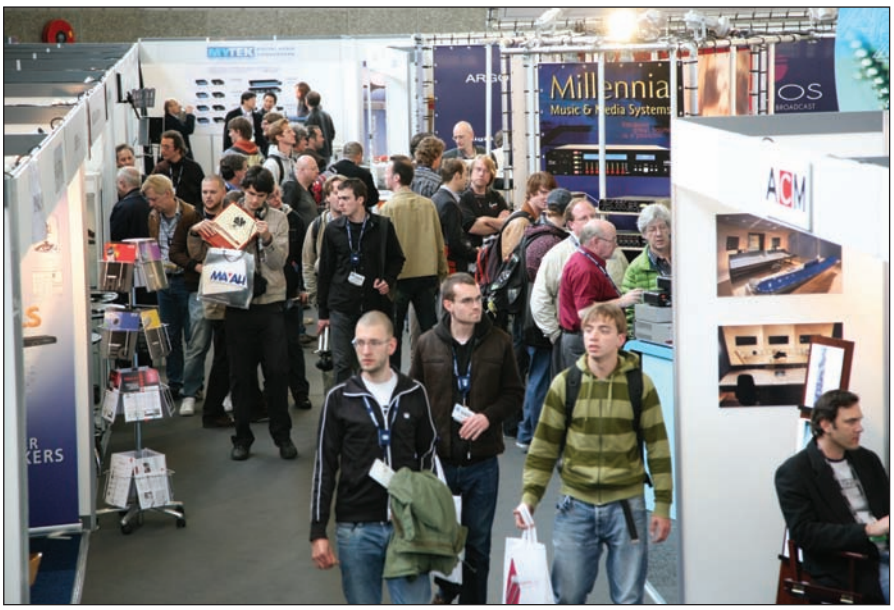

It's been a while since I attended an AES convention. Many years ago, I noticed a gradual shift away from practical audio presentations to more fundamental research, heavy on high-level mathematics and material physics. But the pendulum seems to have reversed direction. Lately, I see an increase in practical audio again—articles, presentations on the application of amplifiers and loudspeakers, and workshops on audio recording, multimedia production, and the like.

Because this year's spring convention was within easy driving distance for me, I thought I'd go again. And, I was not disappointed; quite the contrary!

A typical AES convention has many parallel threads with presentations, workshops, poster sessions, and tours, so the main problem you face is making choices. I'll report on the events I attended, but I missed a few I would have liked to attend as well, such as the presentation by Menno van der Veen, of audio transformer fame, about mechanical resonances in capacitors, and the one by Guido Tent of Tentlabs on the 100th birthday of the triode.

Speakers have no parameters! Believe it or not, but none other than Neville Thiele said that. But, he added quickly, models have parameters! Thiele and Richard Small combined to present the history and development of the Thiele/Small parameters. Thiele related that he had developed his model and the famous T/S parameters already in the late 50s, but that somehow it didn't catch on. Then, ten years later, Dick Small arrived in Australia to do doctoral work on speakers. He complained to a colleague that he couldn't find any info on speaker modeling in any US university library. The colleague advised him to look in the library of Melbourne University, where he discovered Thiele's work. The rest, as they say, is history.

They mentioned two recent areas of study they undertook. One was the change of alignment of speakers with aging. Although driver compliance does change over time, it appeared to have negligible impact on the speaker alignment.

The other area of research was the development by Richard Small of a software application able to "tweak" model parameters in such a way that the resulting response matched a given set of measurements. That seemed like a step forward in efficient design. You measure the response of a driver, and let the software figure out the T/S parameters that go with it.

Another application of that software is to compare measured parameters with those from the matching process I mentioned. The delta would be the influence of the measuring room. This influence can then be compensated for, so that in theory you would be able to make anechoic measurements in an ordinary room.

There were many presentations that focused on sound localization and understanding and interpreting messages and spoken text. One research report was on localization of obstacles by visually impaired persons using sound cues. It highlighted the uncanny performance of these people to neatly circumvent obstacles by only listening to changes in a sound field caused by the obstacles, changes that a seeing person would probably not even be aware of. The results were much better with the person moving rather than static. Changes in

reflection, inter-aural time differences, and level changes all helped the person make a mental integrated picture of the physical environment. It would be possible, in principle, to guide blind people with a "sound projector," much as seeing people would use a flashlight.

DISTRIBUTE THE WOOFER!

Ben Kok, who made a name designing listening venues and studios, gave an interesting two-hour presentation to a packed audience about the acoustics of small rooms—where you and I listen to our audio systems. Room acoustics is quite important, and can make a huge difference in sound perception, but seems difficult to grasp in a couple of simple equations.

For instance, if you have widely different length, width, and height dimensions of your room, the standing waves are distributed, which helps to avoid pronounced resonances at certain frequencies. So, you would think that someone would have figured out the optimum ratio of those dimensions. Well, they have. It is called the "golden ratio" of length, width, and height, which is 1:1.77:1.47. Or, you might find it as 1:1.45:2.1, which is incorrect, according to Ben Kok. He uses 1:1.12:1.41.

And oh, by the way, if you think that subwoofer in your 5.1 surround system improves the low-frequency response, think again. It is very difficult to control LF resonances in a room. One good way is to use "distributed subwoofers" at different positions. That way, the level at each position is lower, and the resonance pattern becomes more diffuse, without strong peaks. So, if you use speakers with good LF response in a 5.0 system *sans* subwoofer, you have, in effect, five distributed subs and potentially much better LF reproduction.

BATTLING THE LOUDNESS WARS

Quite a few workshops related to what I would loosely call Battling the Loudness Wars[1]. As broadcasters and CD producers alike vie for the attention of listeners, they try to make the program sound as loud as possible. So, you record the sound with the peaks up to 0dB of

your recording medium, and then you pull up the soft passages to almost as high. The music will sound very loud, but will have very little dynamic range, often just 10dB or so, but the lower the dynamic range, the louder it sounds, and the more attention it gets. You may have noticed the same effect with TV commercials: They sound much louder (and much more annoying!) than the program material. Music compressed in this way sounds awful, and you turn it off in disgust after more than a few minutes.

It's not the recording engineer; it's the stations and the marketing people that drive it, and sometimes the artists. One panelist related that in a particular case the band he was producing wanted "clean, dynamic sound," only to come back after a few days asking "why is my CD not as loud as xyz's?" Especially young people growing up with the current loudness wars may not even have any notion of dynamic range and how that may play a role in appreciating and even understanding music. One example mentioned a classical concert given for school classes, in which many students seemed quite puzzled and asked "why are some of the passages so quiet?"

There are other subtle effects. If you record up to 0dB digitally, that means the digital samples remain at or below 0dB. But when the signal becomes reconstituted in the DAC, it may well be that the analog output signal is higher in level. An unclipped digital signal may generate analog clipping. If you record a piece of music once at 0dB and once at -6dB, there is a big difference in distortion products at the analog side.

It was also found that data reduction schemes such as MP3 can lead to an increase in program peak levels and thus to added distortion. One demo played the S signal from an M-S AAC-128kbps compressed signal. In theory, all things being ideal, this would be total silence, but in reality distortion, artifacts, and noise were audible to the point where you could actually recognize the music.

Generally, the opinion was that music recorded and mastered in the early 90s was probably the best in terms of dynamic range, and that it has been going downhill ever since. For example, a 1992 album from Bonnie Raitt with great dynamics contrasted with a recent album that was more loud than anything else. But the tide seems to be turning.

Several initiatives are underway, notably by Italian and Canadian broadcasters, to measure and control program loudness rather than peak levels. It involves new ways of assessing program "loudness," and several companies were showing special-purpose metering equipment for this. One interesting metering system from TC Electronics gave two numbers related to the dynamic range of the signal. "Center of Gravity" (CoG) showed the level of the signal most of the time, while "Consistency" (Cs) indicated the variations of the signal from the Center of Gravity.

Some very loud and compressed tracks were played that gave CoG and Cs numbers indistinguishable from those of white noise. And, to be honest, they also sounded a lot like white noise! At any rate, the attention on loudness issues can only lead to better sound for all of us.

BACK TO AUDIO DESIGN

Douglas Self, who shouldn't need any introduction, gave a master class on analog amplifier design. Basically, he covered the wealth of material from his two books (*Self on Audio*; *Audio Power Amplifier Design Handbook*). There were a few insightful nuggets for me. For instance, that the dominant pole compensation capacitor Cdom transfers the global feedback at low frequencies to local Vas feedback at high frequencies. As a result, Vas gain changes from Gm * ß * Rc (where Gm is input stage transconductance, Rc is Vas collector load) at low frequencies to Gm/ω*Cdom at high frequencies.

Self investigates these effects with a test amp consisting only of the input- and Vas stages followed by a small class-A output stage to drive the test equipment. In this way he can isolate Vas nonlinearities from output stage ones. He also found that common-mode distortion of the input differential pair at a closed loop gain of 23 is too low to matter, so from this perspective he didn't see any reason to use cascodes in the input stage.

Still on the input "long-tailed pair": it is well known that unbalancing the currents in a LTP increases distortion because the operating point shifts away to a less linear part of the transfer curve. What I hadn't realized, but Self pointed out, is that such an unbalance also results in a lower Gm, in turn leading to less available loop gain for the negative feedback to work its magic. So, there are actually two effects that both lead to increased distortion.

Then, finally, you can minimize the output stage crossover distortion with measures such as the smallest Re value that still gives sufficient thermal stability, multiple parallel output devices, and accurate setting of the bias current. But in the end, Self conceded that it is something you just have to live with, whether in class-A, AB, or B.

My current interest is in Darlingtons with integrated on-die thermal compensation diodes such as the Sanken STD03N and P devices. They have very high gain and very high Ft, and the on-die diodes give almost perfect thermal stability. One disadvantage is that the bases of the Darlington drivers

are not accessible, so the usual resistor to aid in removing base charge to speed up switch-off cannot be fitted. I specifically asked Douglas about this, but in his opinion, with modern devices with Ft of 30MHz or more, he didn't think it would be an issue at all. I was also pleasantly surprised when he mentioned my paX error correction amplifier article in *Elektor Electronics* as an interesting new way to design power amplifiers, although he thought it would be very difficult to match his "blameless amplifier" design in performance, for the same parts outlay.

In contrast, the next workshop was by Bruno Putzeys from, among others, Hypex class D fame. Putzeys' talk was aptly called "Life on the Edge." He started by impressing on us that "digital amplifier" is an oxymoron. Class D amplifiers deal with current, voltage, time; all purely analog quantities. So, maybe the contrast with Doug Self's analog design wasn't that great.

Interestingly, Putzeys echoed many concerns voiced just a few weeks earlier by Martin Colloms in his talk at the Munich Hi-End show. Then, Colloms was quite critical about the (lack of) sound quality in many class-D amplifiers. He blamed the fact that many class-D amps were not well designed with regard to the high-frequency radiation and conduction requirements necessary to keep the output signal as well as the immediate environment of the amp clean. Putzeys concentrated more on the technical issues, which were not as focused on getting the amp linear (which is relatively straightforward) as on exactly these EMI and filtering issues.

There are many high-level signals within a class-D design with spectral components into the hundreds of MHz. The picture that emerged from both Colloms' and Putzeys' talks was that the best class-D designs are those that control the EMI issues. Physical component location with respect to each other, physical properties of filtering components, PCB layout and orientation, and power supply decoupling are where the money is.

You must realize that a filter inductor at one frequency can be a series capacitor at another and vice versa. Supply loops that carry currents with very high frequency components need to be absolutely minimized or they radiate all over the place, even into other parts of the sound system to which they aren't even connected. Bottom line: if you are in the market for class-D, you must look at different specs than what you would normally look at for a non-switching amp. And you should listen, of course.

AES conventions are always a smorgasbord of subjects and events. There is truly something for everyone; if you have the opportunity, do attend. Joining the AES is now easier than ever; even serious audio enthusiasts who don't make their money in audio can join online at www.aes.org. Recommended. *aX*

REFERENCE

1. "Bringing back dynamics into the mix," panel chair Ronald Prent, Galaxy Studios, and "Wow, this sounds great," panel chair Christian Hugonnet, Consultant.