Active Loudspeakers: Design and Optimization Masterclass

Cambridge-based seminars & live global webinars

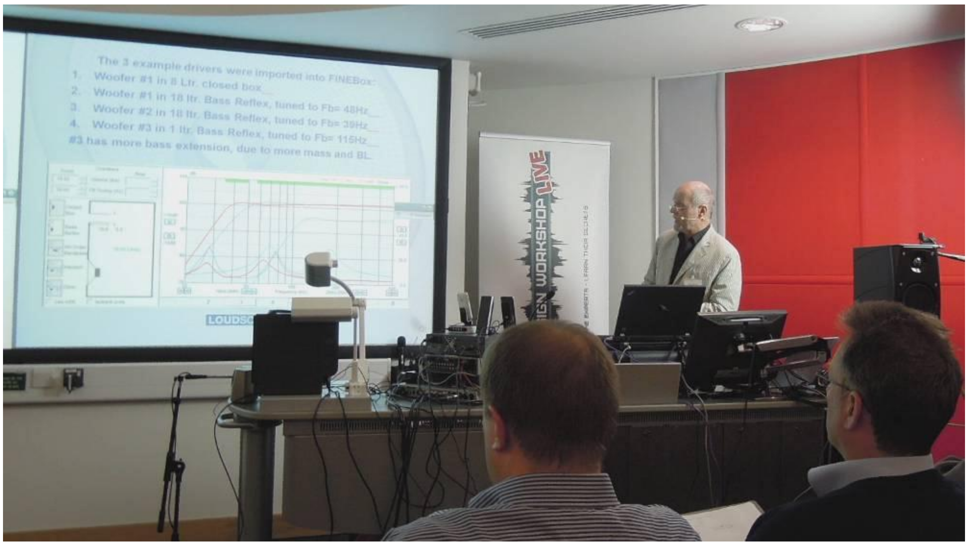
Monday 25th March 2013, 13:30 - 19:00 GMT
Prism Sound and Oxford Digital, in conjunction with the AES, LOUDSOFT and TTid, organised an audio engineering masterclass at the Anglia Ruskin University, Cambridge. This was an opportunity to meet with some of the most experienced engineers in the industry, and to hear about their practical experiences in analogue and digital audio design in particular relating to active loudspeakers especially compacts. In addition to seeing some of the latest developments in design, evaluation and voicing, we learned from the many decades of collective audio experience of our panel of presenters.

The sessions were also webinar broadcast, the whole ably controlled and directed by Simon Woollard of Prism, and ended with a live Q/A ,also taking internet questions for the expert panel and the attendees.

1. LOUDSOFT's Peter Larsen kicked off the sessions, showing in real time how the loudspeaker development process can be greatly streamlined with this impressive suite of modern CAD tools. Titled

Compact Loudspeakers - a Practical Transducer Design Workshop

Peter seamlessly took us through two worked examples for 6.5inch and 3 inch drivers with simulation and optimisation routines for the detailed design of the drivers and for matching enclosures, including low frequency alignments. The pros and cons of ferrite v neodymium magnets were covered, also short and long voice coils, even modelling of the excursion linearity of the magnet design and how to optimise Blx for lowest distortion and maximum sensitivity. He showed how transducer diaphragm behaviour could be modelled to significant accuracy with FEA style animations, clearly revealing the complex coupled behaviour of cone, voice coil, dust cap and surround and suspension.



Peter Larsen demonstrating interactive low frequency speaker design

His software suite is organised as a loop or circle where each software component supports the other and includes an R&D section for measurement and analysis, and also a quality control suite including error bands and Pass Fail criteria. With reasonable acquaintance it was clear that these relatively straightforward tools were rapid and effective in modern transducer and speaker system development. He showed how much of the simulation aspects closely matched sophisticated Klippel measurements of these devices when built .With years of transducer design behind him, he remarked that a key aspect of driver performance, dependant on design and production accuracy was minimising the voce coil offset in the flux field.

2. The second session began with John Richards of Oxford Digital, and supported later by co founder Peter Eastty, in a session entitled:

Optimizing Compact Loudspeaker Performance - the role of DSP

Here the application of DSP techniques is used to maximise the subjective performance of compact loudspeakers, for a given budget and footprint. However there was much more to this seemingly innocent subject. The Oxford team have been plugging away at this topic for a few years now and showed us a sophisticated suite of filters and equalisers, threshold limiters, elegant peak compressors and others, where the narrow range, highly peaky frequency response, and limited dynamic headroom aspects of a miniature loudspeakers could be made to sit up and beg. Here we are talking of perhaps of 18mm diaphragms in 2cc of enclosure driven to within an inch of their lives, where in context, though a process of measurement and subjective analysis, and with interactive tuning, a truly appalling noise is turned into acceptable audio reproducer of greater subjective bandwidth and loudness.
A number of internationally famous names, in mobile phones, personal audio/video and hand-held gaming have benefitted from the evidently powerful and versatile approach offered with this hands-on voicing tool. Tips were also given on session duration, the need for rests, use of acoustic

references, and a sensible approach to maintaining similar loudness for comparisons, in particular when equalisation is being applied on an iterative basis: here a key question is
‘Are you levelling up, or are you levelling down?’

Ian Dennis introducing speaker John Richards (right)

The control interface comprises a number of well arranged screens whereby the corrections may be viewed and auditioned, the facilities, including automatic curve fitting with selectable limits for degree and overall frequency range, to avoid unstable optimisation and excessive corrections.
The eq can be programmed to track the dynamic envelope of the test object in time and frequency and thus the eq. dynamically fitted to it.

A small computer speaker of unpleasantly coloured and thin sound was optimised in real time for numerous parameters including active bass control and while not Hi Fi as we know it, was rendered surprisingly plausible, a change which would have gained the product many more marks in comparative review.

The bottom line is that nearly all of the smaller modern electronic speaker equipped devices have some silicon, enough to take an upload of the DSP parameters generated using the Oxford Digital suite, and thus enjoy a near instant sound quality improvement, with no other changes to the build.

I could also imagine an early prototype of even a Hi Fi speaker, one where the starting point was a good enclosure and drivers and a crossover which integrated the drivers well in phase and amplitude, but where the overall sound was far from properly voiced. Then pop the Oxford suite into the digital replay chain and tweak away until the sound is ‘right’, followed by a reworking of the speakers, now with a micro tuned amplitude response executed in crossover hardware to match the results of the now prototyped DSP solution.
If the speaker is a digital active the route is still more obvious.
To a large degree it could certainly help a designer to predict how the finished loudspeaker could sound. Unpromising driver combinations could be weeded out early in development.

While the equalisation offered by the suite was clearly sophisticated, in my view the particular value lay in the quality of the user interface which was evidently well sorted.

3. *Audio Power Amplifiers; EMC Best Practice Revealed*

Tony Waldron of EMC control fame took us through a commonsense reading of the EMC situation facing the industry, the degree of radio frequency pollution in which we are all immersed, and the consequences for audio equipment, both emissive, radiating problems, and susceptibility, ie suffering problems. He decried the ubiquitous pro connector, the XLR as poorly behaving in respect of EMC and explained that balanced working and the associated benefit of common mode rejection, and not least screening for RFI, only worked up to a few kHz and was primarily developed to control hum induction in long mic cables, these including microvolt output ribbon types. At higher frequencies you are on your own since the cm or so of wire from pin1 to ground is already self resonant in the UHF range, and all bets are off for the higher microwave frequencies from mobiles and WiFi.
Tony noted that the regulations for field strength included what seems a massive 3V/m specification and yet that power is what numerous ubiquitous pocket transmitters, namely mobile phones can produce. He explained that there was no substitute for lots of high conductivity metal screening and trunking and very short low impedance ground connections, plus enclosures made without sets of slots acting as aerials or aerial arrays. Cable screens should be strongly bonded to case metalwork at the exit and entry to the equipment. For the insides he reiterated the importance of large area ground planes for analogue PCB design and for digital, that multilayer and ground plane practice is essential if the devices are to meet the onerous RFI emission regulations.

Tony Waldron with Simon Woollard

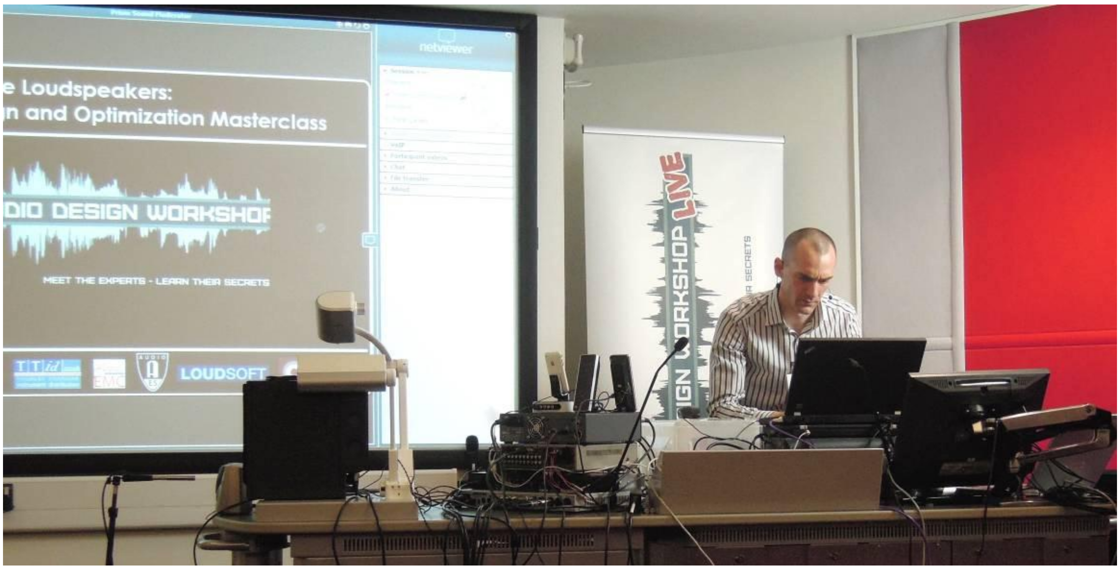
He explained that if, for example, a power amp design passed the radiation requirements even with a switch mode power supply and with Class D

amplification, it was very likely then to pass the susceptibility tests, thanks to a measure of reciprocity. He also warned of the poor output filter design often found with Class D amplifiers the latter capable of considerable RFI.
He advised the use of pre-tested OEM power supplies where possible noting that these will have survived very costly EMC testing and also pointed out that computers remained a significant noise source where 4GHZ processor clocks generate spuriae right down to 10kHz, i.e. into the audio band.
A final caution related to the ubiquitous in-line chassis filters which should be inspected for topology, i.e. there should be a double pole filter, one for the input side and one for the output side, often visible in a small diagram printed on the filter casing.
The discussion also included the noted sensitivity of audio ‘men’ and the like working in studios, to really small differences, such as 0.1dB and less of volume, and it was accepted that while such people couldn’t always explain what was causing the difference, they could indeed hear the difference.

This matter was raised in the later Q/A sessions where a web listener reiterated the book value of 1dB being the usual observable level difference. At this point virtually the entire panel agreed that for critical comparisons 0.1dB was indeed detectable and attendee Jon Honeyball also confirmed that in critical, blind, multiple tests he had organised, for example on CD players, 0.1dB level matching was required for reliable results from a practiced listening panel.

4. *Audio System Analysis - Tips and Tricks to Verify Your Designs*

Simon Woollard of Prism Sound, supported by Ian Dennis, spun the audience around with rapid fire set ups of the Prism audio test set, a highly versatile analogue and digital audio test suite where both the practicality and the deep analytical aspects of the supporting software were showcased. All the classic audio measurements were handled with ease, not forgetting the loudspeaker test facility including an efficient logchirp, gated functionality. The discussion then turned to a philosophical debate on classic measurements such as THD+N routinely used by the industry but which all agreed were lacking meaning.



Simon showed that separating harmonics from noise, and analysing and weighting these components separately using selective FFT methods could be very useful in showing the true performance, and thus readily guide the designer to product improvement. Of particular interest to me was the burst distortion capability where the DUT in particular a loudspeaker system could be exercised non-destructively at higher powers. The ability to gate the received data, window it properly, and then use FFT to assess the distortion, this including harmonic analysis, is a marvellous facility and could prove very useful in exploring distortion at peak programme levels in loudspeakers, an important but at present poorly investigated area. The versatility of such instruments is dependant on the control software and here the art has advanced by leaps and bounds.
The point made was that such versatility and insight helps guide the improvement of all aspects of active loudspeakers, both the electrical and acoustic domains. Simon crammed in many demonstrations, retaining our attention to the close of the presentations.

5. Q and A: After a short break the formal sessions ended with a Q and A where the whole team contributed.

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The event was packed out, also with up to a 100 also participating via the webinar link, and was rated a great success by all involved.

Many thanks to the entire team, including AES Cambridge branch, for their excellent preparation and commitment, sharing their technical resources, their learning and experience with the many participants.

Martin Colloms   HIFICRITIC AUDIO MAGAZINE   29 03 2013