AES 51st International Conference

Loudspeakers and Headphones

21–24 August 2013

Helsinki, Finland

CONFERENCE REPORT

Helsinki, Finland is known for having two seasons: August and winter (adapted from Connolly). However, despite some torrential rain in the previous week, the weather during the conference was excellent. The conference was held at the Helsinki Congress Paasitorni, which was built in the first decades of the twentieth century. The recently restored building is made of granite that was dug from the ground where the building now stands. The location near the city center and right by the harbor proved to be an excellent location both for transportation and the social program.

An unexpectedly large turnout of 130 people almost overwhelmed the organizers as over 75% of them registered around the time of the “early bird” cut-off date. Twenty countries were represented with most of the participants coming from Europe, but some came from as far away as Los Angeles, San Francisco, Lima, Rio de Janeiro, Tokyo, and Guangzhou. Companies such Apple, Beats, Comsol, Bose, Genelec, Harman, KEF, Neumann, Nokia, Samsung, Sennheiser, Skype, and Sony were represented by their employees. Universities represented included Aalto (in Helsinki), Aalborg, Budapest, and Kyushu.

A packed House of Science and Letters for the Tutorial Day

Juha Backmann insists that "Reproduced audio WILL be better in the future."

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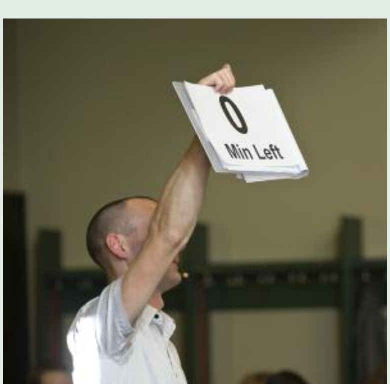
Wolfgang Klippel tells everyone to "Settle down now."

The conference was chaired by the ever-exuberant Juha Backman, and the excellent papers were collated and reviewed by Aki Mäkivirta. They were assisted by an able committee consisting of Andrew Goldberg (sponsors), Ilkka Huhtakallio (publicity and everything else), Asta Kärkkäinen (demos), Julia Turku (treasurer), Jussi Rämö (website), Miikka Tikander (facilities), and Markus Vaalgamaa (social program). From Aalto University, Lauri Mela, Pekka Rönkkö, and Jukka Saarelma stepped in at the last minute to give additional support to the committee.

TUTORIAL DAY

Juha Backman welcomed everyone to the tutorial day. This was a special event held at the House of Science and Letters and organized in conjunction with the Finnish Acoustical Society and the Audio Engineering Society. Graham Boswell then introduced the Audio Design Workshop for Loudspeakers. This was the first time the AES had done a webcast from a conference and over 160 people tuned in, which more than doubled the total audience to almost 300 people. The workshop was jointly run by Prism Sound, Oxford Digital, Texas Instruments, Klippel, Loudsoft, and Audio EMC, and was a repeat of a similar event held a few months ago in Cambridge, UK. This was an opportunity to hear about practical experiences in analog and digital audio design, and to see the latest developments in technology.

Wolfgang Klippel in his typically infectious, enthusiastic, and confident style discussed his ongoing efforts to define and develop meaningful audio performance metrics for today's smart devices. So many areas were covered in such a short space of time that it's not possible to even attempt to summarize the content here in this small space. One simply has to see Klippel in action to understand this. Peter Larson of Loudsoft then explained how the loudspeaker development process can be greatly streamlined with modern CAD tools. Motors for small drivers, bass alignments, and low-frequency performance can still be designed using Thiele-Small parameters in a simulation, and the effect of individual parameters (such as voice coil length and pole piece size) on the system performance can be seen directly. John Richards of Oxford Digital showed us how the application of DSP techniques can maximize compact loudspeaker performance within a given budget and footprint. Even as products shrink in size, customer expectations are rising, but DSP can be used to counteract these contradictory demands. Lars Risbo of Texas Instruments addressed the performance benefits afforded by a system-level approach to active loudspeaker design, as opposed to optimizing individual components in isolation. Looking at it this way means that overall system limitations come from peak excursion or the thermal average. Simon Woollard of Prism Sound demonstrated how modern digital audio analysis techniques can provide far greater and more rapid insight into system performance. A practical demonstration showed measurements of digital interfacing bit transparency, drop outs, and glitches. Bin-based multitone FFT rapidly acquires many curves in a short time (approximately 30 per minute) and measures THD+N, adding filtering can give THD or N only. For acoustical measurements, swept sine waves are often used to calculate an impulse response, whereupon windowing extracts the distortion harmonics. Anthony Waldron of Audio EMC provided practical solutions to the often-overlooked problems of electromagnetic interference and its impact on audio system performance. Old amplifiers can still pass modern EMC tests due to the shielding provided by metal cases. Now radio interference comes from mobile phones, computers, SMPS, lighting, and motors. Interconnecting cables act as antenna and audio frequency pickup occurs because audio circuits can demodulate RF. Ground loops can be avoided by connecting the metal chassis of the equipment to metal trucking of the installation and clamping the cable shielding to the metal case. Common-mode filters and power supply decoupling also help. A good ground plane is essential and can improve EMC performance by tens of dB.



Simon Woollard tries subterfuge to get his turn on stage a bit earlier.

After the workshop we visited the National Library of Finland. In the audio department they are about halfway through a digitizing project. Currently all the Finnish 78 rpm records have been successfully digitized and now they have approximately 150,000 recordings from throughout the Finnish history. The earliest one is from 1901, and about 80% of all Finnish recordings can be heard by appointment.

Following the library visit we were bussed over to the Aalto University's Espoo campus for a barbecue and sauna evening. Traditional Finnish makkara (a sausage primarily consisting of fat and water that needs some serious burning before it has any taste at all) was served together with slightly more healthy salad and slightly less healthy Finnish beer. Judging by the noise and smiles,

Sauna, beer, and makkara on the beach.

this proved to be a very popular ice breaker. The stillness of the trees and bright shining moon in the clear sky added to the atmosphere which set the scene nicely for the official start of the conference.

DAY ONE

Juha Backman officially welcomed everyone to the conference and introduced the committee. Then Aki Mäkivirta introduced the keynote speech by Ilpo Martikainen of Genelec. He explored the evolution and history of studio monitoring loudspeakers and explained that we know a lot about loudspeakers, however there is still a long way to go. Sound sources have many different directivities but the loudspeaker has only one. He asked what the reference is: a kitchen radio or the best possible technology? Humans and rooms are also part of the listening experience and they vary greatly, also loudspeakers have a frequency response, distortions, and directivity. While the goal has changed from the AM radio era to higher quality today, the basic principles of loudspeakers are unchanged since the early 1900s. Basic human perception has also been well understood for many decades, however it takes a long time for research to become accepted by the mainstream. Martikainen then toured through various design aspects seen in studio monitoring such as coaxial drivers, beryllium and soft cones, directivity (mid-high frequencies and also bass), room response controls, edge diffraction and cabinet shapes, motional feedback, amplifier output impedance, compression horns, the room interface and perceived audio quality, and most recently DSP processing. He concluded by noting how many of the features seen in today's designs originally came from demands made by broadcasters, and that loudspeakers for professional applications have a much longer lifetime (typically 10–20 years) than consumer products. Over the years, size has been reduced, materials have improved, frequency responses are flatter, directivity control is smoother, knowledge of perception is better understood, electronic equalization is more extensive, and prices have dropped.

Top: Ilpo Martikainen (keynote speaker) making history flow.
Bottom: A packed lecture room for the Testing and Measurement session chaired by Jorma Salmi (standing)

Aki Mäkivirta then chaired the loudspeakers and applications session, which started with a description given by Dodd of the development of the KEF LS50 that has a coaxial driver arrangement. In a model, a radial channel waveguide is used to optimize the controlled directivity producing a spherical wave front to give a wide and even dispersion, with the driver surround having a minimized diffraction. The cabinet, driver, and port were analyzed in another model to show how they perform as a complete system, and this prompted some design changes. Daniel Beer then followed with an exploration of the influence of flat loudspeaker enclosures on the electromechanical properties of light and small electrodynamic transducers. These drivers are often seen in wave field synthesis, automotive, mobile devices, and the home. Flat enclosures have a lower resonance frequency than conventionally shaped designs and THD is improved. Also, below the resonance frequency the output is higher but the opposite is seen above. All this is due to boundary layer effects increasing friction in the direction of the short dimensions. Next van Maanen outlined the requirements for loudspeakers and headphones in the "High Resolution Audio" era. He stated that the response above 20 kHz is important for audio quality so improving it is also important, and that audio systems should be optimized for the time domain response. The yardstick used for evaluating system performance was the speed of decay of the impulse response level. Is 2 dB/μs sufficiently good for loudspeakers? Following lunch Mark Dodd took over the chair's role and Bank described a flexible loudspeaker and room response equalization technique using parallel filters, which are computationally efficient and simple. He specifically compared pole positioning strategies such as predetermined, ripple density, standard warping, multiband warping, and custom warping. This paper was followed by Temme describing how harmonic distortion audibility can be measured using a simplified psychoacoustic model. THD does not correlate with perception, but a perceptually motivated test (PTHD) does work. PEAQ techniques can be used in loudspeaker measurements but the model has to be tuned for the application: take a stimulus, apply ear transfer function and critical bands, add noise and apply masking, then extract results by tuning a neural network. This requires a lot of subjective data. Finally in this session, Rose explored the feasibility of Class D amplifiers for active loudspeaker applications. Class D output stages offer high efficiency and are increasingly accepted for high-quality applications. Today, the best value for low- to-mid-frequencies (woofer amplification in a two-way loudspeaker) can be obtained using Class D amplifiers. Low on-resistance FET switches allow higher switching frequencies compared to earlier designs thereby improving performance. Comparing costs shows that Class D is the cheapest solution when factoring in a heatsink and has the highest efficiency, whereas thick film hybrid designs still have the lowest noise and distortion.

The next session chaired by Jorma Salmi tackled testing and measurement, which started with auralization of signal distortion in audio systems by Klippel. Linear distortion, regular nonlinear distortion and noise, as well as irregular nonlinear distortion, can be added to a clean signal. Linear distortion does not need a model but regular nonlinear distortion does. For rub and buzz, a linear model is needed. In automotive applications it is generally the environment that adds distortion, not the drivers. These distortions can be exaggerated in the auralization, and thus more easily identified, and then a threshold defined for design optimization. NaiZhong then presented a new way to measure the Young's modulus and loss factor of loudspeaker cone materials. A rectangular sample with more than a 10:1 dimension ratio is fixed at one end leaving the other end to move freely. Then a frequency curve is generated and a loss factor calculated using an automatic test system consisting of a displacement laser that moves along the same curve as the cone trajectory. Closing this session, a fast way to measure the parameters of a truncated generalized Maxwell loudspeaker model was


Siegfried Linkwitz (invited speaker) shows a miniature microphone and his miniature dummy head.

presented by Moreno. When comparing a simulation of a driver mounted in a cabinet against real-world constructions, the resonant frequency was correct but the peak value was not, so he considered a better fitting of the mechanical impedance including the viscous losses. A laser-based method with a sine wave stimulus is faster, and the measured results and modeled values matched very well. The completely automated measurement time is five minutes, with further speed improvements expected soon.

To round off the day's presentations Siegfried Linkwitz was invited to describe his search for the ideal stereo loudspeaker. Starting with a mono loudspeaker, a male voice should be perfectly reproduced but to do this it would have to look like a head and torso. Reproduction of a grand piano requires a larger device and places big demands on the dynamic range capabilities. Back in the 1960s sealed cabinet designs were popular and tweeters were not very good. In the early 1970s, ported designs started to appear; they were called "air pressure relief holes." Linkwitz then described how new frequency response measurement techniques were developed before FFT-based tools become prevalent. Together with Riley, he proposed a crossover filter design to replace the commonly used Butterworth filters. Additionally, low-frequency extension was deepened as port design was improved, and an early form of bass management was also developed to cancel turntable vertical rumble. Active loudspeakers were designed, where each driver had its own amplifier driven from a low-level crossover and equalizer, and the bass driver was mechanically clamped at the magnet to damp the high-Q magnet-basket mass-spring system. Then the drivers were equalized and an LR4 crossover added together with an all pass filter. The overall system was equalized for the free-field. Later a tall dipole loudspeaker was found to sound rather like a Quad ESL so a number of smaller line radiator designs followed. Using a scale model of a room and a tweeter equalized flat down to 50 Hz, monopole and dipole radiators could be studied in a 1:10 scale model of a listening room. A two-way loudspeaker design using a PVC tube was created to understand the sound of omnidirectional radiators in a room. The tube has a 40-dB return loss and made a very stable cabinet design. Adding a separate rear tweeter to the ORION design maintained the desired dipole pattern. In nature we are interested in the direction and distance of sources: we segregate signal streams and focus attention. Stereo is an auditory illusion so loudspeaker designers should avoid giving misleading cues. This means that the room sound should have the same timbre as the loudspeaker's direct sound. To do this, room reflections should be symmetrical, delayed (>6 ms, 2 m) and suppressed. There should be diffusion behind the loudspeakers, and the space behind the listener should be open to avoid reflections. Reverberation in untreated rooms is typically more than 0.4 s, which is not a problem for dipole loudspeakers. In terms of signal-to-noise ratio, this is equivalent to a monopole in a much drier room, such as a studio.

From left, Juha Backmann thanks the invited speakers: Axel Grell, Siegfried Linkwitz, and Ilpo Martikainen.


Copious notes were taken.

Ville Pulkki: "Did you just call me a professor, or what?"

In the evening we had a traditional Finnish dinner at the nearby Ravintola Kaisaniemi, which is notable for having a tree growing through the building. Certainly there was plenty of discussion about the day's presentations to be heard around the room.

DAY TWO

Andrew Goldberg briefly introduced Axel Grell from Sennheiser, who described how headphones have developed over time into a mass phenomenon. Headphones are increasingly used everywhere: work, communication, and enjoyment. When did this trend start and what progress has been made since then? The communication breakthrough was Alexander Graham Bell's telephone patent, which showed how sound could be transmitted over a distance, and listening devices were needed for that. The gramophone brought music reproduction to consumers but those devices typically had their own mechanical loudspeakers. In 1922 Baldwin designed and sold 200,000 headphones for listening to radio. This device needed electricity to function and so headphones made sense. Basic elements were already visible: headband, diaphragms, ear pads, etc. Stiff diaphragms were used leading to a harsh sound and much leakage. In 1924 Kellogg and Rice invented the dynamic driver, which leads naturally to the dynamic headphone and dynamic microphone. The Bayer DT 48 was produced from 1937 until 2012. STAX launched the first electrostatic headphone in 1960, the SR 1. Sennheiser invented the HD 414 open headphone in 1968, and over 10 million units were sold in 10 years, which far surpassed the original sales estimate of 2000 units. The driver was based on a microphone diaphragm, and the foam pieces increased comfort and sound quality by allowing some low-frequency leakage to avoid a boomy sound. Next the K 1000 created a sound field close to the head. It did not touch the pinna at all so the user's style was rather compromised. The sound image was very natural but bass reproduction was limited. More recently, the HD 800 is an open headphone using a ring radiator to double the active surface area, but the modes are lower and less well damped, which are overcome using some special remedial actions. Diffuse-field or free-field equalized designs were heavily discussed in the 1980s. A demonstration of the advance in sound quality in headphones showed band-limited (no bass) mono reproduction in the 1930s, by 1972 stereo was possible and there was some bass. In 1993 CD had already been launched but headphones showed little change in reproduction quality, however more recently the HD 800 is quite close to the original recording. Closed headphones designs have also improved greatly over the same time period.

Miikka Tikander then chaired the headphones session, which started with an augmented reality audio system designed by Rämö to capture sounds around the user, equalize them, and reproduce them via in-ear headphones. The goal is to increase sound quality and avoid overexposure to high sound pressure levels at concerts. In-ear headphones provide passive attenuation, and sound quality is recovered using equalization. Leaked sound coming in from the outside brings some comb filtering so only low-latency (< 1ms) DSP is used. Kärkkäinen followed with a description of a practical procedure for large-scale personalized HRTF acquisition. A picture or laser scan is used to create an element model mesh and 150 plane waves are used to simulate HRTFs in the model. Shoulder reflections can be seen and these change as the head moves. Interference patterns can be seen as a plane wave hits the head and torso. The virtual sound source can be moved to any position around the listener. Each frequency takes 150–300 seconds to calculate so a few days of simulation provides a complete picture, but ultimately the goal is to move the computation to the cloud. Next Christensen showed how the magnitude and phase response could be measured at the eardrum. Compared to a set of real ear measurements, an ear simulator overestimates the sound in the 6-kHz region and a tube microphone underestimates it. Phase was split into three areas: linear (delay), minimum (transfer function), and all-pass. Each subject showed big differences but the variation within a headphone was small. Good headphones were deemed to be close to a diffuse-field target. Hoffmann followed by explaining how differences are seen between open and blocked ear canal measurements. This needs to be understood when calibrating insert earphones for hear-through applications. Nonindividual calibrations are preferable but accuracy is compromised. The ear canal response needs to be replicated, and off-the-shelf in-ear headphones need different compensation curves. Two transfer functions are required: blocked entrance eardrum response and earphone-to-eardrum response. Olive concluded the session by presenting a virtual headphone listening test methodology. Double-blind listening tests are time consuming and costly, so recorded virtual headphones are reproduced over an equalized headphone to save time and reduce visual and physical biases. The virtualized headphones showed good agreement with real listening tests (preference and spectral balance) but some differences remain due to factors such as physical biases (weight, clamping force etc.).

"…and the headphones we used in the test were this big!" demonstrated Sean Olive.

In the industry applications session chaired by Asta Kärkkäinen, Marttila described how lumped models are quick to setup and good at low frequency but nonlinearities are missing, whereas an FEM allows all physics and nonlinearities to be included. However FEM is computationally heavy, especially in 3D. Combining the two models brings faster optimizations and realistic results. The spider, magnet and voice coil can be lumped (electronics + magnetic + some mechanics) and FEM can handle the cone, suspension, dust cap, and air (remaining mechanics and acoustics). Next Weckström studied power compression in a miniature speaker box where Vdmax is seen to be the most important parameter. Very thin speakers show thermal compression at 1 kHz, and at an elevated temperature the resistance increases and the resonant frequency drops. Using any available heatsinking helps to avoid these negative effects.

Immediately after the last paper of the day we walked a short distance to the harbor where two boats were waiting to take us around to the Helsinki waterfront. We disembarked to take stroll along the sunny seafront and to find a wind-powered art installation. Then it was back to the boat to travel on towards Tervasaari (Tar Island) for the banquet evening. We were greeted by the Restaurant Savu manager who introduced the building, which was originally constructed in 1805 for storing tar barrels. Smokers were warned not to partake in the weed nearby as there is still some gunpowder remaining in the walls from World War 2. Given the history it is natural that the restaurant specializes in smoke-curing.

After a short "thank you" speech by Sean Olive on behalf of the AES Headquarters we were treated to a live performance from Anna Pudas and Patrik Weckman. The duet played folk tunes from the

"Follow that balloon" to the harbor in order to catch a boat.

countryside using traditional instruments, many of which are not easy to play. They played music from the Middle Ages on the harp, bowed lyre, long flute. Also we were presented Piae Cantiones songs from Finland on the Estonian bagpipes, recorders, pipe, tabor, one-handed flute, and drum. From the 1700s came some Polonaises, Finnish waltzes and marches using a wooden flute, baroque violin, and moraharpa. There were also minuets, marches, waltzes, and Swedish music books played on wooden flute and baroque violin, and some 1800s folk music from Satakunta played on the violin and clarinet. So all in all we had a very wide selection of music that almost no one had heard before played on traditional instruments many of which are rarely heard these days. Certainly this created a lovely atmosphere for further discussions on the day's activities and presentations.

DAY THREE

Juha Backman gave a short tribute to A. Neville Thiele (1920–2012). He had told Juha to experiment, fail, learn, and improve. In Juha's opinion, Neville was the most friendly and kindest person in audio.

The loudspeaker modeling session chaired by Leo Kärkkäinen started with Nisula calculating sound radiation from loudspeaker enclosures using a coupled FEM. Eigenfrequency analysis, loaded mechanical analysis, and acoustical eigenfrequency analysis cannot accurately predict the resonant behavior, but coupled analysis improves the accuracy of modeling. The Helmholtz resonance, internal standing waves, port resonance, and diagonal resonances were found and compared to measurements on a real loudspeaker. The use of coupled modeling is recommended despite the larger computational load. In an overview of simulations Skov stated there are now three legs: theory, measurement, and simulation. Simulations are used to make informed design decisions. Symmetry simplifies the model but cyclic models catch additional information. A full 3D model is computationally expensive but provides a more complete picture. In a nonlinear model, distortion is also seen. Additionally, there are electromagnetic, structural, vibroacoustic, pseudovibroacoustic, and viscothermal models. Analysis types are static, modal, harmonic, and transient. 3D vibroacoustic and 2D/3D vibroacoustic frequency responses are not efficient but this can be overcome using pseudovibroacoustic postprocessing and multitone signals. It is not possible to simulate viscothermal distortion on tweeters. Next Holm demonstrated quantification of diffraction in the time domain using FEM. A Gaussian-windowed 5-kHz sine burst was used to spread the spectrum slightly and a reflection off square corners is clearly visible. Adding a waveguide increases directivity by distributing diffraction along the edge of the waveguide. Rounding the corners reduces diffraction substantially, and in the frequency domain the spectrum of the diffraction is modified. Finally, Cantillon summarized the specifications and reliability of loudspeakers for automotive applications. They must be small (diameter and depth), can only be fitted into certain places, often there is little box volume, and typically f_0 =60 Hz and Q_{ts} >0.6. The speaker and listening positions are suboptimal, but the room acoustics are fixed and known so some consistency is possible. The environment is noisy bringing high LF masking so a high SPL is required with good sound quality. Amplifier output is limited by the single-sided 12-V power rail so a high driver sensitivity is needed. Additionally, vibration, mechanical shocks, liquid spills, dust, salt, and fast temperature (–30 to 130°C) and humidity swings are common. Door-mounted speakers are located in the boundary between wet/dirty (outside) and dry/clean (inside). Driver costs are typically 5–20 USD and then mounted in a 10–100k USD vehicle. Driver failures cause production stops where normally a finished car is produced once a minute and recalls are very expensive. Many environmental tests are done: power tests are for 100 and 1000 hours, temperature and humidity cycling and shocks, mechanical vibration and shocks, drip, spray, dust, chemical tests and complete liquid immersion. Lifetime is expected to be ten years so testing is quite extreme. Statistical data inform the test conditions which typically take two months to do. Certainly this last paper of the conference sparked a lot of interest as most of the delegates are used to designing loudspeaker systems for use indoors.

Top: Finnish traditional music is performed.
Middle: An AES full moon.
Bottom: Ulrik Skov in triumphant mood

A lively poster session saw lots of interest and discussion and the demos proved to be busy during every coffee and lunch break. Genelec demonstrated two-way, three-way, and subwoofer products in the Smart Active Monitoring (SAM) line which enables accurate automatic calibration and alignment of professional monitoring systems in acoustically challenging environments. Sennheiser showed some of its high-end headphones (HD 800, HD 700, Momentum) and the new digital input headphone pre-amplifier (HDVD 800), while Neumann played the critically acclaimed KH 120A two-way and KH 310A three-way active studio monitors. Klippel showed his measurement systems and tools for loudspeaker development and production control. They provide acoustical, electrical, and mechanical measurements, nonlinear and thermal diagnostics, distortion measurement, numerical simulation, and digital auralization. They also evaluate speakers under normal working conditions both at small and high amplitudes and detect the cause of distortions. Pulkki presented Aalto University's spatial sound research. Spatial audio is captured and reproduced using parametric time-frequency-domain techniques adapted for different microphone configurations, and for either loudspeaker or headphone playback. Auditory models are also built to describe the mechanisms in play. Harman presented its virtual headphone listening test system (see paper summary above). Geoff Hill demonstrated how a tetrahedral loudspeaker measurement chamber can reduce variance in driver measurements between suppliers and customers. Skov showed how it is possible do efficient and decomposed 3D vibroacoustic simulation-driven investigations of transducers with nonaxisymmetric geometric featured cones, surrounds, spiders, and back volumes. Ruiz showed how a laser velocity transducer can be used to help find loudspeaker parameters. Only three measurements are required: loudspeaker terminal voltage, voice coil current, and cone velocity. Finally Bank presented his methods for loudspeaker and room response equalization using parallel filters (see paper summary above).

CONCLUSIONS

Juha Backman closed the conference by thanking the AES Headquarters, the AES Finnish Section, the Acoustical Society of Finland, the venue, the sponsors, the organizing committee, the paper writers, and, last but not least, the attendees. As voted for by the conference participants, the most popular paper (with a margin of 10 dB!) was from Mark Dodd. Finally, the huge AES helium balloon used to guide everyone during the walks was released into the blue Helsinki sky as a way to pass the baton onto the organizers of the next “Loudspeakers and Headphones” conference, wherever it might be.

Top right: invited speakers, committee, and attendees on the Paasitorni main staircase.

2nd from top: the organizing committee (left to right): Jukka Saarelma, Lauri Mela, Aki Mäkivirta, Miikka Tikander, Pekka Rönkkö, Asta Kärkkäinen, Juha Backmann, The Balloon, Jussi Rämö, Ilkka Huhtakallio, and Julia Turku (not pictured are Andrew Goldberg and Markus Vaalgamaa).

Right: Juha Backman releases the conference balloon.

Far right: Benedict Slotte, the primary conference photographer, in action.

Editor’s note: You can download papers from this conference from the AES e-library at: http://www.aes.org/e-lib/