Audio Engineering Society

Detailed Conference Program

v1.0

Committee

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Audio Engineering Society

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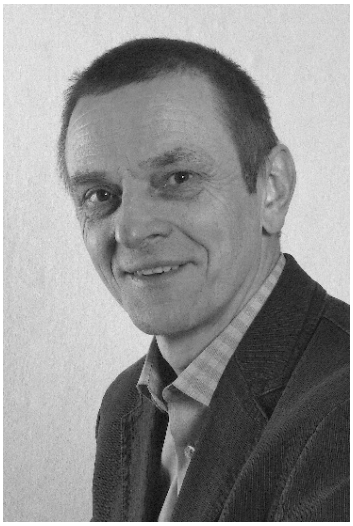
Keynote Talk #1, Wednesday, 9:25 am

Wolfgang Klippel: "Micro-Speakers – Hybrids between headphones and loudspeakers"

ABSTRACT: Micro-speakers based on the electro-dynamical transduction principle show a particular transfer behavior, which cannot be described by conventional models developed for loudspeakers and headphones. This keynote talk presents an advanced lumped parameter model, which considers dominant nonlinearities, time variance, visco-elastic behavior at low frequencies, modal vibration and sound radiation at high frequencies. The relevance of the new model parameters for the design and evaluation of microspeakers and other transducers are discussed.

CV



Wolfgang Klippel studied electrical engineering at the University of Technology in Dresden, in the former East Germany, where his initial studies focused on speech recognition. Afterwards, he joined a loudspeaker company in the eastern part of Germany where he was engaged in transducer modelling, acoustic measurement and psychoacoustics. He later returned to his studies and received a Ph.D in Technical Acoustics in 1987.

After spending a post-doctoral year at the Audio Research Group in Waterloo, Canada and working at Harman/JBL in Northridge, CA, he returned to Dresden in 1997 and founded Klippel GmbH, a company that develops novel control and measurement systems dedicated to loudspeakers and other transducers.

Dr. Klippel has also been engaged as Professor of Electro-Acoustics at the University of Technology in Dresden since 2007. His papers and tutorials on loudspeaker modeling and measurement – particularly those on large signal behavior and physical distortion mechanisms – are considered reference works in the field.

Keynote Talk #2, Thursday, 8:30 am

Volker Hohmann: "Auditory signal processing for assistive listening devices (ALD)"

ABSTRACT: Acoustic communication is ubiquitous and relevant for social interaction and information retrieval in many different acoustic environments, at home, at social gatherings, at the work place and in public spaces such as, e.g., lecture halls, concert halls, train stations, shopping malls, or supermarkets. Electroacoustic systems to support acoustic communication in these environments have evolved rapidly in the last decades, in particular due to the extensive use of digital signal processing in combination with the constantly growing processing power of general-purpose and application-specific signal processors. Today, a large amount of such systems, e.g., announcement systems, home entertainment systems with loudspeaker and headphone presentation, hearing aids and mobile-phone based wearable assistive listening devices ("hearables"), have been or are being developed, which need to function in these vastly different acoustic environments and for users with different requirements, e.g., young and old listeners, as well as hearing-impaired listeners. This talk presents recent research on headphone-based assistive listening devices (ALD) with hearing support for the elderly and the hearing impaired. Signal processing for wearable systems to support binaural spatial perception and speech communication as well as data and methods on how to fit the processing parameters, in particular frequency-specific amplification and dynamic compression, to the individual needs will be presented. Possibilities of implementing the fitting and processing methods on a mobile-phone based hearable with an acoustically transparent headset will be discussed.

CV

Volker Hohmann, PhD, is a professor of Applied Physics at Oldenburg University in Germany and is an internationally renowned expert in auditory modeling and signal processing for hearing devices and assistive listening devices. He is involved in building up and directing several research groups for fundamental, applied and translational research in hearing technology, including the Auditory Signal Processing group at the University of Oldenburg, Hoerzentrum Oldenburg GmbH and HoerTech Oldenburg gGmbH, where he is heading the R&D division.

Dr. Hohmann received the German President's Award for Technology and Innovation in 2012 together with Prof. Dr. Dr. B. Kollmeier (speaker of the award-winning team) and Dr. T. Niederdränk from Siemens AG for their project "Binaural Hearing Aids - Stereo Hearing for Everyone".

Keynote Talk #3, Thursday, 9:20 am

Veronique Larcher: "Some trends and their impact on headphone development"

ABSTRACT: Changes in economic, environmental or social contexts influence the way people use headphones and have therefore a strong impact on the shapes and functions they expect for it. For example, the demand for mobility has lead headphones to take over loudspeakers as the most used playback device to listen to music. In turn, the prevalence of headphones is creating an opportunity for content to adapt and be optimized for binaural playback. While sound quality remains an important driver in selecting a pair of headphones, how users define the quality of their headphone experience very much vary with these evolving uses. We will discuss several of these major influences, as well as some of the technology challenges they create and on-going progress made to address them.

CV

Dr. Veronique Larcher holds a Bachelor's degree in Economics and finances from the Paris institute of Political Studies and a Ph.D. in 3D Audio for Virtual Reality from Ircam (France). In 2001 Dr. Larcher joined Creative Labs in Santa Cruz (USA) as a 3D Audio Scientist and worked for four years on Audio effects for consumer's soundcards and audio interfaces. She went on to join Sennheiser in the Fall 2005 to create and manage their Strategic Innovation office in San Francisco, California, a research entity responsible for incubating and developing new digital technologies and disruptive concepts. Now in Zurich, Veronique is responsible for the AMBEO program, a brand for 3D Audio technology and products by Sennheiser.

Keynote Talk #4, Friday, 8:30 am

Andrew P. Bright: "Headphones, known-knowns and unknown-unknowns"

ABSTRACT: Basic headphone design theory has it that they should emulate the sound field in the ear as experienced when listening to loudspeakers, a live event. It is well known that this approach prevents individual differences in how that sound field should be emulated from being reproduced. This would lead one to conclude that it is unlikely that there could be one set of generally liked headphones; in fact, however, we find that there are. At the same time, headphone users are rarely if ever fooled into thinking sound they hear from headphones is coming from loudspeakers, much less a real-life seagull. Objective measurements give us only partial clues as to why this is so. There are some aspects of headphone design, especially at high frequency, where objective measurements seem to give us no guide at all. Preferred and unpreferred products measure similarly, and preferred products measure differently. A deeper study of wave propagation in the ear canal only seems to increase despondence. Typical ear canal geometries exhibit not only significant axial modes but also radial and circumferential modes – problems not obviously addressed by what is essentially a single point-source at the end of tube. This presentation will summarize what we do and what we don't know about how we can quantify users' preferences for headphones, what it is physically possible and impossible to control, and what users are likely to want from their headphones even if they don't know it.

CV

Andrew P. Bright is a native of Dallas, Texas, received his bachelor's degree in Physics and Philosophy at Lafayette College in Pennsylvania in 1994 and a masters in Acoustics from the ISVR of the University of Southampton in 1995. Andrew worked for Nokia from 1996 until 2009, and while there received a Ph.D. in acoustics from the DTU in Copenhagen in 2002. Andrew helped introduce smart speaker processing and noise cancellation technology while at Nokia. In 2009 Andrew joined Apple Inc. to lead Audio development for the iPhone and other products, introducing multiple smart audio processing technologies into those products, and overseeing the team that designed the Apple Ear Pods.

In 2015 Andrew returned to Denmark, founding Goertek Audio Technologies in Copenhagen, a new research and development centre for audio and other technologies.

Wednesday, August 24

9:00 Registration

9:20 Opening Address

Wed 9:25	**Keynote 1** Wolfgang Klippel	**Micro-Speakers - Hybrids between Headphones and Loudspeakers**	Klippel GmbH & Technical University of Dresden

10:15 Break

Detailed Session Program — Paper Session 1: Headphone Design

Date/ Time	Authors	Title	Affiliation
	Abstract		
Wed 10:40 1-1	Isao G. Anazawa	**Headset EMI Verification and Measurement Method**	NY Works, Ontario, Canada

EMI performance is, other than electro acoustic performance, one of the most important aspects of designing headsets specifically for cellular phones and other portable handset applications. A system to measure the EMI performance of headsets was studied, built and tested with almost two dozen headsets which were pre-packaged and after- market headsets. The results of tests show that the differences of cabling methods greatly influence both EMI and crosstalk performance. The impact of different cabling methods was simulated and is documented herein.

Wed 11:05 1-2	Benichoux V., Chang-Hao C., Tollin D.	**A low-power programmable completely-in-the-canal (CIC) hearing aid for auditory neuroscience**	University of Colorado, CO, USA

This paper documents the development of a miniature programmable completely-in-the-canal hearing aid ("active earplug") aimed as a tool for auditory neuroscience research. The main motivation of this project is to provide researchers with the ability to chronically change chosen aspects of the auditory experience in animals or human subjects. The active earplug is designed around a compact system-on-a-chip package (Belasigna 300, ON Semiconductors, Phoenix, USA) comprising digital converters, preamplifiers and a DSP processor. This chip interfaces with miniature receiver and microphone (Knowles, Itasca, IL, USA) with minimal supporting circuitry. The active earplug is designed to be custom-fitted in each subject's ear canal using a silicone mold. Arbitrary signal processing algorithms can be implemented on the DSP, therefore modifications of the acoustic inputs of the ear with an earplug can be chronically tested. This active earplug will enable researchers to study the effects of chronic expose to arbitrary signal degradations or augmentations, including hearing aid algorithms.

Paper Session 2: Headphone Personalization

Date/ Time	Authors	Title	Affiliation
	Abstract		
Wed 11:30 2-1	Rishabh Ranjan, Jianjun He, and Woon Seng Gan	**Fast Continuous Acquisition of HRTF in 2D for Human Subjects with Unconstrained Random Head Movements**	Nanyang Technological University, Singapore

Head-related transfer function (HRTF) is essential to realize an immersive listening experience over headphones, which is unique for every individual. Conventionally, HRTFs are measured using discrete stop-and-go method for multiple loudspeaker positions, which is a tedious and time consuming process, especially for human subjects. Recently, continuous HRTF acquisition methods have been proposed to improve the acquisition efficiency. However, these methods still require constrained or limited movements of subjects and can only be used in a controlled environment. In this paper, we present a novel fast and continuous HRTF acquisition system that incorporates head-tracker to allow unconstrained head movements in 2D. An improved adaptive filtering approach that combines conventional progressive based normalized least mean square algorithm (NLMS) and previously proposed activated based NLMS is proposed to extract HRTFs on-the-fly from such binaural measurements with random head movements in 2D. Experimental results demonstrate that the proposed approach significantly enhances the performance of conventional progressive NLMS for short duration measurements and further validates the accuracy of HRTF acquisition in 2D.

Date/ Time	Authors	Title	Affiliation
	Abstract		
11:55	**Posters and Demonstrations Preview**		
12:20	**Lunch**		
13:35	**Poster Session & Demonstrations**		
Wed 14:25 2-2	Andrea F. Genovese, Jordan Juras, Chris Miller, Agnieszka Roginska	**The Effect of Elevation on ITD Symmetry**	New York University (NYU), NY, USA
	Typical HRIR modeling techniques ignore the issue of asymmetric head and pinnae characteristics among listeners, thus missing a possibly important correction parameter needed for realistic simulations. In fact, morphological asymmetry is very likely to be a cause of ITD asymmetry. In a previous exploratory study, a common region of sensitivity between datasets of individual HRIR measurement, where the ITD asymmetry is more prominent, was found in the azimuth range of ±90° to ±130° on the horizontal plane. This paper further expands the investigation of ITD asymmetry to the elevation dimension. Two widely used publicly available databases of individually measured HRIRs were selected and analyzed. Due to different measurement techniques and sample rates, the analysis was performed separately for each set. Results found that an increase or decrease in elevation angle; would affect the sensitivity region by reducing the maximum and mean ITD asymmetry value in a roughly linear fashion. This fact implies that the impact of morphological asymmetry is gradually less severe as the elevation angle moves away from the horizontal plane.		
Wed 14:50 2-3	Chris Miller, Jordan Juras, Andrea F. Genovese, and Agnieszka Roginska	**Interaural Distances In Existing HRIR Repositories**	New York University (NYU), NY, USA
	With the recent development of low-cost and efficient methods for generating individualized Interaural Time Differences (ITDs), this paper investigates the distribution of interaural distances among certain populations in order provide a framework for improving the performance of individualized binaural audio systems across a wider range of head morphologies. Interaural distances are extracted from the publicly available LISTEN and CIPIC spatial audio databases in order to generate distributions across subjects, and from the MARL-NYU database in order to investigate measurement stability across testing sessions. The interaural difference is shown to be a means to measure the magnitude of an individual's set of ITDs. Furthermore, the constraints introduced on the precision of measured ITDs by limited sampling rates across all three datasets are explored, and the authors motivate the use of higher sampling rates in the development of spatial audio databases.		
Wed 15:15 2-4	Jianjun He[1], Woon-Seng Gan[2], and Ee-Leng Tan[2]	**Can One „Hear“ the Shape of a Person: Anthropometry Estimation via Head-related Transfer Functions**	[1]Nanyang Technological University, Singapore [2]Beijing Sesame World Technology Co. Ltd, Beijing, China
	Individualized head-related transfer functions (HRTFs) are closely related to anthropometry (measurements of torso, head, and pinna) of listeners. This relation not only derives the individualized HRTFs from anthropometric measurements, but can also be viewed as a means to derive the anthropometry of the listener from his/her measured HRTFs (bypass direct anthropometric measurements). In this study, we propose to estimate a person's anthropometry information using the linear representation obtained from the individualized HRTF features of the person and a HRTF feature database with a number of subjects. Five different HRTF features as well as their best combination are considered in the training stage. Although our experiments showed that the performance of these methods varies in general, the best combination method yields considerable accuracy for the estimation of most anthropometric features. The proposed idea also provides further insights on the complex relation between anthropometry and HRTFs. Our experiment revealed that the anthropometric features that are not well estimated could be removed from HRTF individualization process without causing significant performance degradation.		
15:40	**Break**		
Wed 16:05 2-5	Rishabh Ranjan, and Woon-Seng Gan	**Adaptive Equalization of Natural Augmented Reality Headset Using Non-stationary Virtual Signals**	Nanyang Technological University Singapore, Singapore
	A natural integration of virtual sound sources with the real environment soundscape is discussed in this paper. This natural augmented reality (NAR) headsets consists of dual sensing microphones at each earcup and using adaptive filtering technique to achieve natural listening in augmented reality applications. We propose an online adaptive equalization of the open-end NAR headsets using non-stationary virtual signals to compensate for individualized headphones transfer function (HPTF) and acoustic coupling to seamlessly mix virtual sound with the environmental sound. Training of the NAR headsets are carried out using fast-converging normalized filtered-x least mean square algorithms to respond to changing sound variation. Significant changes in HPTF can be detected online and fast HPTF estimation using normalized least mean square algorithm is employed to update the secondary path estimates.		

Date/ Time	Authors	Title	Affiliation
	Abstract		
Wed 16:30 2-6	Jan Rennies, Dirk Oetting, Hannah Baumgartner, Jens-E. Appell	**User-Interface Concepts for Sound Personalization in Headphones**	Fraunhofer IDMT, Oldenburg, Germany
	Sound personalization is very beneficial, especially for users with impaired hearing. However, classical fitting strategies for hearing instruments involving extensive audiological measurements and fine tuning by trained experts are not applicable for consumer devices such as headphones. This study therefore investigates different user-interface designs enabling users to self-adjust an effective hearing-support algorithm to their individual needs. The large number of parameters of nonlinear and frequency-dependent processing requires a significant reduction of the degrees of freedom, which in this study is achieved by a systematical development of presets covering a large range of typical frequency-dependent hearing-threshold elevations. Experimental validations with hearing-impaired listeners are conducted to assess fitting duration, reproducibility, sound preference, and usability. The results show that a fast and reliable self-fitting is possible when interfaces are used which enable the user to modify algorithmic parameters in real-time in an intuitive and easy way. Despite having very similar kinds of hearing loss, the listeners differ widely in their preferred parameter selection, indicating that self-adjusted sound personalization has a large potential for increasing user satisfaction.		

Paper Session 3: Binaural Technologies

Date/ Time	Authors	Title	Affiliation
	Abstract		
Wed 16:55 3-1	César D. Salvador, Shuichi Sakamoto, Jorge Treviño, and Yôiti Suzuki	**Numerical Evaluation of Binaural Synthesis from Rigid Spherical Microphone Array Recordings**	Research Institute of Electrical Communication, Graduate School of Information Sciences, Tohoku University, Sendai, Japan
	Binaural systems seek to convey a high-definition listening experience by re-creating the sound pressure at both of the listener's ears. The use of a rigid spherical microphone array (RSMA) allows the capture of sound pressure fields for binaural presentation to multiple listeners. The aim of this paper is to objectively address the question on the required resolution for capturing an individual space. We numerically evaluated how binaural synthesis from RSMA recordings is affected when using different numbers of microphones. Evaluations were based on a human head model. Accurate synthesis of spectral cues was possible up to a maximum frequency determined by the number of microphones. Nevertheless, we found that the overall synthesis accuracy could not be indefinitely improved by simply adding more microphones. The limit to the number of microphones beyond which the overall synthesis accuracy did not increase was higher for the interaural spectral cues than for the monaural ones.		
Wed 17:20 3-2	Philipp Stade and Johannes M. Arend	**Perceptual Evaluation of Synthetic Late Binaural Reverberation Based on a Parametric Model**	TH Köln, Cologne, Germany and TU Berlin, Berlin, Germany
	Auralizing rooms with data-based dynamic binaural synthesis is an established approach in virtual acoustics. Generally measured binaural room impulse responses (BRIRs) are used to create a virtual acoustic environment (VAE) over headphones. Depending on the application, it is desirable to reduce the amount of data by decreasing the resolution of the BRIRs. For this reason a scalable parametric model for the synthesis of the binaural late reverberation part was developed and is presented. The model reduces the reverberation tail to three features only. Based on these features, BRIRs with synthetic reverberation are generated and compared to the corresponding measured impulse responses. The synthesis is evaluated perceptually in two listening experiments and differences between several settings of the algorithm as well as the performance for various rooms are examined. The results show only small perceptual differences between original and synthesis even with datasets heavily decreased in size.		
18:30	**Transfer**		
19:00	**Mayor's Reception at Musikkens Hus**		

Thursday, August 25

Thu 8:30	**Keynote 2** Volker Hohmann	**Auditory Signal Processing for Assistive Listening Devices**	Hoertech Oldenburg GmbH & University of Oldenburg
Thu 9:20	**Keynote 3** Veronique Larcher	**Some Trends and Their Impact on Headphone Development**	Sennheiser electronic, Zurich, Switzerland
10:10	**Break**		

Paper Session 4: Standards

Date/ Time	**Authors**	**Title**	**Affiliation**
	Abstract		
Thu 10:35 4-1	Christopher J. Struck	**Refinements in the Electroacoustic Testing of Headphones**	CJS Labs, San Francisco, CA, USA
	A number of measurement methods, processing techniques, and data presentation guidelines for improving the standardized testing of headphones are presented. Selected measurements and specifications from the published standards are reviewed. Areas for appropriate modifications, simplifications, and improvements are identified and explained. Relevant new metrics are also introduced. The rationale for each of these changes is described and examples of the new tests are shown.		
Thu 11:00 4-2	Morten Wille and Per Rasmussen	**IEC 60318-4 Ear Simulator for Low Noise Measurements & Anthropometric Rubber Pinna**	G.R.A.S. Sound & Vibration A/S, Holte, Denmark
	This paper investigates the dynamic limitations of the occluded ear simulator and pinna standardized in the IEC 60318 series. A modified ear simulator is introduced based on a low noise microphone. Also discussed is a new rubber pinna and ear canal based on an average of 260 measured human ear canals. Measurement examples using consumer audio headphones are shown and discussed. Results indicate improved repeatability in measurements as well as the possibility of measuring at sound pressure levels close to and below the threshold of human hearing.		
Thu 11:25 4-3	Günther Theile	**Equalization of Studio Monitor Headphones**	Verband Deutscher Tonmeister, Germany
	The frequency response of high-quality studio monitor headphones should provide the same sound colour neutrality as demanded for loudspeaker monitoring in listening rooms according to ITU-Rec. BS 1116. This is obtained by a probe measured frequency-independent diffuse-field transfer function in accordance with ITU-Rec. BS 708. Spectrum level based calibration requires a reference sound field that provides sufficient diffusity as well as a flat frequency response in order to avoid coloration. Headphone manufacturers are interested in an attractive sound designed in accordance with actual preferences of consumers. Alternative target responses are designed to simulate what a listener hears from a high-quality multichannel loudspeaker system in a reference listening room (in-room equalization). It is shown that this intention can only be realized with binaural room synthesis implementation that ensures accurate binaural rendering of the spatial cues, ideally including head tracking and personalization methods. A corresponding suitable standard based on a neutral listening room is desirable, not least in view of multichannel sound headphone reproduction. The virtual 3D listening room would avoid inadequate in-head perception of suboptimal two-channel stereo downmix material and in addition would provide enhanced reproducibility of results. However, alternative in-room based equalization target curves should be documented with measures according to ITU-Rec. BS 708 Annex 2 that offer clear information for the assessment of tone colour, as well as comparability of headphone frequency responses.		
11:50 12:40	**Lunch Poster Session & Demonstrations**		

Paper Session 5: Perceptual Evaluation

Date/ Time	Authors	Title	Affiliation
	Abstract		
Thu 13:55 5-1	Tore Stegenborg-Andersen	**A Comparison of Sensory Profiles of Headphones Using Real Devices and HATS Recordings**	DELTA SenseLab, Hørsholm, Denmark
	This study compares two sets of sensory profiles of 8 headphones, obtained in two different experiments, with the intent of revealing the differences and or limitations of both methods. The first experiment using a double blind approach with headphone auralizations and the second experiment giving assessors access to the actual headphones, as a non-blind experiment. The results of each experiment are analyzed and compared to reveal the differences, and causes for these differences, for each attribute.		
Thu 14:20 5-2	Christer P. Volk[1,2], Torben H. Pedersen[1], Søren Bech[2,3,], Flemming Christensen[2,]	**Modelling Perceptual Characteristics of Prototype Headphones**	[1]DELTA SenseLab, Hørsholm, Denmark, [2]Aalborg University, Aalborg, Denmark, [3]Bang & Olufsen A/S, Struer, Denmark
	This study tested a framework for modelling of sensory descriptors (words) differentiating headphones. Six descriptors were included in a listening test with recordings of the sound reproductions of seven prototype headphones. A comprehensive data quality analysis investigated both the performance of the listeners and the suitability of the descriptors for modelling. Additionally, two strategies were utilised for modelling metrics describing these descriptors, both relying on specific loudness estimations of the listening test stimuli. The stability of the initially found metrics was tested to quantify the potential of the metrics for future predictions within the perceptual space spanned by the headphones.		
Thu 14:45 5-3	Chris Pike[1,2], Frank Melchior[1], and Anthony Tew[2]	**Descriptive Analysis of Binaural Rendering with Virtual Loudspeakers Using a Rate-All-That-Apply Approach**	[1]BBC Research & Development [2]Audio Lab, Department of Electronics, University of York
	Spatial audio content for headphones is often created using binaural rendering of a virtual loudspeaker array. It is important to understand the effect of this choice on the sound quality. A sensory profiling evaluation was used to assess the perceived differences between direct binaural rendering and virtual loudspeaker rendering of a single sound source with and without head tracking and using anechoic and reverberant binaural impulse responses. A subset of the Spatial Audio Quality Inventory (SAQI) was used. Listeners first selected only attributes that they felt applied to the given stimuli. Initial analysis shows that tone colour and source direction are most affected by the use of this technique, but source extent, distance, and externalisation are also affected. Further work is required to analyse the sparse attribute rating data in depth.		
15:10	**Break**		
Thu 15:35 5-4	Samuel Moulin[1], Søren Bech[1,2] and Tore Stegenborg-Andersen[3]	**Sensory Profiling of High-End Loudspeakers Using Rapid Methods - Part 1: Baseline Experiment Using Headphone Reproduction**	[1]Bang & Olufsen A/S, Struer, Denmark, [2]Aalborg University, Aalborg, Denmark, [3]DELTA SenseLab, Hørsholm, Denmark
	Bang & Olufsen initiated a project to investigate the efficiency and the reliability of different rapid sensory profiling methodologies and auralization systems by conducting parallel listening tests. Each combination of sensory profiling method and rendering system will be studied in a specific listening test. The results will be compared to a baseline experiment in which a conventional descriptive analysis method (QDA) is used together with a dynamic binaural rendering. As a first step, this paper explains the motivation of this project and describes the audio stimuli used in each experiment. This paper also presents the results of the baseline experiment with which future experimental results will be compared with.		

Paper Session 6: Digital Signal Processing

Date/ Time	Authors	Title	Affiliation
	Abstract		
Thu 16:00 6-1	Sean E. Olive, Todd Welti, Omid Khonsaripour	**The Preferred Low Frequency Response of In-Ear Headphones**	Harman International, Northridge, CA, USA
	A series of controlled listening tests were conducted to determine the preferred low frequency response of in-ear (IE) headphones. Using a method of adjustment ten trained listeners adjusted the bass level and frequency of 2nd order low shelving filter applied to a high quality IE headphone equalized to the preferred target response of a circumaural headphone [5]. The adjustments were done for three different music programs, and repeated with and without loudness normalization and control of leakage effects. The influence of program, individual taste, and loudness normalization and leakage effects on preferred low frequency response are presented and discussed.		

Date/ Time	Authors	Title	Affiliation
	Abstract		
Thu 16:25 6-2	Pablo Gutierrez-Parera and Jose J. Lopez	**On the Influence of the Frequency Response over Azimuth Localization with Consumer Headphones**	Universidad Politecnica de Valencia, Valencia, Spain
	High quality headphones can generate a realistic sound immersion reproducing binaural recordings. However, most people commonly use consumer headphones of inferior quality as the ones provided with smartphones or music players. Their frequency response shape and equalization can be a determinant factor in the reproduction of spatial sound. In this study, a perceptual test has been carried out to investigate the effects of the frequency response over the azimuth localization in the horizontal plane. A virtual simulation headphone technique was employed to test seven different high and low quality headphones. Results indicate that much front-back confusion is produced in all cases and that some specific frequency bands have an important role on it. Besides, a poor response in some high frequency bands can affect the lateral positions localization.		
Thu 16:50 6-3	Javier Gómez Bolaños[1], Ville Pulkki[1], and Aki Mäkivirta[2]	**Headphone Stereo Enhancement Using Equalized Binaural Responses to Preserve Headphone Sound Quality**	[1]Aalto University, Espoo, Finland [2]Genelec Oy, Iisalmi, Finland
	A criterion is described and evaluated for equalizing the output of binaural stereo rendering networks in order to preserve the sound quality of the headphone. The aim is to equalize the binaural filter so that the sum of the direct and crosstalk paths from loudspeakers to each ear has flat magnitude response. This equalization criterion is evaluated using a listening test where several binaural filter designs were used. The results show that preserving the differences between the direct and crosstalk paths of a binaural filter is necessary for maintaining the spatial quality of binaural rendering and that post equalization of the binaural filter can preserve the original sound quality of the headphone. Furthermore, post equalization of measured binaural responses was found to better fulfill the expectations of the test participants for virtual presentation of stereo reproduction from loudspeakers. The audio samples used in the tests are available in http://research.spa.aalto.fi/publications/papers/AESheadphones2016/		
Thu 17:15 6-4	Juho Liski[1], Riitta Väänänen[2], Sampo Vesa[2], and Vesa Välimäki[1]	**Adaptive Equalization of Acoustic Transparency in an Augmented Reality Headset**	[1]Aalto University, Espoo, Finland [2]Nokia Technologies, Espoo, Finland
	Headphones are commonly used in noisy environments. Insert headphones attenuate and color the spectra of ambient sounds and thus alter the auditory perception. When the ambient sounds are desirable, a hear-through function can be used to reproduce them naturally while wearing headphones, i.e. to make the headphones acoustically transparent. A novel adaptive hear-through algorithm is proposed, which estimates the isolation and fine-tunes the hear-though equalization for optimal acoustic transparency. Measurements on a prototype headset and simulations show that the proposed algorithm produces acoustical transparency with default settings when the fit is good, and that the adaptation improves the acoustical transparency by up to 6 dB when the headset is poorly fitted. Volume control with additional shelving filter adjustments reduces the comb-filtering effect at frequencies below 1 kHz. The proposed algorithm is a suitable premise for augmented reality audio applications and offers improved behavior when compared to fixed hear-through systems.		
18:10	**Transfer**		
19:00	**Conference Dinner - Robber's Camp Banquet**		

Friday, August 26

Fri 8:30	**Keynote 4** Andrew P. Bright	**Headphones, known-knowns and unknown-unknowns**	Goerteck Audio Technologies, Copenhagen, Denmark

Date/ Time	Authors	Title	Affiliation
	Abstract		

Paper Session 6: Digital Signal Processing (continued)

Fri 9:20 6-5	Jianjun He, and Woon Seng Gan	**Informed Hybrid Primary Ambient Extraction for Spatial Audio Reproduction**	Nanyang Technological University, Singapore

Spatial audio reproduction is essential to create a natural listening experience for digital media. Primary ambient extraction (PAE) is an emerging technique that facilitates an efficient, flexible, and immersive spatial audio reproduction of channel-based audio for any arbitrary playback configurations. This paper presents a hybrid PAE method based on the weighted sum of several existing PAE methods. Using an extraction error performance prior, we derive the optimal weights to combine the candidate PAE methods. Two cases of the error performance are considered: the errors are mutually uncorrelated or partially correlated. Our simulation results indicate that the proposed hybrid PAE method that considers partial error correlation yields even better performance than the best individual PAE method in any cases.

Fri 9:45 6-6	Johannes M. Arend and Christoph Pörschmann	**Audio Watermarking of Binaural Room Impulse Responses**	University of Applied Sciences, Köln, Germany

Digital audio watermarking is an extensive research area, currently with a strong focus on improving established watermarking schemes for conventional audio content. At the same time, headphone-based virtual acoustics, as part of virtual or augmented reality (VR\,/\,AR) technologies, takes on greater significance, and therefore methods are needed to protect the copyright of binaural room impulse responses (BRIRs). To approach this subject, we developed a novel perceptual-based algorithm for watermarking impulse responses. To our knowledge, this is the first study investigating audio watermarking schemes for BRIRs. In a listening experiment, watermark detection thresholds in dynamic binaural synthesis were determined for two rooms and two test signals. A subsequent technical evaluation focused on the robustness against common watermarking attacks. The analysis showed a clear influence of test signal and room on perceptibility and robustness of the watermark. Overall, the results suggest good perceptual transparency and robustness of the proposed method in a reasonable watermark-level range.

10:10 Break

Paper Session 7: Active Noise Cancelling & Listening Comfort

Date/ Time	Authors	Title	Affiliation
	Abstract		
Fri 10:35 7-1	Hatem Röschmann-Foudhaili	**Semi-open ANC Headphone**	Sennheiser Electronic & Co. KG, Wedemark, Germany

In the past, when I was asked about the purpose or benefit of ANC (Active Noise Cancellation) I used to answer “noise damping, isolation, reduction etc”. Receiving the counter argument why not use earmuffs, potentially in combination with earphones, made me realize that the principal benefit of ANC is much more a matter of wearing comfort. Noise isolation as good as in earmuffs for industrial applications is offered by a small, lightweight comfortable headphone. This is realized by augmenting the passive noise isolation of a headphone by active noise reduction. Here, typically a closed headphone is used. Extrapolating the primary benefit of ANC headphones, leads to the vision that ANC headphones in the future should mainly gain in wearing comfort. This paper deals with a first technological approach towards this target. A semi-open headphone with its typical wearing comfort advantages providing the broadband noise isolation of closed earmuffs.

Fri 11:00 7-2	Jay Kirsch, Meenakshi Barjatia, Ajay Iyer, Russell Lambert	**Suppression of Radio Transmission Burst Noise in Headphones**	Harman International, South Jordan, UT, USA

Simplex and half-duplex communication systems (e.g. radios) require the talker to press a Push-To-Talk (PTT) button to enable the transmitter (and disable the receiver). The pressing of this PTT button or the release (when the talker is finished talking) of the button generates a short burst of static noise. This burst noise, although brief, is often louder than the speech, and can be painful when listening in headphones. Algorithms to detect and attenuate the level of the burst are described and tested, including several existing and newly developed algorithms.

Fri 11:25 7-3	Denk, Florian; Kollmeier, Birger; Ernst, Stephan	**High-Fidelity Hearing Instruments: Evaluating Listening Quality of a New Prototype Using a Method for Evaluating Modified Listening (MEML)**	University of Oldenburg, Oldenburg, Germany

A novel prototype of an individualized electronic earpiece providing acoustic transparency, i.e., a sound impression that is perceptually equivalent to the open ear, is evaluated in terms of its listening quality. However, methods allowing for testing such advanced hearing devices in a way that is comprehensive, subjective blind, realistic and easy to use for the subject, are not readily available. We therefore present an according modular framework and use it to evaluate the new prototype operated as a personal assistive listening device in combination with advanced signal enhancement schemes. The introduced evaluation method is directly applicable for testing all hearing devices affecting a listener's live sound perception.

11:50	**Lunch**		
12:40	**Poster Session & Demonstrations**		
Fri 13:55 7-4	Dorte Hammershøi[1], Rodrigo Ordóñez[1], and Anders Tornvig Christensen[2]	Dose Estimate for Personal Music Players Including Earphone Sensitivity and Characteristic	[1]Aalborg University, Aalborg, Denmark [2]Independent

Personal music players can expose their listeners to high sound pressure levels over prolonged periods of time. The risk associated with prolonged listening is not readily available to the listener, and efforts are made to standardize dose estimates that may be displayed for the user. In the present paper, the significance of including the specific sensitivity and frequency response of the earphone type is considered, and a proposal for a revised approach to the implementation of such standardized estimates is assessed. The study is based on simulation of given use-case scenarios using earphone measurements published in the past. The work is on-going.

14:20	**Closing Remarks & Awards**

Posters – Displayed all days

#	Authors	Title	Abstract
	Abstract		
P-1	Guido Baldovino and Michele Geronazzo	**Audio Augmented Reality Headset: A Product Requirements Research in Today's Available Technologies ***	University of Padova, Padova, Italy

In our everyday lives, in every instant, we are surrounded by sound. Unfortunately, technological supports for recording and reproduction have somehow made the audio material artificial, contributing to provide us with a limited listening experience which does not give us any feelings of space and naturalness in sounds as we would be used to perceiving them in reality. Binaural audio technologies have the objective to let us perceive sounds in an acoustic environment which is similar to the way our ears are used to perceive sound. This article gives a brief introduction on the subject of spatial hearing to motivate a product requirements research aimed at identifying a starting technological supports for the implementation of mobile audio augmented reality systems. Five binaural headsets marketed today, which promise to give the consumer a new experience of spatial listening, will be analyzed. Our final objective is to provide a revision of such products in terms of four main criteria: individualization, acoustic-world knowledge, tracking, and ergonomics. We report headset characteristics and limits in order to assess the readiness in future prototypes of audio augmented reality system, where virtual and real sources smoothly merged in one single acoustic scene.

P-2	Todd Welti, Sean Olive, Omid Khonsaripour	**A Validation Study of a Method for Virtualizing In-Ear Headphones ***	Harman International, Northridge, CA, USA

A method for virtualizing in-ear (IE) headphones is described that allows controlled, reliable, double blind listening tests on different models of headphones. Virtualization is achieved by equalizing a replicator headphone to match the frequency responses of the target headphones measured in an IEC 711 coupler. However, since the virtualization process can produce errors in amplitude, phase and nonlinear distortion, a validation study is required to assess their audibility. A panel of trained listeners evaluated binaural recordings made of eleven models IE headphones both real and virtualized versions reproducing different music programs. All of the recordings were reproduced through a high quality reference headphone that was equalized flat at the eardrum reference point (DRP). In this way, visual biases and errors related to headphone fit and leakage were removed. The results show no statistically significant difference in preference ratings between the real and virtualized versions suggesting that the virtualization method is sufficiently accurate to make valid subjective evaluations.

P-3	Dieter Leckschat, Nicolas Suenn, Christian Epe	**Design of a Virtual Sound Reinforcement Laboratory using Headphone Technology ***	Düsseldorf University of Applied Sciences, Düsseldorf, Germany

In the course of re-building University of Applied Sciences Duesseldorf (Germany), a unique laboratory has been planned and built for education and practise in live sound applications. Sound levels which are common in live concerts, however, cannot be applied inside the building for practical reasons. As a solution, a setup was developed which creates a virtual acoustic environment simulating a live performance situation. The system consists of extraaural headphones which are specifically engineered such that the desired high sound levels are achieved with low distortion. The operator´s position is tracked and a dynamic binaural synthesis is applied. The system is complemented by a large shaker-driven floor panel for the low-frequency range.

P-4	Steve Temme	**Challenges of High Resolution Headphone Measurements ***	Listen, Inc., Boston, MA, USA

Hi-Res audio is yet another controversial topic in the audio market, similarly to the importance of high quality audio cables and ultra-low distortion. In this short educational (non-commercial) presentation we will discuss the following: - Industry definition of "Hi Res Audio" - Frequency range of musical instruments and human hearing - Why would we record or measure above 20kHz? - What test equipment is needed to measure above 20kHz – the audio interface, microphone etc. - Examples of headphones that extend beyond 20kHz - New design challenges for making Hi Res audio devices e.g. intermodulation distortion - Conclusions

#	Authors	Title	Abstract
	Abstract		
P-5	Rasmus Jensen, Nikolaj Lauridsen, Andreas Poulsen, Casper Tofte, Flemming Christensen	**Analysis of Subjective Evaluation of User Experience with Headphones**	Aalborg University, Aalborg, Denmark
	The aspects of what provides a good user experience with headphones is initially investigated by an exploratory study (experiment I). Using KJ-Technique, 5 workshop teams of 4-6 participants each provide a number of aspects influencing their experience with headphones. Analysing the aspects for uniqueness and relatedness provides 144 aspects of user experience with headphones, arranged in 12 categories. The 144 influencing aspects from experiment I are condensed, and 24 attributes regarding user experience with headphones are selected. These attributes are tested in regard to their correlation with and effects on overall evaluation of headphones in a second experiment, thus investigating which attributes are most influential for user experience. Using a within-subject design, eight different headphones are evaluated according to the attributes along with an overall evaluation. The attributes are listed in the following categories: sound quality, comfort, build quality, design and brand. A factor analysis shows that the categories fit the attributes. Furthermore, some attributes show high correlations with the overall evaluation, suggesting that these attributes are important for user experience with headphones. The highest rated attributes are shape, design, quality of contact surfaces, comfort, goodness of fit and build quality. An interpretation of which attributes are the most influential in relation to user experience with headphones is discussed.		
P-6	Rodrigo Diaz	**Live panorama and 3-D audio streaming to mobile VR: A Case Study**	Fraunhofer Heinrich-Hertz-Intitute, Berlin, Germany
	Modern mobile virtual reality headsets present an opportunity to display live panoramic video and 3D audio content. In this paper we describe a recent real-world case study where live panoramic video and 3D audio where streamed to the Samsung GearVR virtual reality headset and to the web.		
P-7	David Griesinger	**Accurate timbre and frontal localization by non-invasively equalizing headphones to an individual**	David Griesinger Acoustics, Boston, USA
	Timbre is the vital clue the ear and brain need to localize sounds of all types, but timbre, as perceived by the eardrum, depends dramatically on the pinna, concha, and ear canal resonances that concentrate sound pressure on that surface. For the author and many others this pressure increase can be as much as 18dB at 3000Hz. But these resonances are highly individual, sufficiently so that they can be used as fingerprints. Current measurement techniques for headphones ignore these resonances, and the resonances are sufficiently different for different individuals that a universally accurate equalization for headphones does not exist. We have developed a software application that allows a user to accurately match the timbre of a headphone to that of a frontal loudspeaker, using the user's own eardrum as a microphone. The procedure is simple, painless, quick and inexpensive. The result is accurate timbre and frontal localization for sounds and music of all types. Binaural recordings from microphones equalized the same way can be stunningly realistic		

*** Non-peer-reviewed contribution**

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Demonstrations – Accessible all days

#	Exhibitor	Short Description
D-1	Smyth Research	Loudspeaker reproduction vs. Smyth Realiser of audio formats such as Auro 3D
D-2	BBC Research	Presentation of an interactive VR piece.
D-3	TU Berlin	Presentation of Interactive Recordings of Beethoven Symphonies made with a Motion Tracked Binaural Sound (MTB) - Microphone Array.
D-4	Harman	Virtual headphone evaluation Demonstration.
D-5	Marko Hiipakka, Hefio	Demonstration of the 'Hefio One' self-calibrating headphones. The individual frequency response at the eardrum is measured and actively calibrated to produce a desired frequency response.
D-6	Sennheiser electronic	Ambeo Binaural 3D Audio Demonstration.
D-7	Klippel GmbH	Parametrized Auralization of Distortion Products.
D-8	Nanyang Technological University	Real-time natural 3D sound rendering system for headphones.
D-9	Oldenburg University	High Fidelity Hearing Instruments: Evaluating Listening Quality of a new Prototype using a Method for Evaluating Modified Listening.
D-10	Rodrigo Diaz	A case study in which a concert of the Berlin Philharmonic Orchestra was streamed in panoramic video and 3D audio
D-11	Yamaha	A method for obtaining a generic non-individualized head-related transfer function based on the arithmetic mean of human ear shapes.
D-12	OwnSurround Ltd.	Simulation based personalization of HRTFs.
D-13	AURO TECHNOLOGIES N.V.	Auro-3D and Auro-Matic rendered using Auro-headphones realtime processing.
D-14	Aalto University	Electronic hearing protection for musicians.
D-15	Bragi GmbH	Sensing in the ear - Towards a non-intrusive personal assistant.
D-16	Oxford Digital	Automatic Headphone Equalisation based on measurements using MajE-Fx software.
D-17	Loudsoft	Loudsoft QC: Measurement of active noise control performance