Monday, July 18 **11:00**

OPENING ADDRESS AND KEYNOTE

Sound Field Control: A Brief History —*Philip Nelson,* University of Southampton, UK **(Keynote)**

This presentation will deal with aspects of controlling sound fields through "active" (electroacoustical) intervention with the objectives of either reducing unwanted sound or enhancing wanted sound. The lecture will begin with a review of some examples of work of researchers in the latter part of the nineteenth century and early part of the twentieth century that still have relevance today. An incomplete review of the literature on the subject finds many examples, especially during the twentieth century, of researchers who foresaw potential applications of active methods for the reduction of unwanted noise, but whose efforts were constrained by the available electronic technology. Similarly, although significant practical advances were made in the development of effective methods for the reproduction of sound, it was not until the 1980s that the ready commercial availability of digital signal processors stimulated many new avenues of exploration. The paper will attempt to illuminate the connections between the active control of sound and contemporary approaches to sound reproduction within the consistent framework provided by multichannel digital signal processing and the physical behaviour of linearly superposed sound fields.

12:20

PAPER SESSION 1: SOUND ZONES

1-1 **Planarity-Based Sound Field Optimization for Multi-Listener Spatial Audio**—*Philip Coleman, Philip Jackson*, University of Surrey, Guildford, Surrey, UK

Planarity panning (PP) and planarity control (PC) have previously been shown to be efficient methods for focusing directional sound energy into listening zones. In this paper we consider sound field control for two listeners. First, PP is extended to create spatial audio for two listeners consuming the same spatial audio content. Then, PC is used to create highly directional sound and cancel interfering audio. Simulation results compare PP and PC against pressure matching (PM) solutions. For multiple listeners listening to the same content, PP creates directional sound at lower effort than the PM counterpart. When listeners consume different audio, PC produces greater acoustic contrast than PM, with excellent directional control except for frequencies where grating lobes generate problematic interference patterns.

1-2 **Generalized Singular Value Decomposition for Personalized Audio Using Loudspeaker Array**—*Philippe-Aubert Gauthier*,[1,2] *Yann Pasco*,[1,2] *Alain Berry*[1,2]
[1]GAUS, Groupe d'Acoustique de l'Université de Sherbrooke, Université de Sherbrooke, Sherbrooke, Canada
[2]CIRMMT, Centre for Interdisciplinary Research in Music, Media and Technology, McGill University, Montréal, Canada

Personalized audio is the creation of independent sound zones. The zones are distinguished as the bright zone and the dark zone. The desired audio signal should be audible in the bright zone and reduced in the dark zone. Known methods are the pressure matching method, acoustic contrast maximization, beamforming, and high-pass filtering of cylindrical harmonic expansions. Several challenges are related to personalized audio, including the sound pressure level difference between bright and dark zones. This paper presents a theoretical investigation of a new potential method to achieve personalized audio: Generalized singular value decomposition of multichannel transfer matrices for the automatic creation of source distributions that independently operate on each zone. Results of simulations provide convincing results.

1-3 **Improvement of Personal Sound Zones by Individual Delay Compensation**—*Markus Christoph, Matthias Kronlachner*, Harman/Becker Automotive Systems GmbH, Straubing, Germany

The creation of a personal sound zone (also known as individual sound zone (ISZ)) in car interiors involves certain challenges. Some of these challenges are based on the irregular distribution of the loudspeakers, installed in the car, others with the varying, spectral bandwidth of the same. Another specialty of this environment is, that the personal sound zones are known in advance—they are given by the potential head positions at different seats. Assuming a car with four seat positions, which can be regarded as the normal case, it is obvious that to each of those potential, personal zones some speakers are in close proximity while others are much further away. As was discovered, this variation in the delay time leads to acoustical artifacts, perceivable at the bright zones. Cutting out the common delay (also known as bulk delay) of all room impulse responses (RIS's), which is defined by the most adjacent speaker to all considered personal zones, this problem could somehow be reduced. Thereby the acoustical effect heavily depends on the speaker setup—the more the remaining delay variation, after cutting out the common delay, the less the effect. This holds especially true, if speakers are installed very close to the potential zones, such as speakers in the headrests and/or at the headliner above each zone. Since it is desired to have speakers as close as possible to the potential personal sound zones, in order to enlarge the useful, spectral bandwidth of an ISZ system, we are facing conflicting requirements. In order to solve this conflict, we introduced an individual delay compensation that will be applied to all RIS's to the bright zone(s) prior to the calculation of the ISZ filter sets. The resulting ISZ filter show an improved acoustical performance but cannot be used like this. The resulting ISZ filter have to be time delayed, corresponding to the previously applied, channel dependent, individual delay reduction, before being allowed to be utilized. Thereby the positive acoustical performance does not change. Further it could be proved, that the final resulting ISZ filter show the same acoustical contrast as if calculated without the application of an individual delay compensation.

1-4 Robust Personal Audio Reproduction Based on Acoustic Transfer Function Modelling—*Qiaoxi Zhu*,[1] *Philip Coleman*,[2] *Ming Wu*,[1] *Jun Yang*[1]
[1]Institute of Acoustics, Chinese Academy of Sciences, Beijing, China
[2]University of Surrey, Guildford, Surrey, UK

Personal audio systems generate a local sound field for a listener while attenuating the sound energy at pre-defined quiet zones. Their performance can be sensitive to errors in the acoustic transfer functions between the sources and the zones. In this paper we model the loudspeakers as a superposition of multipoles with a term to describe errors in the actual gain and phase. We then propose a design framework for robust reproduction, incorporating additional prior knowledge about the error distribution where available. We combine acoustic contrast control with worst-case and probability-model optimization, exploiting knowledge of the error distribution. Monte-Carlo simulations over 10,000 test cases show that the method increases system robustness when errors are present in the assumed transfer functions.

14:40

INVITED TALK

Ultrasonic Manipulation—Tweezer Beams—
Bruce Drinkwater, University of Bristol, UK

Ultrasonic manipulation techniques such as acoustical tweezers and sonic tractor beams have attracted significant recent research interest. By carefully controlling the output of arrays of loudspeakers, objects can be either held in place, moved or rotated. An inverse problem is solved that shows that various distinct shapes of acoustic force fields work as tractor beams. The most versatile fields resemble a pair of high intensity ultrasonic fingers. Acoustic vortices also work well and the objects are trapped at the vortex core. Applications at larger scales include container-less processing kidney stone debris removal. At the microscopic scale applications include 3D tissue engineering as well as the assembly of engineering materials such as composites.

15:20

SPONSOR TALK: COMHEAR INC.

15:40

WORKSHOP 1

Creative Applications of Sound Field Control

In recent years there have been significant advances in sound field control technologies. Some of these technologies provide engineering solutions to everyday problems, however, a significant proportion will only be fully exploited when they are adopted by creative practitioners. In this workshop the panel members will motivate a wide discussion on creative applications of sound field control technology, the barriers that might currently be limiting its application, and future routes to support and encourage creative exploitation

17:20

PAPER SESSION 2: HIGH-ORDER AMBISONICS & MODE MATCHING TECHNIQUES

2-1 Efficient Compression and Transportation of Scene-Based Audio for Television Broadcast—*Deep Sen, Nils Peters, Moo Young Kim, Martin Morrell*, Qualcomm Multimedia R&D, San Diego, CA, USA

Scene-based audio is differentiated from channel-based and object-based audio in that it represents a complete soundfield without requiring loudspeaker feeds or audio-objects (with associated meta-data) to recreate the soundfield during playback. Recent activity at MPEG [1], ATSC, and DVB has seen proposals for the use of Higher-Order-Ambisonics (HOA) for scene-based audio. The many benefits of scene-based audio is countered by the bandwidth requirements as well the ability to transport the multitude of HOA coefficient channels through current day Television plants. In this paper we report on research and standardization activities directed at solving these issues. These solutions enable the Television broadcast and delivery of both live-captured and artistically-created sound scenes using HOA.

2-2 Presenting Spatial Sound to Moving Listeners Using High-Order Ambisonics—*Jorge Trevino, Shuichi Sakamoto, Yôiti Suzuki*, Tohoku University, Sendai, Japan

High-order Ambisonics (HOA) presents spatial sound by controlling the sound field inside a compact region. Interactive applications require the listener to move freely; therefore, the listening zone must be shifted accordingly. Dynamic HOA systems face two challenges. First, the angular distribution of the loudspeakers depends on the listener's viewpoint, requiring different decoders as they move. Second, the presented sound field must gradually change, matching what would be heard in the actual world. This research tackles both problems with a modified HOA decoder that makes use of spherical harmonic shift operations. The proposal is evaluated by considering a 157-loudspeaker array. Results show no significant degradation in reproduction accuracy as the listening region is shifted, compared with the accuracy of conventional HOA.

2-3 AmbiFreeverb 2—Development of a 3D Ambisonic Reverb with Spatial Warping and Variable Scattering— *Bruce Wiggins, Mark Dring*, University of Derby, Derby, UK

A new measurement system has been developed for hearing In this paper the development of a three dimensional Ambisonic reverb based on the open source Freeverb algorithm will be presented and discussed. This model is then extended to include processing in over-specified A-format, rather than B-format, variable scattering between channels along with controls for warping the distribution of the reflections to implement a reverb that is able to react to the source position in a spatially coherent way with an acoustical analysis of its performance.

Tuesday, July19 09:00

KEYNOTE

The Psychoacoustics of Reverberation—*Steven van de Par,* University of Oldenburg, Germany **(Keynote)**

On a physical level, reverberation has a significant influence on the acoustic signals that we receive at our eardrums. For our auditory perception, the influence of reverberation often seems to be relatively small. In this presentation some important psychoacoustical principles will be presented by which the auditory system can reduce the detrimental effect of reverberation. These insights will be discussed in the context of modelling speech intelligibility in reverberant environments. An example will be given of how such models of speech intelligibility can be used in sound control to reduce the detrimental effect of reverberation on understanding speech. Finally, also some perceptual consequences of reverberation in the rendering of none-speech audio will be discussed.

10:20

PAPER SESSION 3: PSYCHOACOUSTICS

3-1 Height Loudspeaker Position and its Influence on Listeners' Hedonic Responses—*Sungyoung Kim,*[1] *Mark Indelicato,*[1] *Imamura Hidetaka,*[2] *Hideo Miyazaki*[3]
[1]Rochester Institute of Technology, Rochester, NY,USA
[2]Tokyo University of the Arts, Tokyo, Japan
[3]Yamaha Corporation, Hamamatsu, Japan

Height channels are used to enhance immersiveness and presence of reproduced music. This comparative study focused on possible impact of height-loudspeaker positions, coupled with room acoustics of a listening room, on listener's hedonic responses. Moreover, the study aimed to discover a physical parameter that co-varies with the responses. The results show that the position of height-loudspeakers is a significant factor that differentiated listeners' hedonic responses. To determine the physical parameter affecting listener responses, the authors measured and simulated loudspeaker-to-listener transfer functions. The results show that hedonic responses associated with the height loudspeaker positions were correlated with the ratio between frontal and overhead acoustic energy.

3-2 Perceptually Motivated 3D Diffuse Field Upmixing— *Hyunkook Lee*, University of Huddersfield, Huddersfield, UK

This paper presents a novel 2D to 3D diffuse-field upmixing method developed based on two recent psychoacoustic studies on vertical stereophonic perception: Perceptual Band Allocation (PBA) and phantom image elevation. The PBA renders vertical image spread by allocating each sub-band signal to either lower or upper loudspeaker layer according to the desired vertical position of the band. Horizontally oriented phantom image is perceived above the listener when coherent signals are presented from side or rear loudspeakers, and this effect is most salient for frequency bands centered at 500 Hz and 8 kHz. The proposed upmixing method first splits input signals into sub-bands, each of which is then processed for controlling horizontal image spread (HIS) and the "aboveness" of the image. The resulting signals are upmixed for enhancing vertical image spread (VIS) by PBA. Pilot listening tests conducted using pop, funk rock and classical sources suggest that the proposed method can produce greater HIS and VIS compared to decorrelation-based upmixing methods.

3-3 The Perception of Auditory Height in Individualized and Non-Individualized Dynamic Cross-Talk Cancellation— *Gavin Kearney*, University of York, York, UK

This paper investigates sound source localization, and in particular source elevation in a motion tracked crosstalk cancellation system in comparison to dynamic virtual source rendering over headphones. An experiment is presented where subjects are asked to localize elevated sources using a low-latency motion-tracked cross-talk cancellation system under anechoic conditions using both individualized and non-individualized head related impulse responses (HRIRs). The transaural system utilizes a hybrid Ambisonic and Phantom Sound Source Crosstalk Cancellation (CTC-PSS) approach. Results demonstrate that individualized HRIRs reduce the deviation in localization accuracy in comparison to dummy head HRIRs in the motion tracked transaural system investigated.

3-4 Dynamic CTC with and without Compensation of Early Reflections—*Michael Kohnen, Jonas Stienen, Lukas Aspöck, Michael Vorländer*, RWTH Aachen University, Aachen, Germany

In virtual reality systems such as the aixCAVE (Cave Automated Virtual Environment) multiple screen-walls provide a 3-D image of the virtual world for a tracked listener. Loudspeakers that surround the user, as demanded by VBAP and Ambisonics, would interfere with the user's field of view. Headphone reproduction in contrast lacks a presentation of whole-body vibration for lower frequencies and additional user-attached devices distract the user from the virtual environment. These conditions make crosstalk-cancellation (CTC) systems favorable. As reflections of the acoustically hard surfaces of the screens superimpose the signal of the CTC system, this paper investigates the performance of a multichannel CTC system with compensation of first and second order reflections in a CAVE system.

12:00

POSTERS

P-1 Validation of Sound Field Duplication for Device Testing —*Glenn Dickins,*[1,2] *Prasanga Samarasinghe,*[2] *Thushara Abhayapala*[2]
[1]Dolby Laboratories, McMahons Pt., Australia
[2]The Australian National University

Audio processing in consumer devices is incorporating more spatial information. Performance assessment of such devices requires accurate automated simulation of real world acoustic scenarios. This paper reviews the design and validation of a sound field duplication system for this purpose. Theoretical considerations provide guidance on the design and expected performance; an objective measure of the array setup is used to validate construction; and a subjective test approach is proposed to verify accurate duplication is achieved. The key contribution of the work is to introduce and demonstrate the concept of generational loss for comparative performance assessment of sound field duplication. This approach will be used to further investigate robustness and choose from alternative reconstruction approaches towards creating testing standards.

P-2 Sound Zones: On Performance Prediction of Contrast Control Methods—*Martin Bo Møller,*[1,2] *Martin Olsen*[3]
[1]Aalborg University, Aalborg, Denmark
[2]Bang & Olufsen A/S, Struer, Denmark
[3]Harman Lifestyle Division, Struer, Denmark

Low frequency personal sound zones can be created by controlling the squared sound pressure in separate spatially confined regions. Several methods have been proposed for realizing this scenario, with different constraints and performance. Extrapolating knowledge of the resulting acoustic separation from predicted results is a challenge, since the obtainable performance relies on both the physical setup and the chosen evaluation procedure. In this paper, the influence of the evaluation method is discussed. Using the proposed evaluation, four different control strategies for generation of low frequency sound zones are compared in an experimental study with eight woofers surrounding two control regions.

P-3 Personal Sound Zones: The Significance of Loudspeaker Driver Nonlinear Distortion—*Xiaohui Ma,*[1,2] *Patrick J. Hegarty,*[1] *Jan Abildgaard Pedersen,*[1] *Lars G. Johansen,*[2] *Jakob Juul Larsen*[2]
[1]Dynaudio A/S, Skanderborg, Denmark
[2]Aarhus University, Aarhus N, Denmark

The influence of loudspeaker nonlinear distortion on personal sound zones is studied through simulations under anechoic conditions. Two sound zones, one bright, one dark are created by four linear loudspeaker arrays placed at the edges of a 2.5 m x 2.5 m square. Two methods for controlling the zones, acoustic contrast control and planarity control are employed. Loudspeaker nonlinear distortion is modeled with either second or third order nonlinearities. Without nonlinear distortion, simulations produce a contrast of 80.0 dB for both control methods. When nonlinear distortion is added, the contrast is reduced mainly due to new, uncontrolled components in the dark zone. The impact of nonlinear distortion can be tuned through regularization governing the loudspeaker control effort and the contrast can be optimized.

P-4 Performance Comparison of Filters Designed in Time and Frequency Domains for Personal Audio—*Homin Ryu, Semyung Wang, Kihyun Kim*, Gwangju Institute of Science and Technology, Gwangju, Korea

To realize a personal audio, zone control techniques with loudspeaker array have been used. Previously frequency domain control methods formed the mainstream, however recently, time domain control methods have attracted attention. One of causes is non-causality of frequency domain control results, time domain control can overcome that. This research studied performance comparison of filters designed in time and frequency domains for personal audio through numerical simulations. Filters are designed by same objective function with bright zone response and input power constraints. For comparison of performances, time domain response evaluation is conducted.

12:00

DEMONSTRATIONS

D-1 Bridging Near and Far Acoustical Fields; A Hybrid Systems Approach to Improved Dimensionality in Multi-Listener Spaces—*Peter Otto, Eric Hardman*

D-2 Planarity-Based Sound Field Control and Robust Personal Audio—*Philip Coleman, Qiaoxi Zhu*

D-3 Demonstration of a spherical source with 32 loudspeakers, *Angelo Farina*

D-4 Listener Position Adaptive Loudspeaker Array for Binaural Audio and Personal Audio Reproduction—*Marcos F. Simón Gálvez*

D-5 Listener Position Adaptive Object-Based Stereo—*Dylan Menzies*

D-6 Personal Sound Zones Using a Compact Hemi-Cylindrical Loudspeaker Array—*Falk-Martin Hoffmann*

D-7 Cross-Talk Cancellation for Headrest Sound Reproduction—*Charlie House*

D-8 The Effect of Acoustic Crosstalk on Phantom Image Elevation Over Headphones—*Hyunkook Lee*

D-9 Ultrasonic Levitation—*Bruce Drinkwater*

D-10 Audio Upmixing Using Single-Channel Source Separation—*Gerard Roma*

14:40

PAPER SESSION 4: SOUND FIELD CONTROL THEORIES —PART 1

4-1 Sound Field Control With Hemi-Cylindrical Loudspeaker Arrays—*Falk-Martin Hoffmann,*[1] *Filippo Fazi,*[1] *Simone Fontana*[2]
[1]University of Southampton, Southampton, UK;
[2]Huawei European Research Centre, Munich, Germany

An acoustical model for the sound field generated by hemi-cylindrical loudspeaker arrays is presented and a method for beamforming is derived. The sound field model is obtained by introducing two independent boundary conditions for the sound field of a single impinging plane wave. The model for the radiation from a single loudspeaker in the array is then obtained from the reciprocity principle. Various beam patterns are presented and evaluated as to their dependency on frequency. The obtained theoretical results are discussed, along with a plan for future experimental work.

4-2 Cross-Talk Cancellation and Equalization for Headrest Sound Reproduction—*Stephen Elliott, Charlie House, Jordan Cheer, Marcos F. Simón-Gálvez*, University of Southampton, Southampton, UK

Loudspeakers mounted in the headrest of a seat could potentially be used for spatial sound reproduction in a number of applications. The natural cross-talk cancellation is found to be large in such an arrangement but the response from the loudspeakers to the closest ear depends strongly on the position of the head. In order to improve the reproduction of spatial audio with such a system, various methods of cross-talk cancellation and equalization have been investigated. An averaged cross-talk cancellation system is introduced that minimizes the reproduction error over a number of closely spaced positions, which causes less degradation of the performance away from the assumed head location than exact crosstalk cancellation at one point in space. This is found not to be necessary in this application, however, and a simpler system has been implemented that changes the overall gain of the signals driving the loudspeakers to compensate for the effect of the head position, as measured by a head tracker.

4-3 Comparison of Listener-Centric Sound Field Reproduction Methods in a Convex Optimization Framework—*Andreas Franck, Filippo Maria Fazi*, University of Southampton, Southampton, UK

Sound field control techniques aim at recreating a target sound field within a defined listening region. While this objective can generally be expressed as an optimization problem, the objective function, error norms, and additional constraints used vary widely between the different methods. In this paper we focus on methods that describe a target sound field centered around a single point of the listening region, which includes mode matching, (higher order) Ambisonics but also amplitude panning techniques. By expressing these methods in a common convex optimization framework, we highlight commonalities and differences between the approaches but also the effects of additional constraints such as energy vector maximization, gain non-negativity, and sparsity constraints. These theoretical comparisons are complemented by numerical examples.

4-4 Time Domain Description of Spatial Modes of 2D and 3D Free-Space Greens Functions—*Mark Poletti,*[1] *Thushara D. Abhayapala,*[2] *Paul D. Teal*[3]
[1]Callaghan Innovation, Lower Hutt, New Zealand
[2]Australian National University, Canberra, Australia
[3]Victoria University of Wellington, Wellington, New Zealand

In the design of multichannel sound reproduction systems for semi-reverberant rooms, time-domain solutions allow the use of methods that can shape the combined responses. This is useful for reducing perceptual artifacts. In this paper the time domain functions associated with the radial functions of 2D and 3D point sources are derived. These functions consist of the product of a Bessel and a Hankel function. The time-domain functions provide insight into how the sound field is produced. The solutions are used to develop a mode-based, time-domain approach to sound field reproduction. Comparisons are made with frequency-domain solutions.

16:40

PAPER SESSION 5: SOUND FIELD CONTROL THEORIES—PART 2

5-1 Source-Location-Informed Sound Field Recording and Reproduction: A Generalization to Arrays of Arbitrary Geometry—*Shoichi Koyama*, Graduate School of Information Science and Technology, The University of Tokyo, Japan

A sound field recording and reproduction method that takes into account prior information on the locations of primary sources is proposed. Current recording and reproduction methods are generally based on sound field analysis and synthesis in the spatial Fourier domain. However, the size of the reproduced region and the maximum reproduced frequency are limited by the spatial Nyquist frequency. We previously proposed a source-location-informed sound field recording and reproduction method for several array geometries, which enables higher reproduction accuracy above the spatial Nyquist frequency. We here consider to generalize this method for applying to arbitrary array geometries. It is shown that two requirements must be satisfied by using an example of cylindrical arrays of microphones and loudspeakers.

5-2 Sound Source Modelling and Synthesis by the Equivalent Source Method for Reproducing the Spatial Radiation Characteristics—*Wan-Ho Cho*, Korea Research Institute of Standards and Science, Daejeon, Korea

The method based on the sound source reconstruction using the acoustical holography is proposed to make a model of spatial information of sound source. Also, the concept of modularized source using the combination of designed control filter generating radiation patterns of various ideal sources, having same directional pattern as each order term of spherical harmonics, is applied as a reproduction system for the sound source synthesis. As a demonstration system, a spherical source array consists of 26 loudspeakers is designed to generate the radiation patterns of ideal sources. With the designed system, the radiation patterns of several musical instruments are synthesized.

5-3 Filtering Strategies for Listener-Adaptive Binaural Audio Reproduction over Loudspeaker Arrays—*Marcos Felipe Simón Gálvez, Filippo Maria Fazi*, University of Southampton, Southampton, UK

This paper describes the creation of dynamic filters for listener position adaptive personal audio reproduction with loudspeaker arrays. The proposed filters allow for the delivery of personalized audio programs to a pair of listeners. The filters are adapted in real-time to the listeners' position, which can be estimated by a listener-tracking device. This is obtained by expressing the impulse response of each filter as a network of variable gain-delay elements that are updated so that the filters adjust the reproduced beam pattern to the listeners' position, assuming that each loudspeaker behaves according to a point-source radiation model. This work introduces the formulation of the filters and presents numerical simulations of the system performance.

5-4 Discussing the Applicability of Sound Field Techniques for Larger Audience Entertainment—*Glenn Dickins,*[1,2] *Peter Lennox*[3]
[1]Dolby Laboratories, McMahons Pt., Australia
[2]The Australian National University
[3]University of Derby, Derby, UK

Sound field reconstruction techniques for recreating immersive audio in entertainment applications are well established. However, these techniques and their underlying principles do no readily upscale to cover larger listening areas with a sizable number of either static or ambulant listeners. In this work we review the theory and considerations of sound field control and contrast that to the requirements for creating a consistent experience across a large audience. An argument is made that precise sound field control is neither necessary or sufficient, and we propose key challenges and hybrid approaches for further research and development beyond sound field control.

Wednesday, July 20 **09:00**

KEYNOTE

Differential Microphone Arrays—*Gary Elko,* mh acoustics LLC, Summit, NJ, USA **(Keynote)**

Acoustic noise and reverberation can significantly degrade both the microphone reception and the loudspeaker transmission of speech and other desired acoustic signals. Directional loudspeakers and microphone arrays have proven to be effective in combatting these problems. This talk will cover the design and implementation of a specific class of beamforming microphone arrays that are designed to respond to spatial differentials of the acoustic pressure field in order to attain a desired directional response. These types of beamformers are therefore referred to as differential microphone arrays. By definition, differential arrays are small compared to the acoustic wavelength over their frequency range of operation. Aside from the desirable small size of the differential array, another beneficial feature is that its directional response (beampattern) is generally independent of frequency. Differential arrays are superdirectional arrays since their directivity is typically much higher than that of a uniform delay-and-summed array having the same geometry. It is well known that superdirectional arrays are subject to implementation robustness issues such as microphone amplitude and phase mismatch, sensitivity to microphone self-noise, and input circuit noise. Thus, the design of practical differential microphone arrays can be cast as an optimization problem for a desired beampattern response with constraints on the amount of SNR loss through the beamformer. Spherical differential arrays have been of interest for the spatial recording of sound fields for over 40 years. These arrays were initially first-order systems but higher-order spherical arrays are becoming available and are an active area of research for spatial sound pickup and playback. Several microphone array geometries that use robustness design constraints covering multiple-order differential arrays will be shown.

10:20

PAPER SESSION 6: MICROPHONE ARRAYS FOR SOUND FIELD CAPTURING

6-1 A Qualitative Analysis of Frequency Dependencies in Ambisonics Decoding Related to Spherical Microphone Array Recording—*Jens Meyer, Gary Elko*, mh acoustics, Summit, NJ, USA

Encoding 3D spatial audio signals by spherical microphone array eigenbeamforming which is closely related to Higher-order Ambisonics (HOA), is a growing field of interest. These spatial sound field encoding schemes allow for spatially realistic recording and playback of multichannel audio via loudspeakers and headphones. In the simplest form of HOA signal encoding, a virtual sound source is positioned at any angle in space. The source is then encoded into Ambisonic signals using simple scalar multiplications that are based on spherical harmonics that depend only on the source location. For playback the Ambisonic signals are decoded with a decoding matrix that depends on the loudspeaker positions. Both encoding and decoding are simple vector operations that are frequency independent. However, things are different when dealing with Ambisonic recordings from an eigenbeamforming spherical (or similar) microphone array. Due to the physics of acoustic wave propagation the sound field captured by the microphone arrays for HOA processing will inherently introduce frequency dependency into the HOA signals. Fortunately the inherent frequency dependency can be generally compensated by applying appropriate post-filters. For higher orders of the recorded Ambisonic signals (Eigenbeams), some frequency dependency will naturally remain. The main object of this paper is to describe the frequency dependency and compensation of the recorded Ambisonic signals and present a qualitative analysis of its effects on the decoder. Finally, we will suggest some measures for the decoder to account for this frequency dependency.

6-2 Microphone Arrays for Vertical Imaging and Three-Dimensional Capture of Acoustic Instruments—
Bryan Martin,[1,2,3] *Richard King*,[1,2] *Wieslaw Woszczyk*[1]
[1]McGill University, Quebec, Canada
[2]Centre for Interdisciplinary Research in Music Media and Technology (CIRMMT), Montreal, Quebec, Canada
3Enhanced Reality Audio Group (ERA), Montreal, Quebec, Canada

A great deal of research is currently underway in spatial sound reproduction through computer modeling and signal processing, while less focus is being placed on actual recording practice. This study compares a selection of microphone arrays for music recording to create vertical and three-dimensional images without the use of added processing. Three separate arrays were investigated in this study: 1. Coincident, 2. M/S-XYZ, and 3. Noncoincident. Instruments of the orchestral string, woodwind, and brass sections were recorded with each array. Two Triad/ABX listening tests were conducted to determine if the subjects could distinguish the arrays from a level-matched mono/point-source presentation and if they could distinguish the arrays from one another. The results of the listening tests strongly suggest that the subjects could discern the difference between the three arrays and the mono/point-source signal, and also from one another.

6-3 Comparing Ambisonic Microphones: Part 1—
Enda Bates,[1] *Marcin Gorzel*,[2] *Luke Ferguson*,[1] *Hugh O'Dwyer*,[1] *Francis M. Boland*[1]
[1]Trinity College Dublin, Dublin, Ireland
[2]Google, Dublin, Ireland

This paper presents some initial experiments devised to assess the performance of a number of commercially available microphones in capturing 360 audio. The subjective audio quality of four microphones (Soundfield MKV, Core Sound TetraMic, MH Acoustics Eigenmike, and Zoom H2n) is assessed using modified Multiple Stimuli with Hidden Reference and Anchor (MUSHRA) tests. The localization accuracy is assessed using an objective directional analysis of recordings made with a spherical array of 16 loudspeakers. Intensity vectors are extracted from 25 critical frequency bands and are used to compute the directivity and diffuseness of each recorded signal. Both studies reveal significant differences between the microphones.

12:00

WORKSHOP 2

Linking Sound Field Capture to Sound Field Reproduction

Many approaches exist to capture and reproduce sound fields, and these often have a similar theoretical basis. However, research into the two areas is often independent. Moreover, sound field control techniques sometimes require content to be captured using a certain approach. This workshop will explore the relationships and interconnections between techniques to capture and reproduce sound fields, the implementation challenges and perceptual implications of these techniques, and future opportunities arising from an integrated perspective on sound field capture and reproduction.

14:00

PAPER SESSION 7: EMERGING TECHNIQUES AND APPLICATIONS WITH ARRAY TRANSDUCERS

7-1 Bridging Near and Far Acoustic Fields: A Hybrid System Approach to Improved Dimensionality in Multi-Listener Spaces—*Eric Hamdan, Andrew Allen, Raphael Melgar, Peter Otto*, Qualcomm Institute, University of California San Diego, CA, USA

Historically, far-field loudspeaker systems have been challenged to provide convincing near-field auditory images for multiple listeners, e.g., realistic buzzing of bees perceived closely around a listeners head. Conversely, experience with recent near-field systems in the form of compact transaural beamformers suggests these personal loudspeaker arrays do not produce satisfying far-field auditory images for multiple listeners; large audiences are a particularly problematic case. In this article the authors propose a hybrid system that consolidates the near-field performance of compact transaural beamformers and the far-field qualities of large loudspeaker installations typically employed in cinemas and theme parks. A first approach to combine these two systems is defined. We present a brief overview of the design of digital filters required by a loudspeaker array to form transaural beams for two listeners. Analytical results are presented.

7-2 Measuring Spatial MIMO Impulse Responses in Rooms Employing Spherical Transducer Arrays—*Angelo Farina, Lorenzo Chiesi*, University of Parma, Parma, Italy

The paper presents a new measurement method aimed to characterize completely the sound propagation from a point to another point inside a room, taking into account the directionality of the source and of the receiver. The method makes use of two spherical arrays of transducers, almost-uniformly scattered on the surface of a rigid sphere, which can synthesize arbitrary polar patterns. The paper describes the beamforming method employed for synthesizing the required polar patterns over a wide frequency range, and how to process the results in various ways: graphical mapping of sound reflections inside the room, reconstructing the trajectories of such reflections, and auralization when for the first time both source and receiver can be allowed to freely rotate during the test.

7-3 A Self-Configurable, Wireless Audio System with User-Tracking Ability—*Jung-Woo Choi, Jungju Jee*, Korea Advanced Institute of Science and Technology (KAIST), Daejeon, Korea

Multichannel audio systems are often installed in arbitrary layouts, which are completely different from predefined positions in audio standards. Automatic identification techniques for loudspeaker layouts and user locations have been developed for various audio systems, but many of them utilize microphones positioned near the user's side. In this work we propose a method for constructing smart audio systems that automatically identify loudspeaker layouts and user locations using a microphone embedded in itself. The identification of loudspeaker layout is done under asynchronous condition, in which each loudspeaker is connected via a wireless network with non-identical time references, and a user location is identified using acoustic scattering characteristics of human body.

7-4 Reshaping of Room Impulse Responses over Wireless Acoustic Networks—*Gema Piñero, Juan Estreder, Francisco Martínez-Zaldívar, María de Diego, Miguel Ferrer*, Polytechnic University of Valencia, Valencia, Spain

The sound inside an enclosure is always perturbed by the room impulse response (RIR) between the source and the listener. The RIR reshaping technique processes the sound reproduced by the loudspeakers to obtain a desired RIR. This paper reformulates the RIR reshaping technique based on p-norm optimization algorithm in order to implement it on a wireless acoustic network that runs a personal sound zone system (PSZS). A distributed computation of the original algorithm is proposed. Experimental results of a real PSZS implemented on an acoustic network of commercial devices are presented and compared to those of the RIR reshaping algorithm. The new RIRs slightly improve the attenuation between zones and eliminate the late echoes of the real signals.

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