AES 52ND INTERNATIONAL CONFERENCE

Sound Field Control
Engineering and Perception

University of Surrey, Guildford, UK
2–4 September, 2013

Technical Sessions

This preliminary program is accurate as of press time. See updates at www.aes.org/conferences/52

Monday, September 2 — 10:40

OPENING REMARKS AND KEYNOTE SPEECH

[Keynote] Active Control of Sound Fields: Enhancing Signals and Attenuating Noise—*Steve Elliott*, ISVR University of Southampton, Southampton, UK

In my opening remarks the acoustics and signal processing involved in active control will be reviewed: both for enhancing audio signals and for attenuating acoustic noise. The physical limits due to the geometry of practical arrays will be discussed, together with the signal processing design methods that can be used to implement robust practical arrays, whose performance approaches these geometric limits.

Monday, September 2 — 13:40

PAPER SESSION 1: SOUND FIELD CONTROL THEORY AND APPLICATIONS—PART 1

1-1 [Invited] Source-Width Extension Technique for Sound Field Reproduction Systems—*Jung-Woo Choi*, Korea Advanced Institute of Science and Technology (KAIST), Daejeon, Korea

A source width extension technique for the sound field reproduction system is proposed. The extension of the perceived source width has been extensively discussed for the discrete multichannel systems, and many of them make use of the all-pass filters and pseudo-stereo technique to decorrelate two signals at the listener's ear positions. However, for the sound field reproduction, such property is needed to be realized at the various listener positions in space. In this paper we attempt to design a sound field that produces decorrelated ear signals at various listener positions. The radiation pattern from a virtual sound source is modified such that any listener heading to the virtual source location can experience the same amount of source-widening effect.

1-2 [Invited] Design of a Source Array for the Rendering of a Desired Sound Field Using the Equivalent Source Method—*Wan-Ho Cho*,[1] *Jeong-Guon Ih*[2]
[1]Korea Research Institute of Standards and Science (KRISS), Korea
[2]Korea Advanced Institute of Science and Technology (KAIST), Korea

A source design approach using the equivalent source method (ESM), one of the representative acoustical holography methods was suggested to generate a desired sound field. The reformulation of transfer matrix constructed by the spherical harmonic function was suggested for the design purpose. Also, the method to make a model of source with equivalent source was considered to construct an efficient model. In the example of multi-zone control with 16 sources, the suggested method shows similar accuracy and stability to the inverse BEM-based approach. Moreover, the suggested method was applied to inverse design of the modular array system that is the multi-source system allocated to a single channel with pre-determined controller. If the modular sources are constructed properly, the modular source approach is possible to obtain a stable solution against to the noise condition including discrepancy in comparison with the case using same number of sources independently.

1-3 [Invited] Is Sound Field Control Determined at All Frequencies? How Is it Related to Numerical Acoustics?—*Franz Zotter*,[1] *Sascha Spors*[2]
[1]University of Music and Performing Arts Graz, Graz, Austria
[2]University of Rostock, Rostock, Germany

Long before audio technologists have been researching the acoustic reproduction of entire sound fields using loudspeaker arrays, numerical acousticians began to study the solution of boundary integral equations. This is particularly interesting because the fundamental question of uniqueness also appeared in numerical acoustics much earlier than it recently did in the theory of sound field synthesis, where it still appears to be pending. There were two main approaches that proved non-uniqueness at certain frequencies to be soluble by enforcing all the necessary mathematical constraints. Both are directly applicable to generally ensure uniqueness in the sound field synthesis theory based on free field monopole sources.

Monday, September 2 **15:20**

PAPER SESSION 2: SOUND FIELD CONTROL THEORY AND APPLICATIONS—PART 2

2-1 **[Invited] Sound Field Reproduction of Real Flight Recordings in Cabin Mock-up**—*Philippe-Aubert Gauthier, Cédric Camier, Olivier Gauthier, Yann Pasco, Alain Berry*, Université de Sherbrooke, Sherbrooke, Ontario, Canada, and CIRMMT, McGill University, Montreal, Quebec, Canada

Sound environment reproduction of various flight conditions in aircraft mock-ups is a valuable tool for the study, prediction, demonstration, and jury testing of interior aircraft sound quality. To provide a faithful reproduced sound environment, time, frequency, and spatial characteristics should be preserved. Physical sound field reproduction approaches for spatial sound reproduction are mandatory to immerse the listener body in the proper sound field so that localization cues are recreated. For sound field reproduction inside cabin mock-up, our approach relies on multichannel equalization using least-mean-square formulation. In this paper a modified multichannel equalization procedure is proposed to simplify the selection of reproduction sources regularization. The paper presents objective evaluations of reproduced sound fields on the basis of real flight recordings using an 80-channel microphone array and 41 actuators in the cabin mock-up.

2-2 **On the Potential for Scene Analysis from Compact Microphone Arrays**—*Glenn Dickins,*[1] *David Gunawan,*[1] *Dong Shi*[2]
[1]Dolby Laboratories, Sydney, NSW, Australia
[2]Dolby Laboratories, Beijing, China

Identifying acoustic objects from a low order compact microphone array is a challenge. Of interest is deriving context from voice activity in a reverberant room from a table top device; many assumptions of the classic problem formulation are violated by broadband, high dynamic range, reverberation and moving sources. Office noise is rarely stationary and not easily classified as desirable or detracting for the capture. Intuitive constraints of a real meeting environment, higher order statistics, and longer term observation show promise to achieve useful performance in an embedded demonstration system. Using a simple statistical framework, physical source object modeling, and operational heuristics it is possible to decompose a meeting scene with low latency from an array of three co-incident directional microphones.

2-3 **Acoustic Element and Array Design Approaches**—*Graeme Huon*, HuonLabs Pty. Ltd., Melbourne, Australia

Wave propagation theory relevant to array design is summarized. Broad-fire and End-fire array types are considered with regard to array element directivity and point source behavior in particular. Factors including the aperture limit, spatial aliasing, truncation effects, and driver directivity control techniques are considered and strategies summarized. A novel approach to driver design allowing flexible control over directivity at manufacture is then explored and a prototype reported on. It is verified that the aperture limit can be exceeded. The implications for array design and reciprocal capture approaches are considered. Patent applications apply to aspects of this work.

Monday, September 2 **16:20**

WORKSHOP 1

The Interplay between Engineering and Perception in the Design of Sound Systems for Listeners

Moderator: **Frank Melchior,** BBC Research & Development, MediaCity, Salford, UK

Panelists: *Jung-Woo Choi*, Korea Advanced Institute of Science and Technology (KAIST), Daejeon, Korea
Glenn Dickins, Dolby Laboratories, Sydney, NSW, Australia
Armin Kohlrausch, Philips Group Innovation and Eindhoven University of Technology, Eindhoven, The Netherlands

One need look no further than MP3 for an example in audio engineering where incorporating consideration of the listener has led to engineering design that the consumer has embraced, with huge benefits in terms of practical products and services, as well as investment in psychoacoustical science. This workshop examines the interplay between technical and subjective aspects of audio system design, which may be treated as distinct yet complementary disciplines or increasingly as a powerful inter-disciplinary enterprise.

Monday, September 2 **17:20**

POSTER PREVIEW

Monday, September 2 **18:30**

POSTERS

P-1 **GPU-Based WFS Systems with Mobile Virtual Sound Sources and Room Compensation**—*Jose A. Belloch, Miguel Ferrer, Alberto Gonzalez, Antonio M. Vidal*, Universitat Politecnica de Valencia, Valencia, Spain

Wave Field Synthesis (WFS) is a spatial audio reproduction system that provides an accurate spatial sound field in a wide area. This sound field is rendered through a high number of loudspeakers to emulate virtual sound sources. WFS systems require high computational capacity since they involve multiple loudspeakers and multiple virtual sources. Furthermore improvements of the spatial audio perception imply even higher processing capacity, mainly to avoid artifacts when the virtual sources move, and compensate the room effects at certain control points within the listening area. Graphics Processing Units (GPUs) are well-known for their potential in highly parallel data processing. In this paper we propose a GPU implementation that yields maximum parallelism by adapting the required computations to the different GPU architectures (Tesla, Fermi and Kepler).

P-2 **Alternate Sound Reproduction Formats**—*Graeme Huon, Stephen Huon*, HuonLabs Pty. Ltd., Melbourne, Australia

Loudspeaker formats that can maintain imagery with observer movement and turning are investigated. Simple array configurations using controlled directivity sources are used to create shaped wave-fronts that allow placed broadband sources to be represented over significant solid angles. The perception of fields of placed sources that remain so with observer movement and turning is explained with reference to the ITD/IID perceptual model.

It is found that significant improvements over present equidistant formats can be achieved. Prototype designs and practical issues are reported. The implications for a new generation of reproduction formats and related capture issues are considered. Patent applications apply to aspects of this work.

P-3 Real-Time Sound Field Transmission System by Using Wave Field Reconstruction Filter and Its Subjective Listening Test—*Shoichi Koyama,*[1] *Ken'ichi Furuya,*[2] *Hisashi Uematsu,*[1] *Yusuke Hiwasaki,*[1] *Yoichi Haneda*[3]
[1]Nippon Telegraph and Telephone Corporation, Tokyo, Japan
[2]Oita University, Oita, Japan
[3]The University of Electro-Communications, Tokyo, Japan

For real-time transmission of a sound field in a large area, it is necessary to transform received signals of a microphone array into driving signals of a loudspeaker array to reproduce the sound field. We previously proposed a signal transform method for planar or linear arrays of microphones and loudspeakers that uses an analytically derived filter, called the wave field reconstruction filter (WFR filter). By using the WFR filter for linear arrays with 64 channels, we developed a prototype real-time sound field transmission system over an IP network. The results of the subjective listening tests indicated that the source localization accuracy for virtual sound sources is comparable to that for real sound sources in a large listening area.

P-4 Loudspeaker Array Processing for Multi-Zone Audio Reproduction Based on Analytical and Measured Electroacoustical Transfer Functions—*Ferdinando Olivieri,*[1] *Mincheol Shin,*[1] *Filippo Fazi,*[1] *Philip A. Nelson,*[1] *Peter Otto*[2]
[1]Institute of Sound and Vibration Research, Southampton, UK
[2]University of California San Diego, San Diego, CA, USA

Methods for generating multiple zones of sound with loudspeaker arrays have been implemented by using various control algorithms. In the sound control process, the electroacoustical transfer functions between the loudspeakers and the control points in the target region are important because they are used for the computation of the digital filters used to generate the input to the loudspeakers. For a given array, acoustic control performance depends mainly on these digital filters. The transfer functions of a prototype linear loudspeaker array composed of 16 loudspeakers are measured in specific acoustic environments and also calculated analytically assuming a model for the directivity of the individual loudspeakers. Using one multi-zone sound field control algorithm, the performance of multi-zone audio reproduction based on these different sets of transfer functions has been analyzed by computer simulations and experiments.

P-5 The Uncanny Valley of Spatial Voice—*Glenn Dickins,*[1] *Xuejing Sun,*[2] *Richard Cartwright,*[1] *David Gunawan*[1]
[1]Dolby Laboratories, Sydney, NSW, Australia
[2]Dolby Laboratories, Beijing, China

In computer animation there is a known dip in comfort with increasing fidelity or likeness of a human image. A similar subjective phenomenon has implications to spatial voice. A suggested explanation is that large errors tend to create a sense of separation and associate with channel degradation - for example our tolerance of the conventional phone. Smaller errors, as the channel improves, may be ascribed to the source or person speaking triggering a sense of unease. Practical constraints often lead to distortion in the capture, transport and reproduction of voice; certain attempts to disguise and mask distortion may lead into the 'uncanny valley'. This paper combines a subject review, proposed framework and set of observations and examples for discussion.

P-6 Sound Field Rendering for Distributed Audiences—*Gavin Kearney*, University of York, York, UK

In this paper an investigation into the localization accuracy of Higher Order Ambisonic sound fields with reference to Vector Based Amplitude Panning (VBAP) is presented. The analysis is considered in relation to listeners who are not located in the acoustic sweet spot. The performance of velocity, energy, in-phase, and psychoacoustically optimized Ambisonic decoders are compared over a horizontal-only 8-channel array. Results demonstrate that 3rd order Ambisonics with energy decode gives equivalent localization performance to VBAP for off-center listening.

P-7 Data-Based Binaural Synthesis including Rotational and Translatory Head-Movements—*Frank Schultz, Sascha Spors*, Universität Rostock, Rostock, Germany

Several approaches to data-based binaural synthesis have been published that are based on the analysis of sound fields captured by spherical microphone arrays. The captured sound field is decomposed into plane waves which are then auralized with head-related transfer functions (HRTFs). So far head-rotations of the listener were considered for dynamic binaural synthesis. We propose an analytic method to consider translatory head-movements as well. A straightforward calculus in the spatial frequency domain is presented and will be evaluated.

P-8 GPU Implementation of a Frequency-Domain Modified Filtered-x LMS Algorithm for Multichannel Local Active Noise Control—*Jorge Lorente, Miguel Ferrer, Maria De Diego, Jose A. Belloch, Alberto Gonzalez*, Universitat Politecnica de Valencia, Valencia, Spain

Multichannel active noise control (ANC) systems are commonly based on adaptive signal processing algorithms that require high computational capacity, which constraints their practical implementation. Graphics Processing Units (GPUs) are well known for their potential for highly parallel data processing. Therefore, GPUs seem to be a suitable platform for multichannel scenarios. However, efficient use of parallel computation in the adaptive filtering context is not straightforward due to the feedback loops. This paper presents a GPU implementation of a multichannel feedforward local ANC system based on the modified filtered-x LMS algorithm working over a real-time prototype. Details regarding the parallelization of the algorithm are given. Experimental results are presented to validate the real-time performance of the multichannel ANC GPU implementation.

P-9 Objective Evaluation of Sound Field and Sound Environment Reproduction in Aircraft Mock-Ups Using Acoustic Imaging—*Philippe-Aubert Gauthier,*[1] *Cédric Camier,*[1] *Thomas Padois,*[1,2] *Olivier Gauthier,*[1] *Yann Pasco,*[1] *Alain Berry*[1]
[1]Université de Sherbrooke, Scherbrooke, Ontario, Canada
[2]McGill University, Montreal, Quebec, Canada

Audio is also an active topic for acousticians in the aircraft manufacturing industry. Typical concerns are sound quality, noise annoyance or realism, in case of flight simulators. Accordingly, sound field reproduction of real flight conditions in aircraft mock-ups is a valuable tool for jury testing, listening test or marketing purposes. Recently, a cabin mock-up was developed to achieve multichannel sound field reproduction. In this paper acoustic imaging with an 80-channel microphone array is applied to the objective evaluation, in terms of spatial sound accuracy, of sound field reproduction in the mock-up. Two imaging algorithms are tested, compared, and discussed with respect to their ability to provide meaningful attributes for an objective evaluation and comparison of sound field reproduction performance of environmental source distributions. General conclusions about the applicability of acoustic imaging and tested algorithms for the spatial quality assessment of spatial reproduction systems are also provided.

P-10 Relaxation Effects of Binaural Phenomena—*Zlatko Baracskai, Saoirse Finn,* Birmingham City University, Birmingham, UK

The aim of this study is to identify tendencies in the effectiveness of relaxing audio stimuli that could be verified through further focused experiments. A series of brainwave entrainment (BWE) techniques for inducing relaxation will be presented consisting of different binaural phenomena (BP). The BP will derive from the binaural sine wave beat, widely acknowledged in rhythmic BWE literature. New methods of entrainment will include binaural amplitude and precedence modulation, which will be compared to a monaural drum beat and environmental soundscapes. Their ability to induce relaxation will be assessed using Skin Conductance Level (SCL) to measure physiological changes in the body.

P-11 Sound Field Simulation Using Extrapolated Loudspeaker Impulse Responses—*Nara Hahn,* Seoul National University, Seoul, Korea

In the area of sound field synthesis, reproduced sound fields are commonly visualized using simulations. In such simulations, the loudspeakers are often modeled as simplified acoustic sources that are analytically tractable, such as point sources. This simplification limits the accuracy of the simulation, particularly at high frequencies. Considering that sound field synthesis methods also suffer from artifacts at high frequencies, it is desirable to simulate the reproduced sound fields more accurately. The aim of this paper is to simulate the sound fields using the impulse responses of a real loudspeaker. The spatio-temporal impulse responses were sampled at discrete positions and were used to obtain the impulse responses for other positions. This extrapolation method was based on the Kirchhoff-Helmholtz integral equation. The proposed method can be used to visualize the detailed structure of the reproduced sound fields. As evidence of this, some exemplary simulation results are presented herein.

Tuesday, September 3 09:00

KEYNOTE SPEECH

Evaluation of Spatial Sound Fields: How Far Can We Get with Perceptual Models?—*Armin Kohlrausch*, Philips Group Innovation and Eindhoven University of Technology, Eindhoven, The Netherlands

The development of sound reproduction systems is usually influenced by knowledge of human auditory perception. First, this perception knowledge provides input to the definition of system requirements, e.g., when defining limits for the frequency range. Second, it plays a role in the evaluation of a given solution, for instance, by using perception models as a replacement for human listeners. With the development of more and more advanced methods of sound field creation and sound field control the requirement for perceptual models has also grown, in particular with respect to spatial sound field characteristics. In this presentation, I will give an overview of recent developments in the modeling of spatial hearing and give examples of what type of spatial sound characteristics have successfully been evaluated by purely algorithmic means.

Tuesday, September 3 10:00

PAPER SESSION 3: PSYCHOACOUSTICS—PART 1

3-1 Apparent Source Width and Listener Envelopment in Relation to Source-Listener Distance—*Hyunkook Lee,* University of Huddersfield, West Yorkshire, UK

The perceptions of ASW and LEV at different source-listener distances in a reverberant sound field were investigated subjectively and objectively. Twelve subjects graded the magnitudes of perceived ASW and LEV for trumpet and conga sounds that were convolved with binaural room impulse responses measured at 3 m, 6 m, and 12 m from the source. Both ASW and LEV were found to decrease significantly and almost linearly per doubling the distance. The distance-dependent ASW was best predicted using objective measures called GE (early sound strength) while the LEV results were highly correlated with GL (late sound strength) and B/F ratio (Back/Front energy ratio of late sound). Such conventional measures as [1-IACCE], [1-IACCL], and LF did not agree with the perceived results. The findings of the current study are expected to provide psychoacoustic basis for controlling virtual front-back listening positions in a reproduced sound field.

3-2 The Prediction of the Acceptability of Auditory Interference Based on Audibility—*Khan Baykaner,*[1] *Christopher Hummersone,*[1] *Russell Mason,*[1] *Søren Bech*[2]
[1]University of Surrey, Guildford, Surrey, UK
[2]Bang & Olufsen, Struer, Denmark

In order to evaluate the ability of sound field control methods to generate independent listening zones within domestic and automotive environments, it is useful to be able to predict, without listening tests, the acceptability of auditory interference scenarios. It was considered likely that a relationship would exist between masking thresholds and acceptability thresholds, thus a listening test was carried out to gather acceptability thresholds to compare with existing masking data collected under identical listening conditions. An analysis of the data revealed that a linear regression model could be used to predict acceptability thresholds, from only masking thresholds, with RMSE = 2.6 dB and R = 0.86. The same linear regression model was used to predict acceptability thresholds but with masking threshold predictions as the input. The results had RMSE = 4.2 dB and R = 0.88.

Tuesday, September 3 11:20

PAPER SESSION 4: PSYCHOACOUSTICS—PART 2

4-1 Perceptually Optimized Loudspeaker Selection for the Creation of Personal Sound Zones—*Jon Francombe,*[1] *Philip Coleman,*[1] *Marek Olik,*[1] *Khan Baykaner,*[1] *Philip Jackson,*[1] *Russell Mason,*[1] *Martin Dewhirst,*[1] *Søren Bech,*[2] *Jan Abildgaard Pedersen*[2]
[1]University of Surrey, Guildford, Surrey, UK
[2]Bang & Olufsen, Struer, Denmark

Sound field control methods can be used to create multiple zones of audio in the same room. Separation achieved by such systems has classically been evaluated using physical metrics including acoustic contrast and target-to-interferer ratio (TIR). However, to optimize the experience for a listener it is desirable to consider perceptual factors. A search procedure was used to select five loudspeakers for production of two sound zones using acoustic contrast control. Comparisons were made between searches driven by physical (program-independent TIR) and perceptual (distraction predictions from a statistical model) cost functions. Performance was evaluated on TIR and predicted distraction in addition to subjective ratings. The perceptual cost function showed some benefits over physical optimization, although the model used needs further work.

4-2 Cognitive Maps in Spatial Sound—*Peter Lennox*, University of Derby, Derby, UK

This paper discusses the applicability of the "cognitive map" metaphor to potential usages of artificial auditory environments. The theoretical contents of such maps are suggested. Maps are generally considered as having spatial, temporal, causal, and territorial representational character, so that affordances in the environment can be utilized in timely fashion. A goal of this theorizing is that artificial auditory environments could appropriately represent affordances for interaction in entertainment, simulation, and auditory cognitive training.

4-3 Perception of Reconstructed Sound-Fields: The Dirty Little Secret—*Anthony Tucker,*[1] *William Martens,*[1] *Glenn Dickins,*[2] *Michael P. Hollier*[3]
[1]The University of Sydney, Sydney, NSW, Australia
[2]Dolby Laboratories Australia, Sydney, Australia
[3]Dolby Laboratories, San Francisco, CA, USA

This paper presents results of a pilot study regarding the perception of reconstructed sound-fields by a pairwise preference test between reproductions based upon perceptually versus mathematically posed objectives. These preliminary results suggest that increased calibration and accuracy, including loudspeaker signal time-alignment, was found to produce spatial imagery that is less preferred. Therefore a series of experiments have been launched to investigate the impact of reconstruction accuracy on user experience by selectively degrading spatial fidelity over the reproduction area. The dirty little secret, that higher accuracy does not produce preferred results, may well be revealed through such experimental studies.

Tuesday, September 3 13:40

PAPER SESSION 5: SOUND ZONES

5-1 Optimizing the Planarity of Sound Zones—
Philip Coleman,[1] *Philip Jackson,*[1] *Marek Olik,*[1] *Jan Abildgaard Pedersen*[2]
[1]University of Surrey, Guildford, Surrey, UK
[2]Bang & Olufsen, Struer, Denmark

Reproduction of personal sound zones can be attempted by sound field synthesis, energy control, or a combination of both. Energy control methods can create an unpredictable pressure distribution in the listening zone. Sound field synthesis methods may be used to overcome this problem but tend to produce a lower acoustic contrast between the zones. Here, we present a cost function to optimize the cancellation and the plane wave energy over a range of incoming azimuths, producing a planar sound field without explicitly specifying the propagation direction. Simulation results demonstrate the performance of the methods in comparison with the current state of the art. The method produces consistent high contrast and a consistently planar target sound zone across the frequency range 80–7000 Hz.

5-2 A Comparative Performance Study of Sound Zoning Methods in a Reflective Environment—*Marek Olik,*[1] *Jon Francombe,*[1] *Philip Coleman,*[1] *Philip J. B. Jackson,*[1] *Martin Olsen,*[2] *Martin Møller,*[2] *Russell Mason,*[1] *Søren Bech*[2]
[1]University of Surrey, Guildford, Surrey, UK
[2]Bang & Olufsen, Struer, Denmark

While sound zoning methods have typically been studied under anechoic conditions, it is desirable to evaluate the performance of various methods in a real room. Three control methods were implemented (delay and sum, DS; acoustic contrast control, ACC; and pressure matching, PM) on two regular 24-element loudspeaker arrays (line and circle). The acoustic contrast between two zones was evaluated and the reproduced sound fields compared for uniformity of energy distribution. ACC generated the highest contrast, while PM produced a uniform bright zone. Listening tests were also performed using monophonic auralizations from measured system responses to collect ratings of perceived distraction due to the alternate audio program. Distraction ratings were affected by control method and program material.

5-3 Sound Zones: Scattering Study with Head and Torso Simulator—*Martin Olsen, Martin Bo Møller*, Bang & Olufsen A/S, Struer, Denmark

One application of multichannel reproduction systems is to create personal sound zones. The key requirement of this concept is the ability for the system to generate one or more spatially confined regions of silence simultaneously with a single region optimized in some sense. In this paper a scattering study is presented for two different sound zoning algorithms, in order to investigate the influence of a scattering object in the optimized region. The performance degradation when introducing a simple spherical scatterer is compared to a more realistic geometry resembling the listener using Finite Element Method and evaluated against measurements made under anechoic conditions. The results show that the reduction in separation between zones due to the scatterer depends on the characteristics of the wave field.

5-4 A Comparison of Control Strategies for a Car Cabin Personal Audio System—*Jordan Cheer, Stephen J. Elliott*, University of Southampton, Southampton, UK

The generation of personal listening zones in a car cabin would allow the different occupants to listen to different audio programs without the use of headphones. The generation of independent listening zones has been achieved in a number of applications using loudspeaker arrays and superdirective, or optimal beamforming methods. However, in practice these optimized arrays are susceptible to robustness issues. In the context of a loudspeaker array designed to provide independent listening zones in the front and rear of a car cabin sized enclosure, this paper presents a comparison between the potential performance and numerical conditioning of a personal audio system optimized using either the acoustic contrast or the least squares optimization methods.

5-5 The Design of a Personal Audio Superdirective Array in a Room—*Marcos Simon Galvez, Stephen J. Elliott*, University of Southampton, Southampton, UK

When a directional sound radiator is placed inside a reverberant environment, its directional characteristics are reduced, due to the uniform distribution of reverberant pressure all around the volume. The only way of achieving good directional characteristics in a reverberant environment is to minimize the power that is radiated into the reverberant field. This paper covers the design stage of a personal audio device that is intended to work an aid for the hearing impaired, needing a large directive index. The simulation of the array's performance into a reverberant field allows its performance to be estimated at the design stage. The performance of the device is also simulated in terms of Speech Transmission Index (STI), and the array design is optimized to maximizes speech intelligibility.

Tuesday, September 3 16:00

WORKSHOP 2

Emerging Techniques, Applications, and Opportunities for Sound Field Control

Moderator: **Boaz Rafaely**, Ben-Gurion University of the Negev, Beer-Sheva, Israel

Panelists: *Alain Berry*, Université de Sherbrooke, Sherbrooke, Ontario, Canada, and CIRMMT, McGill University, Montreal, Quebec, Canada
Karlheinz Brandenburg, Fraunhofer Institute for Digital Media Technology IDMT, Ilmenau, Germany and Ilmenau University of Technology, Ilmenau, Germany
Emanuël Habets, University of Erlangen-Nuremberg, Erlangen, Germany
Gavin Kearney, University of York, York, UK

The range of potential application domains for the techniques and solutions currently being developed in relation to sound field control spans scenarios from those that are highly specialist to those that appeal to the mass market. This workshop will debate a number of interesting emergent methods and approaches, and provide perspectives on potential opportunities for applications of sound field control in the future.

Tuesday, September 3 17:00

PAPER SESSION 6: TRANSDUCERS, ARRAY DESIGN, BEAM FORMING

6-1 [Invited] Design of a Prototype Variable Directivity Loudspeaker for Improved Surround Sound Reproduction in Rooms—*Mark Poletti, Terence Betlehem*, Callaghan Innovation Research, Lower Hutt, New Zealand

Loudspeakers with variable directivity offer the potential for improved sound reproduction quality in rooms by reducing sound reflections from walls, providing better reconstruction of the desired sound field than an equivalent array of monopole loudspeakers and by allowing optimum active compensation of reverberation. This paper discusses the design of a variable directivity loudspeaker array for 2-D surround reproduction. The loudspeaker uses fifteen drivers in a circular array and Fourier beam-forming to generate up to third order responses in azimuth. Azimuthal and vertical polar responses are presented that quantify the performance of the loudspeaker.

6-2 [Invited] Arrangements of a Pair of Loudspeakers for Sound Field Control with Double-Layer Arrays—*Jiho Chang,*[1] *Finn Agerkvist,*[1] *Martin Olsen*[2]
[1]Technical University of Denmark, Lyngby, Denmark
[2]Bang & Olufsen, Struer, Denmark

Recent studies have attempted to control sound fields,and also to reduce room reflections with a circular or spherical array of loudspeakers. One of the attempts was to suppress sound waves propagating to the walls outside the array with a circular double-layer array of loudspeakers. The double-layer array represents a set of a monopole and a dipole in the Kirchhoff-Helmholtz integral equation, and thus the distance between these layers should be short compared with the wavelength. In practice, however, this condition is occasionally hard to satisfy because of the sizes of loudspeaker cabinets. In order to solve this problem, this study aims to examine several arrangements of a pair of loudspeakers that has a short distance between the acoustic centers. The effect of diffraction of sound waves due to the enclosure of another loudspeaker is investigated in simulations in terms of the position of the acoustic center. As a result, it is shown that a loudspeaker has an approximately omnidirectional radiation pattern at low frequencies in spite of the other loudspeaker cabinet, but the acoustic center is shifted to the opposite direction of the cabinets.

Wednesday, September 4 09:00

KEYNOTE SPEAKER

Creative Sound Field Control—*Frank Melchior*, BBC Research & Development

Sound field control has been developed significantly over the last decades. From Wave Field Synthesis to Higher Order Ambisonics to sound field control in live venues and sound reinforcement, several systems have been proposed and found its way into application. While these systems and reproduction methods are better understood and important links to perception are more and more established, unlocking their creative potential is still at the beginning. From the perspective of a content provider the creative use of such potential future reproduction system is promising. Furthermore an implementation in uncontrolled environments and domestic environments is necessary and an important area for further research. This talk will focus on the creative use of sound field control. The applications of sound field reproduction system will be discussed from the creative and user interface perspective. The implementation of these technologies in content production will be discussed. A focus will be how to unlock the creative potential given by novel signal processing and how it can made available to producers and creatives.

Wednesday, September 4 10:00

PAPER SESSION 7: SOUND FIELD CONTROL FOR MULTICHANNEL AUDIO

7-1 [Invited] Control of Velocity for Sound Field Reproduction—*Mincheol Shin,*[1] *Filippo Fazi,*[1] *Philip Nelson,*[1] *Jeongil Seo*[2]
[1]University of Southampton, Southampton, UK
[2]Electronics and Telecommunications Research Institute (ETRI), Daejeon, Korea

The conventional approach of using the method of matching the acoustic pressure on a bounding surface for the reproduction of a sound field in the enclosed volume has been shown to suffer from technical limitations when the loudspeakers used for reproduction are irregularly distributed. The difficulties in sound field reproduction with an irregular (spatially non-uniform) loudspeaker arrangement have been resolved by changing the component controlled from acoustic pressure to acoustic particle velocity. The velocity controlled sound field shows better reproduction of the time averaged intensity flow in the reproduced field, and this can be shown to be related to the main cues for sound localization regardless of loudspeaker arrangement. This is accomplished by matching radial particle velocity components in the target and reproduced fields on the surface of the control volume whose size is made frequency dependent. The performance of velocity matching method is discussed in this paper and compared with the conventional pressure matching method through computer simulations based on a loudspeaker layout. Furthermore, the simulated results were verified experimentally.

7-2 [Invited] Spatial PCM Sampling: A New Method for Sound Recording and Playback—*Angelo Farina, Alberto Amendola, Lorenzo Chiesi, Andrea Capra, Simone Campanini*, University of Parma, Parma, Italy

This paper presents the mathematical and physical framework of a new technology named SPS (Spatial PCM Sampling). It is the equivalent, in a two-dimensional spherical-coordinate space, of the traditional PCM representation of a waveform (in the one-dimensional time domain). It is nowadays possible to record an SPS multichannel stream (also called P-format) by processing the signals coming from massive microphone arrays, now widely employed in the broadcasting industry and in research labs. Some types of sound processing are easy when operating on P-format signals; some, indeed, require more work. At playback, it is possible to drive loudspeaker arrays of arbitrary shape and complexity, providing in general better spatial accuracy than competing well known methods, such as Ambisonics or WFS.

Wednesday, September 4 11:20

PAPER SESSION 8: ROOM ACOUSTICS CONTROL

8-1 Sound Field Control for a Low-Frequency Test Facility—*Christian Sejer Pedersen, Henrik Møller*, Aalborg University, Aalborg, Denmark

The two largest problems in controlling the reproduction of low-frequency sound for psychoacoustic experiments is the effect of the room due to standing waves and the relatively large sound pressure levels needed. Anechoic rooms are limited downward in frequency and distortion may be a problem even at moderate levels, while pressure-field playback can give higher sound pressures but is limited upwards in frequency. A new solution that addresses both problems has been implemented in the laboratory of Acoustics, Aalborg University. The solution uses one wall with 20 loudspeakers to generate a plane wave that is actively absorbed when it reaches the 20 loudspeakers on the opposing wall. This gives a homogeneous sound field in the majority of the room with a flat frequency response in the frequency range 2-300 Hz. The lowest frequencies are limited to sound pressure levels in the order of 95 dB. If larger levels are needed, a hybrid mode can be used to utilize the pressure-field conditions at frequencies up to approximately 30 Hz while the higher frequencies are controlled by plane- wave generation. This approach allows for playback of levels at the lowest frequencies in the order of 125 dB while still maintaining a homogeneous sound field for the entire frequency range 2-300 Hz.

8-2 [Invited] Active Acoustic Absorbers Revisited—
John Vanderkooy,[1] *Martial Rousseau*[2]
[1]University of Waterloo, Waterloo, Ontario, Canada
[2]B &W Group Ltd., Steyning, West Sussex, UK

This paper continues an earlier exploration of using low-frequency loudspeakers as acoustic absorbers, with absorption cross-sections that far exceed their geometric size. Theory for a point active absorber immersed in the acoustic field of a point source is reviewed as it would apply to normal loudspeakers used as either sources or absorbers at low frequencies for which they act as compact sources. This theory contains an extra problematic term, which is suppressed if averaged over frequency or distance. In rooms, such suppression is expected due to the varying distances from all the source images to the absorber. Impulse responses in several small rooms were measured from a few sources and absorber loudspeakers to both a few listening microphones and near-field microphones mounted at the absorbers. These data were used to implement active absorption. The efficacy of the active absorber is assessed and the results are somewhat enigmatic. A room simulation was done to check on the actual measurements, and results are similar to theoretical expectation. The discrepancy should perhaps be attributed to the rather complex implementation issues.

8-3 [Invited] Sound-Field Control in Enclosures by Spherical Arrays—*Hai Morgenstern, Noam Shabtai, Boaz Rafaely*, Ben-Gurion University of the Negev, Beer-Sheva, Israel

Spherical microphone and loudspeaker arrays have been widely studied in the past decade, with applications ranging from room acoustics to active control and speech communication. A significant advance has also been presented with spherical loudspeaker arrays, with respect to array design and array processing. This paper presents new developments in spherical loudspeaker arrays for sound-field control in rooms, showing how spherical multiple-input multiple-output systems can be employed to facilitate the control of sound-field radiation in enclosures.

Wednesday, September 4 13:40

PAPER SESSION 9: WAVE FIELD SYNTHESIS

9-1 [Invited] Intelligent Multichannel Signal Processing for Future Audio Reproduction Systems—*Karlheinz Brandenburg*,[1,2] *Martin Schneider*,[3] *Andreas Franck*,[1] *Walter Kellermann*,[3] *Sandra Brix*[1]
[1]Fraunhofer Institute for Digital Media Technology IDMT, Ilmenau, Germany
[2]Ilmenau University of Technology, Ilmenau, Germany
[3]University Erlangen-Nuremberg, Erlangen, Germany

It is possible to improve audio reproduction, active noise reduction systems, and sound field control by using multi-loudspeaker systems. Distributed arrays of transducers with intelligent audio signal processing need sophisticated signal processing algorithms. The paper looks both at some of the psychoacoustic fundamentals of auditory illusion (still partly unsolved) and proposals to build solutions based on the combination of wave field synthesis (WFS) and wave domain adaptive filtering (WDAF). We include the sketch of application scenarios for future systems with a large number of loudspeakers.

9-2 [Invited] Quasi Wave Field Synthesis: Efficient Driving Functions for Improved 2.5D Sound Field Reproduction —*Dylan Menzies*, De Montfort University, Leicester, UK

Optimized driving functions for a rectangular array of loudspeakers are approximated with driving functions with similar form and cost to Wave Field Synthesis (WFS) functions, by using a linear combination of several pre-filters and a delay for each function. The accuracy of the resulting reproductions are compared with WFS reproductions. The aim is provide improved efficient source driving functions in horizontal reproduction.

9-3 A Vector Quantization-Based Compression Scheme for Wave Field Synthesis Source Signals—*Georgios N. Lilis*, Technical University of Crete, Kounoupidiana, Chania, Greece

This paper introduces a lossy data compression scheme of the signals used for the synthesis of narrow-band virtual point wave fields. The fields are synthesized by a set of distributed point wave sources at fixed locations inside a wave medium. The data compression scheme is implemented using vector quantization. An application example referring the synthesis of a virtual point field, using three point sources, is considered. Its performance is assessed by defining and calculating a mean distortion measure for different number of quantization levels. Extension to the synthesis of broad-band wave fields of arbitrary shape is straightforward.

9-4 Multichannel-to-Wave Field Synthesis Upmixing Technique Based on Sound Source Separation—
Keunwoo Choi, Tae Jin Park, Jeongil Seo, Kyeongok Kang, Electronics and Telecommunications Research Institute (ETRI), Daejeon, Korea

In this research, we propose a multichannel-to-loudspeaker array upmixing algorithm. To take advantage of a loudspeaker array, we introduce an approach based on audio source separation. During the analysis phase, multichannel signals are separated into element signals in stereo format through stereo-channel extraction. They are then separated into source signals by the Laplacian Mixture Models of the features from the stereo signals. During the synthesis phase, the sources are rendered as virtual sources through Wave Field Synthesis, or as focused sources. Subjective tests show that the proposed algorithm outperforms the comparison algorithm in terms of localization quality.

Sponsor Seminars

Tuesday, September 3 18:00

An Introduction to Yamaha AFC3—3rd Generation Acoustic Field Control
Ron Bakker, Yamaha Commercial Audio Systems Europe, London, UK
Takayuki Watanabe, Yamaha Corporation, Spatial Audio Group, Hamamatsu, Japan,
Tim Harrison, Yamaha Commercial Audio Support Centre, London, UK

Until very recently, acoustic enhancement systems required large numbers of independent channels and powerful DSP processing, making them expensive, hard to integrate and inflexible. With the introduction of the latest powerful DSP platforms—and more specifically FIR processing—the number of transducers and hardware infrastructure required can be drastically reduced (along with the associated costs), whilst increasing the reliability and functionality of the installation. This development makes new DSP based acoustic enhancement systems suitable for a wider variety of venues. Not only concert halls, but also small theatres, practice rooms, churches, multi-purpose halls and auditoria can now benefit from the technology to support performances that require a naturally enhanced acoustic environment.

An overview of the types of systems and their applications that have been installed in the past decades is presented, focusing in on the basic theoretical concepts, advantages and disadvantages. The presentation will include a demonstration using a temporarily installed AFC3 system.