IWAENC 2012

International Workshop on Acoustic Signal Enhancement

Conference Guide

RWTH Aachen University, Germany

September 4th – 6th, 2012

Sponsored by:

intel

HEAD acoustics

AB

ERICSSON

Google

Microsoft Research

NUANCE

QUALCOMM

SIEMENS

skype

Texas Instruments

VDE ITG INFORMATION TECHNOLOGY SOCIETY WITHIN VDE

RWTH AACHEN UNIVERSITY

Table of Contents

Herzlich Willkommen!

It is our great pleasure to welcome you to the 13th International Workshop on Acoustic Signal Enhancement (IWAENC) and to the city of Aachen.

As a friend of IWAENC you may have recognized that the interpretation of the acronym IWAENC has undergone some changes which reflect the evolution of our technical field from pure echo control in the early days via combined echo and noise control to a wider spectrum of topics, including beamforming, blind acoustic source separation, speaker localization, speech enhancement, active noise control, dereverberation, echo cancellation, and finally various audio signal processing applications in hearing aids, cell phones, and audio conferencing systems.

This year the workshop comprises 82 papers which are presented in eight poster sessions and a special session with six oral presentations. Each poster session starts with a keynote talk by a leading expert. In contrast to previous workshops the topics of the poster presentations are distributed over all sessions which gives you the opportunity to meet and discuss the results of your work with colleagues working in the same research area. Two separate demo sessions scheduled in parallel to the poster sessions will offer attendees various types of technical experience. A student prize is awarded to the best paper whose first author and presenter is a registered student. In addition to the technical program, there will be a large variety of social activities – another opportunity to exchange ideas and establish contacts.

In Aachen, the city of the Emperor Charlemagne, you will experience its international flair, 2000 years of history, a rich cultural heritage, a wonderful surrounding landscape, and people who know how to enjoy life. Germany's most western city, with a population of around 245,000 drawn from 156 nations, nestles up against the border with Belgium and the Netherlands. Its universities and colleges, research facilities, and high-tech businesses influence, in no small part, the present and the future of our city.

The conference venue for the IWAENC 2012 is the SuperC Student Service Center of the RWTH Aachen University, one of Germany's Universities of Excellence. This building is a visual highlight in the heart of the city as well as the university complex. It serves as a lively contact point for all students with administrative and study-related queries and includes several conference rooms that offer spectacular views over the city center.

During the preparation of this workshop we have been surrounded by the wonderful and supportive team of the Institute of Communication Systems and Data Processing at RWTH Aachen University. Our special thanks go to the team manager Christiane Antweiler for taking care of almost everything, to the program and publications coordinator Benedikt Eschbach, and to Christoph Nelke who managed the whole review process.

As the excellence of the workshop strongly depends on the quality of the submitted papers, we also thank the authors and reviewers for significantly contributing to the success of IWAENC 2012. We also express our gratitude to the organizing and technical committees for their valuable assistance. Finally, we are very grateful to the sponsoring companies Advanced Bionics, Ericsson, Google, HEAD acoustics, Intel, Microsoft Research, Nuance, Qualcomm, Siemens Audiology Solutions, Skype, and Texas Instruments for improving the working and communication conditions at the workshop.

We look forward to meeting you in Aachen.

Peter Vary, RWTH Aachen University
Walter Kellermann, University of Erlangen-Nuremberg
Hans-Wilhelm Gierlich, HEAD acoustics

Committees

Organizing Committee

Peter Vary (Chair)
RWTH Aachen University

Walter Kellermann (Co-Chair)
University of Erlangen-Nuremberg

Hans-Wilhelm Gierlich (Co-Chair)
HEAD acoustics, Herzogenrath

Christiane Antweiler
RWTH Aachen University

Benedikt Eschbach
RWTH Aachen University

Christoph Nelke
RWTH Aachen University

Technical Committee

Jacob Benesty
Université du Québec
Montreal, Canada

Alberto Carini
Università di Urbino
Urbino, Italy

Tomas Gaensler
mh acoustics
Summit, United States

Sharon Gannot
Bar-Ilan University
Ramat Gan, Israel

Steven Grant
Missouri University of Science and Technology
Rolla, United States

Yves Grenier
Télécom ParisTech
Paris, France

Yoichi Haneda
NTT Cyber Space Laboratories
Tokyo, Japan

Kees Janse
Philips Research
Eindhoven, Netherlands

Walter Kellermann
University of Erlangen-Nuremberg
Erlangen, Germany

Shoji Makino
University of Tsukuba
Tsukuba, Japan

Rainer Martin
Ruhr-Universität Bochum
Bochum, Germany

Marc Moonen
Katholieke Universiteit Leuven
Leuven, Belgium

Patrick Naylor
Imperial College London
London, United Kingdom

Sven Nordholm
Curtin University of Technology
Bentley, Australia

Henning Puder
Siemens Audiologische Technik
Erlangen, Germany

Piet Sommen
Technische Universiteit Eindhoven
Eindhoven, Netherlands

Ivan Tashev
Microsoft Research
Redmond,United States

Peter Vary
RWTH Aachen University
Aachen,Germany

Reviewer

Christophe Beaugeant
Jacob Benesty
Joerg Bitzer
Brian Bloemendal
Herbert Buchner
Markus Buck
Alberto Carini
Israel Cohen
Gerhard Doblinger
Simon Doclo
Ramani Duraiswami
Gerald Enzner
Thomas Esch
Nicholas Evans
Tiago Falk
Christof Faller
Janina Fels
Tim Fingscheidt
Jürgen Freudenberger
Tomas Gaensler
Sharon Gannot
Nikolay Gaubitch
Bernd Geiser
Timo Gerkmann
Hans Gierlich
Simon Godsill
Stefan Goetze
Steven Grant
Yves Grenier
Emanuel Habets
Reinhold Haeb-Umbach
Yoichi Haneda
John Hansen
Richard Hendriks
Richard Heusdens
Ulrich Heute
Rüdiger Hoffmann
Volker Hohmann
Herbert Hudde
Kees Janse
Jesper Jensen
Marco Jeub
Hirokazu Kameoka
Walter Kellermann
Frank Kettler
W. Bastiaan Kleijn
Dorothea Kolossa
Gernot Kubin
Fabian Kuech
Chiong Lai
Christina Leitner
Ludovick Lepauloux
Heinrich Löllmann
Tapio Lokki
Nilesh Madhu
Shoji Makino
Rainer Martin
Bruno Masiero
Alfred Mertins
Marc Moonen
Moctar Mossi Idrissa
Patrick Naylor
Christoph Nelke
Elias Nemer
Francesco Nesta
Robert Nickel
Christoph Norrenbrock
Nobutaka Ono
Matthias Pawig
Alexander Petrovsky
Henning Puder
Augusto Sarti
Magnus Schäfer
Gerhard Schmidt
Jan Skoglund
Dirk Slock
Piet Sommen
Sascha Spors
Sriram Srinivasan
Akihiko Sugiyama
Nordholm Sven
Ivan Tashev
Oliver Thiergart
Mark Thomas
Steven van de Par
Werner Verhelst
Jason Wung
Christelle Yemdji

Keynote Talks

Media Signal Processing in Cell Phones – What is so Smart about it?

Tuesday, September 4th, 9:15, Generali-Saal

Peter Kroon
Intel, Mobile and Communciations Group, Allentown, United States

Modern cell phones have become widely accepted across the world, and have become small wonders of media signal processing. Although voice will always remain the essential media signal for a phone, many other multimedia applications have found its way into cell phones, making it a true media signal processing device. Voice, audio, image, video and graphics are all present, and are processed using techniques based on years of media signal processing research. To make this all work in a device that is constrained by power, size and cost has turned out to be quite a challenge. This talk will review some of the relevant media standards and processing techniques that are commonly found in cell phones. We also highlight some interesting accomplishments, and describe some of the challenges that we will find ahead.

Signal Enhancement for Future High-Resolution Spatial Audio Formats

Tuesday, September 4th, 10:45, Generali-Saal

Peter Jax
Technicolor Research & Innovation, Hanover, Germany

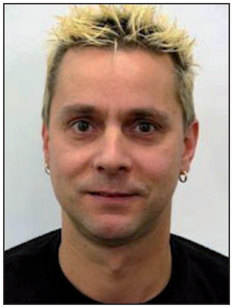
After the well-known standards of stereo and surround sound have served as a robust baseline for sound creation during the last decades, there is currently a rising trend in the media and entertainment industry towards more sophisticated spatial audio formats. Proposed technical solutions include evolutionary concepts that extend the conventional stereo and surround sound specifications by additional loudspeakers at specific positions, as well as more universal techniques that represent spatial audio content in ways that are intrinsically independent of the target loudspeaker configuration. The latter techniques comprise object-oriented and sound-field-oriented approaches, as well as hybrid technologies.

With the increasing complexity of loudspeaker settings and spatial audio formats, there is also a rising need for more advanced spatial audio signal capturing and post production techniques. Sound mixers need at least the same order of quality and similar ways to manipulate recorded sound as they are used to have within todays production workflows for stereo and surround sound. It is one of the goals at Technicolor to provide sound mixing artists with advanced signal enhancement and processing tools that enable them to create considerably more immersive and compelling spatial audio content than it was possible with stereo and surround formats.

In this talk, we will highlight constraints, technical concepts and open research challenges from this application area. One of the most important constraints is the necessity of having an interacting creative person looped in. This provides interesting opportunities for designing very powerful algorithms without the need for fully automated processing, while creating at the same time the need for careful conditioning of the spatial audio signals for the manual interaction.

Interactive Audio in a Web-Based World

Tuesday, September 4th, 14:00, Generali-Saal



Jan Skoglund
Chrome at Google, Mountain View, United States

This talk will discuss the challenges of building audio processing applications at scale and the need for robustness of these tools when used by millions of users having diverse usage environments ranging from mobile devices to multi-channel home theatre setups. The approach we have chosen to address these challenges is to drive both real-time and non-real-time applications from a web browser. By using an open source model collaboration between industry and academia is easily enabled and can also accelerate development faster than a closed system. Examples of such collaboration will be given.

Paths toward HD-Voice Communication

Tuesday, September 4th, 16:30, Generali-Saal

Bernd Geiser
RWTH Aachen University, Aachen, Germany

These days, the telecommunication world is undergoing a major technology change toward a universal, packet-based network architecture for both fixed and mobile communications. The main motivations and incentives behind this effort are presumably improved flexibility and cost-efficiency. But in particular for speech and audio communication applications, the opportunity should be seized to promote high quality services which are far superior to the long-accustomed narrowband speech telephony experience. Indeed, new audio codecs, delivering additional functionality and a much better audio quality, are deployed much quicker within such a (future) network environment.

But, as a matter of fact, very little is done to improve the audio quality for today's communication networks. Instead, "least common denominator solutions" are pursued, keeping up the status quo of narrowband speech. At first sight, this might appear reasonable from the economic and marketing perspectives. However, it is nevertheless true that subscribers of new services will still experience inferior quality if their communication partner uses an old telephone or circuit-switched network access, e.g., via GSM/UMTS speech channels or private/government subnetworks. Large parts of the worldwide telephone network are in fact based on such legacy technology and can be expected to prevail for a long time. To this end, new, more advanced methods and algorithms for "High Definition" audio transmission and reproduction are required that maintain interoperability with legacy network components.

In this contribution, current developments in packet-based HD-voice communication are summarized and a future perspective toward systems for binaural/ambient audio communication is given. Moreover, the, usually problematic, interoperability issue is addressed. Therefore, several algorithmic approaches—including embedded coding, receiver- or network-based parameter estimation, and steganographic parameter transmission—are discussed based on the practically relevant example of parametric bandwidth extension for speech and audio signals.

Acoustic Signal Processing in Noise: It‘s Not Getting Any Quieter

Wednesday, September 5th, 9:00, Generali-Saal

Patrick Naylor
Imperial College London, London, United Kingdom

Processing signals degraded by noise brings specific challenges, particularly for speech signals which may also suffer reverberation associated with room acoustics and nonlinear distortions such as caused in voice networks. Our societes' economic and social ambitions, together with a quickly growing global population and increasing urbanisation, point only to the increasing importance of human communication technology that is robust to noise, and which might eventually even help to reduce it. Even today one might sometimes ask: what would we give for a little peace and quiet?

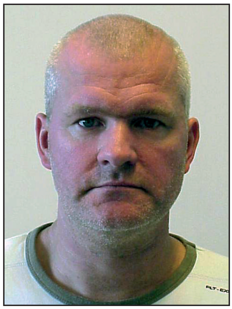
Many researchers are continuously amazed at just how well humans can communicate even in scenarios with severe degradations in the speech signal. It seems intuitive, however, that this human capability does not come without substantial cognitive load. Recent experiments will be used to illustrate how one might quantify the increase in cognitive load associated with human speech understanding as a function of the type and level of degradation applied to speech. This kind of information has the potential then to inform the design of speech enhancement technology so as to maximize listening comfort and intelligibility.

As well as additive noise, speech may be degraded by convolution with an unknown acoustic transmission channel so as to cause reverberation. Estimates of the channel characteristics can be useful to enhance the received speech signal if such characteristics can be estimated blindly and with the necessary accuracy. Recent research on multi-channel blind SIMO acoustic system identification has spawned new research on the resulting acoustic inverse filtering problem which will be described as a way to reduce reverberation, though usually at the cost of introducing some level of undesired artefacts. In the case of single-channel acquisition, alternative approaches to channel estimation will be presented with examples.

The talk will also emphasize the importance of comparable evaluations of acoustic signal processing algorithms, using the same data and the same metrics, such as advocated in the recently launched IEEE AASP Challenges.

Distributed Signal Processing: Application to MVDR Beam-Forming

Thursday, September 6th, 9:00, Generali-Saal



Richard Heusdens
Delft University of Technology, Delft, Netherlands

With the emergence of (large scale) wireless audio sensor networks, there is a need for a new class of algorithms that can implement audio processing algorithms in a distributed fashion. Wireless audio sensor networks consist of a large number of nodes, each having a sensing (microphone), data processing, and communication component. In such networks, due to the absence of a central processing point (fusion center), nodes use their own processing ability to locally carry out simple computations and transmit only the required and partially processed data to neighboring nodes. Despite these simple operations of the individual nodes, jointly they are able to perform relatively complex tasks. The decentralized settings in which signal processing algorithms then have to be deployed are typically dynamic, in the sense that sensors are added, removed, or moving, usually in an unpredictable way. In those settings, the algorithms must allow for a parallel implementation, must be easily scalable, must be able to exploit the possible (large) sparse geometry in the problem and must be numerically robust against (small) changes in the network topology.

We will present an iterative, distributed MVDR beamforming algorithm using message passing. The algorithm is based on probabilistic inference in random Markov fields. At each iteration, each node in the network keeps track of so-called messages received from neighboring nodes, which are used to make a new estimate of the final solution and construct new messages to be transmitted at the next iteration. In the talk we will give an introduction to message-passing algorithms and show their suitability for a variety of signal processing applications in wireless sensor networks, where we use the MVDR beamformer application to illustrate the algorithm.

Optimized Directional Processing in Hearing Aids with Integrated Spatial Noise Reduction

Thursday, September 6th, 11:00, Generali-Saal

Henning Puder, Eghart Fischer, Jens Hain
Siemens Audiologische Technik, Erlangen, Germany

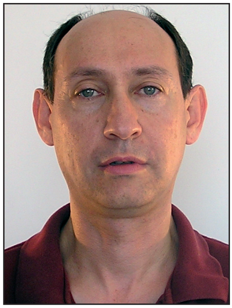
In this contribution a differential beamformer for hearing aids is presented combined with a direction dependent noise reduction. The target is to achieve good speech intelligibility and quality in adverse noisy environments while coping with the hearing aid constraints such as small microphone distances and head shading.

First, a differential beamformer in sub-bands is presented which allows a good interference cancellation for small beamformer apertures. The sub-band structure allows to optimize the direction dependent attenuation by an adaptation to the noise interference and the head shading. When applying those differential structures, high microphone noise amplification occurs which is generally limited by severely constraining the beamformer performance for low frequencies. In this contribution we present a fast adaptation control which allows to simultaneously minimize the ambient interference and the microphone noise: At each time instance, and independently for each sub-band, the adaptation selects the maximum possible interference cancellation where the residual interference after beamforming just masks the microphone noise.

The direction dependent noise reduction complements the well-known stationary and transient noise reduction procedures which are typically applied after the beamformer. It allows to suppress noise components independent of their stationarity properties. A noise reference is calculated within the beamformer by attenuating signals from the front direction which is assumed to be the target signal direction. The direction dependent noise reduction allows to suppress all kinds of interference components arriving from outside the look direction. Finally, we show approaches which allow an optimized combination of the direction dependent noise reduction and the well-known noise reduction procedure in order to minimize artefacts and optimize the sound quality.

Multi-Microphone Speech Enhancement

Thursday, September 6th, 14:00, Generali-Saal



Sharon Gannot
Bar-Ilan University, Ramat Gan, Israel

Microphone array algorithms emerged in the early 1990s as viable solutions to speech processing problems. However, the adaptation of beamforming methods to speech processing is still an open issue. There are many difficulties arising from the characteristics of the speech signal and the acoustic environment. The speech signal is a wide-band and nonstationary signal. Very long, time-varying, room impulse responses may be attributed to multiple reflections of the sound field and to moving objects in the acoustic enclosure.

In this talk, we will focus on spatial processors ("beamformers") based on the linearly constrained minimum variance (LCMV) criterion, and its special case, the minimum variance distortionless (MVDR) beamformer. The implementation of the LCMV beamformer in the short-time Fourier transform (STFT) domain and its structuring as a generalized sidelobe canceller (GSC) facilitate the application of the presented algorithms to speech signals in real acoustic environments. We will show how the powerful LCMV criterion can be applied to various related problems. For example, speech enhancement, extraction of desired speakers in multiple competing speaker environment, and combined noise reduction and echo cancellation. Special attention will be given to blind estimation techniques of the GSC components and to the efficient design of its various blocks. We will also elaborate on the relative transfer function (RTF) and its importance in speech processing. We will conclude our talk with a discussion of the applicability of the LCMV to binaural processing. If time permits, novel distributed microphone array architectures will be reviewed, and the new advantages and challenges they raise will be explored. The presentation will be accompanied by processed audio files demonstrating the algorithms' performance.

Technical Program

Tuesday, September 4th

09:00 – 09:15 Welcome **(Chair: Peter Vary)**

09:15 – 10:15 Keynote Talk

Media Signal Processing in Cell Phones – What is so Smart about it?

Peter Kroon
Intel, Mobile and Communciations Group, Allentown, United States

10:15 – 10:45 Coffee Break

10:45 – 11:15 Keynote Talk (Chair: Hans Gierlich)

Signal Enhancement for Future High-Resolution Spatial Audio Formats

Peter Jax
Technicolor Research & Innovation, Hanover, Germany

11:15 – 12:30 Poster Session A

A-01 **Modeling Late Reverberation in Acoustic Echo Suppression**

Alexis Favrot[1], Christof Faller[1], Fabian Kuech[2]
[1]Illusonic, Uster, Switzerland,
[2]Fraunhofer IIS, Erlangen, Germany

A-02 **Multichannel Adaptive Filtering with Sparseness Constraints**

Karim Helwani[1], Herbert Buchner[2], Sascha Spors[1]
[1]Deutsche Telekom Laboratories, Berlin, Germany,
[2]Technical University of Berlin, Berlin, Germany

A-03 **Reflection Coefficient Estimation by Pseudospectrum Matching**

Dejan Marković[1], Christian Hofmann[2], Fabio Antonacci[1], Konrad Kowalczyk[2], Augusto Sarti[1], Walter Kellermann[2]
[1]Politecnico di Milano, Milano, Italy,
[2]University of Erlangen-Nuremberg, Erlangen, Germany

A-04 **Transient Interference Suppression in Speech Signals Based on the OM-LSA Algorithm**

Ariel Hirszhorn, David Dov, Ronen Talmon, Israel Cohen
Israel Institute of Technology, Haifa, Israel

A-05 **MMSE Log-Spectral Amplitude Estimation Under Speech Presence Uncertainty Using Generalized Gamma Speech Priors**

Balázs Fodor, Tim Fingscheidt
Technische Universität Braunschweig, Braunschweig, Germany

A-06 **A Morphological Approach to Single-Channel Wind-Noise Suppression**

Christian Hofmann[1], Tobias Wolff[2], Markus Buck[2], Tim Haulick[2], Walter Kellermann[1]
[1]University of Erlangen-Nuremberg, Erlangen, Germany,
[2]Nuance Communications Aachen, Ulm, Germany

A-07 **Numerical Near Field Optimization of Weighted Delay-and-Sum Microphone Arrays**

Magnus Schäfer, Florian Heese, Jona Wernerus, Peter Vary
RWTH Aachen University, Aachen, Germany

A-08 **Maximum a Posteriori Trajectory Estimation for Acoustic Source Tracking**

Alessio Brutti, Maurizio Omologo, Piergiorgio Svaizer
Fondazione Bruno Kessler, Trento, Italy

A-09 **Sound Field Model Violations in Parametric Spatial Sound Processing**

Oliver Thiergart, Emanuël Habets
International Audio Laboratories Erlangen, Erlangen, Germany

A-10 **On Blocking Matrix-Based Dereverberation for Automatic Speech Recognition**

Andreas Schwarz, Klaus Reindl, Walter Kellermann
University of Erlangen-Nuremberg, Erlangen, Germany

11:15 – 12:30 Demonstrator Session (Room 5.31/5.32)

DA-01 **Real-Time Semi-Blind Speech Extraction with Motion Tracking on Kinect**

Yuji Onuma, Noriyoshi Kamado, Ryoichi Miyazaki, Hiroshi Saruwatari, Kiyohiro Shikano
Shikano Lab, Ikoma, Japan

DA-02 **Spatial Audio Conferencing Using Binaural HD Voice**

Matthias Rüngeler, Hauke Krüger, Thomas Schlien, Peter Vary
RWTH Aachen University, Aachen, Germany

12:30 – 14:00	Lunch

14:00 – 14:30	**Keynote Talk** (Chair: Israel Cohen)

Interactive Audio in a Web-Based World
Jan Skoglund
Chrome at Google, Mountain View, United States

14:30 – 16:00	**Poster Session B**

B-01 **Robust and Low Complexity Delay Estimation**
Björn Völcker[1], W. Bastiaan Kleijn[2]
[1]Google, Mountain View, United States,
[2]Victoria University of Wellington, Wellington, New Zealand

B-02 **Nonlinear Acoustic Echo Cancellation Based on Piecewise Linear Approximation with Amplitude Threshold Decomposition**
Suehiro Shimauchi, Yoichi Haneda
NTT Cyber Space Laboratories, Tokyo, Japan

B-03 **Fuzzy Sound Field Classification in Devices with Multiple Acoustic Sensors**
Roman Scharrer, Janina Fels
RWTH Aachen University, Aachen, Germany

B-04 **Assessment of Multichannel Acoustic System Identification Using a Spectral-Importance Weighted Misalignment**
Philipp Thüne, Gerald Enzner
Ruhr-University Bochum, Bochum, Germany

B-05 **Extension of Pre-Image Speech De-Noising by Voice Activity Detection Using a Bone Conductive Microphone**

Christina Leitner, Franz Pernkopf
Graz University of Technology, Graz, Austria

B-06 **A New Approach for Reduction of Supergaussian Noise Using Autoregressive Interpolation and Time-Frequency Masking**

Marco Ruhland[1], Stefan Goetze[1], Matthias Brandt[2], Joerg Bitzer[1,2], Simon Doclo[1,3]
[1]Fraunhofer Institute for Digital Media Technology, Oldenburg, Germany,
[2]Jade University of Applied Sciences Oldenburg, Germany,
[3]University of Oldenburg, Oldenburg, Germany

B-07 **Noise PSD Estimation Using Blind Source Separation in a Diffuse Noise Field**

Lin Wang, Timo Gerkmann, Simon Doclo
University of Oldenburg, Oldenburg, Germany

B-08 **Noise Estimation Based on Soft Decisions and Conditional Smoothing for Speech Enhancement**

Pei Chee Yong, Sven Nordholm, Hai Huyen Dam
Curtin University, Bentley, Australia

B-09 **Optimal 3D Beamforming Using Measured Microphone Directivity Patterns**

Mark Thomas, Jens Ahrens, Ivan Tashev
Microsoft Research, Redmond, United States

B-10 **Source Localization Based on the Doppler Effect**

Alexander Schasse, Christian Tendyck, Rainer Martin
Ruhr-University Bochum, Bochum, Germany

B-11 **Single-Microphone Speech Enhancement by Skewness Maximization and Spectral Subtraction**

Saeed Mosayyebpour, T. Aaron Gulliver, Morteza Esmaeili
University of Victoria, Victoria, Canada

B-12 **Binaural Linearly Constrained Minimum Variance Beamformer for Hearing Aid Applications**

Elior Hadad[1], Sharon Gannot[1], Simon Doclo[2]
[1]Bar-Ilan University, Ramat Gan, Israel,
[2]University of Oldenburg, Oldenburg, Germany

B-13 **Noise Power Spectral Density Estimation for Public Address Systems in Noisy Reverberant Environments**

Neda Faraji[1], Richard Hendriks[2]
[1]Amirkabir University of Technology, Tehran, Iran,
[2]Delft University of Technology, Delft, Netherlands

16:00 – 16:30	Coffee Break

16:30 – 17:00	**Keynote Talk**	**(Chair: Osamu Hoshuyama)**

Paths toward HD-Voice Communication

Bernd Geiser
RWTH Aachen University

17:00 – 18:00 Poster Session C

C-01 **Dual Channel Echo Postfiltering for Hands-Free Mobile Terminals**

Christelle Yemdji[1], Moctar Mossi Idrissa[1], Nicholas Evans[1], Christophe Beaugeant[2], Peter Vary[3]

[1]EURECOM, Sophia-Antipolis, France,
[2]Intel, Mobile Communications Group, Sophia-Antipolis, France,
[3]RWTH Aachen University, Aachen, Germany

C-02 **Inter-Channel Decorrelation by Resampling via Time-Domain Interpolation Filters Derived from the Time-Shifting Property**

Jason Wung, Ted Wada, Fred Juang

Georgia Institute of Technology, Atlanta, United States

C-03 **Reverberation-Robust Online Multi-Speaker Tracking by Using a Microphone Array and CASA Processing**

Axel Plinge, Marius Hennecke, Gernot Fink

TU Dortmund University, Dortmund, Germany

C-04 **Speech Enhancement Using Emotion-Dependent Codebooks**

Hanumantha Rao Naidu[1], Sriram Srinivasan[2]

[1]Sri Sathya Sai Institute of Higher Learning, Prasanthi Nilayam, India,
[2]Philips Research, Eindhoven, Netherlands

C-05 **Improved Prediction of Nearly-Periodic Signals**

W. Bastiaan Kleijn[1], Jan Skoglund[2]

[1]Victoria University of Wellington, Wellington, New Zealand,
[2]Google, Mountain View, United States

C-06 **Distributed Delay and Sum Beamformer in Regular Networks Based on Synchronous Randomized Gossip**

Yuan Zeng, Richard Hendriks
Delft University of Technology, Delft, Netherlands

C-07 **Using Statistical Room Acoustics for Computing the Spatially Averaged Performance of the Multichannel Wiener Filter Based Noise Reduction**

Toby Christian Lawin-Ore, Simon Doclo
University of Oldenburg, Oldenburg, Germany

C-08 **Exact Localization of Planar Acoustic Reflectors in Three-Dimensional Geometries**

Antonio Canclini[1], Fabio Antonacci[1], Jason Filos[2], Augusto Sarti[1], Patrick Naylor[2]
[1]Politecnico di Milano, Milano, Italy,
[2]Imperial College London, London, United Kingdom

C-09 **Low Variance Blind Estimation of the Reverberation Time**

Nicolás López[1,2], Yves Grenier[2], Gaël Richard[2], Ivan Bourmeyster[1]
[1]Arkamys, Paris, France,
[2]Télécom ParisTech, Paris, France

C-10 **Beamformer for Driving Binaural Speech Enhancement**

Heinrich Löllmann, Peter Vary
RWTH Aachen University, Aachen, Germany

Wednesday, September 5th

09:00 – 09:30 Keynote Talk (Chair: Andy Khong)

Acoustic Signal Processing in Noise: It‘s Not Getting Any Quieter

Patrick Naylor
Imperial College London, London, United Kingdom

09:30 – 10:30 Poster Session D

D-01 **Modeling External Volume Changes in Stereo Echo Cancellers**

Elias Nemer, Wilf Leblanc
Broadcom Corporation, Irvine, United States

D-02 **Echo Reduction Using Wiener Gains Considering Short-Time Correlation Between Echo and Near-End Speech**

Masahiro Fukui[1], Akira Nakagawa[1], Suehiro Shimauchi[1], Yoichi Haneda[1], Akitoshi Kataoka[2]
[1]NTT Cyber Space Laboratories, Tokyo, Japan,
[2]Ryukoku University, Shiga, Japan

D-03 **Plenacoustic Imaging in the Ray Space**

Dejan Marković, Giorgio Sandrini, Fabio Antonacci, Augusto Sarti, Stefano Tubaro
Politecnico di Milano, Milano, Italy

D-04

Theoretical Analysis of Musical Noise Generation in Noise Reduction Methods with Decision-Directed a Priori SNR Estimator

Suzumi Kanehara[1], Hiroshi Saruwatari[1], Ryoichi Miyazaki[1], Kiyohiro Shikano[1], Kazunobu Kondo[2]
[1]Nara Institute of Science and Technology, Nara, Japan,
[2]Yamaha Corporate Research & Development Center, Shizuoka, Japan

D-05

Multichannel Signal Enhancement Using a Remote Wireless Microphone

Brian Bloemendal[1], Jakob van de Laar[2], Piet Sommen[1]
[1]Eindhoven University of Technology, Eindhoven, Netherlands,
[2]Philips Research Laboratories, Eindhoven, Netherlands

D-06

Two Microphone Array MVDR Beamforming with Controlled Beamwidth and Immunity to Gain Mismatch

Yair Kerner, Harry Lau
Conexant Systems, Newport Beach, United States

D-07

Musical-Noise-Free Blind Speech Extraction Using ICA-Based Noise Estimation with Channel Selection

Ryoichi Miyazaki[1], Hiroshi Saruwatari[1], Kiyohiro Shikano[1], Kazunobu Kondo[2]
[1]Nara Institute of Science and Technology, Nara, Japan,
[2]Yamaha Corporate Research & Development Center, Shizuoka, Japan

D-08

Microphone Array Position Self-Calibration from Reverberant Speech Input

Florian Jacob, Joeg Schmalenstroeer, Reinhold Haeb-Umbach
University of Paderborn, Paderborn, Germany

D-09 **Increasing the Robustness of Acoustic Multichannel Equalization by Means of Regularization**

Ina Kodrasi[1], Stefan Goetze[2], Simon Doclo[1,2]
[1]University of Oldenburg, Oldenburg, Germany,
[2]Fraunhofer Institute for Digital Media Technology, Oldenburg, Germany

D-10 **Speaker Tracking for Teleconferencing via Binaural Headset Microphones**

Hannes Gamper, Sakari Tervo, Tapio Lokki
Aalto University School of Science, Espoo, Finland

10:30 – 11:00	Coffee Break

11:00 – 13:00	**Session E: Selected Papers (Chair: Walter Kellermann)**

E-01 **Fast Stereo Independent Vector Analysis and its Implementation on Mobile Phone**

Nobutaka Ono
National Institute of Informatics, Tokyo, Japan

E-02 **A Maximum A Posteriori Approach to Multichannel Speech Dereverberation and Denoising**

Dominic Schmid, Sarmad Malik, Gerald Enzner
Ruhr-University Bochum, Bochum, Germany

E-03 **Underdetermined DOA Estimation by the Non-Linear MUSIC Exploiting Higher-Order Moments**

Yuya Sugimoto[1], Shigeki Miyabe[1], Takeshi Yamada[1], Shoji Makino[1], Fred Juang[2]
[1]University of Tsukuba, Tsukuba, Japan,
[2]Georgia Institute of Technology, Atlanta, United States

E-04 **Blind Sampling Rate Offset Estimation and Compensation in Wireless Acoustic Sensor Networks with Application to Beamforming**

Shmulik Markovich-Golan[1], Sharon Gannot[1], Israel Cohen[2]
[1]Bar-Ilan University, Ramat Gan, Israel,
[2]Israel Institute of Technology, Haifa, Israel

E-05 **Iterative DFT-Domain Inverse Filter Determination for Adaptive Listening Room Equalization**

Martin Schneider, Walter Kellermann
University of Erlangen-Nuremberg, Erlangen, Germany

E-06 **Exploiting Temporal Correlations in Joint Multichannel Speech Separation and Noise Suppression Using Hidden Markov Models**

Dang Hai Tran Vu, Reinhold Haeb-Umbach
University of Paderborn, Paderborn, Germany

13:00 – 14:00	Lunch / Meeting of Technical Committee
14:00 – 19:00	Social Event
19:00 – 20:00	Cathedral Concert
20:00	Banquet

Sponsored by intel

Thursday, September 6th

09:00 – 09:30 Keynote Talk (Chair: Christiane Antweiler)

Distributed Signal Processing: Application to MVDR Beam-Forming
Richard Heusdens
Delft University of Technology, Delft, Netherlands

09:30 – 10:30 Poster Session F

F-01 **Optimized Preprocessing for Spatially Robust Room Impulse Response Reshaping**
Jan Ole Jungmann, Radoslaw Mazur, Alfred Mertins
University of Lübeck, Lübeck, Germany

F-02 **Scalable, Perceptual Based Echo Assessment Method for Aurally Adequate Evaluation of Residual Single Talk Echoes**
Marc Lepage, Frank Kettler, Jan Reimes
HEAD acoustics, Herzogenrath, Germany

F-03 **MMSE-Based Blind Source Extraction in Diffuse Noise Fields Using a Complex Coherence-Based a Priori SAP Estimator**
Maja Taseska, Emanuël Habets
International Audio Laboratories Erlangen, Erlangen, Germany

F-04 **Blind Audio Source Separation Exploiting Periodicity and Spectral Envelopes**
Siouar Bensaid, Dirk Slock
EURECOM, Sophia Antipolis, France

F-05 **Adaption of a Prediction Model for Noisy Speech Quality Assessment**

Jan Reimes, Hans Gierlich, Günter Mauer
HEAD acoustics, Herzogenrath, Germany

F-06 **Estimation of Direct-to-Reverberation Energy Ratio Based On Isotropic and Homogeneous Propagation Model**

Yusuke Hioka, Ken'ichi Furuya, Kenta Niwa, Yoichi Haneda
NTT Cyber Space Laboratories, Tokyo, Japan

F-07 **Discriminability Measure for Microphone Array Source Localization**

Leonardo Nunes[1], Wallace Martins[1], Markus Lima[1], Luiz Wagner Biscainho[1], Bowon Lee[2], Amir Said[2], Ronald Schafer[2]
[1]Federal University of Rio de Janeiro, Rio de Janeiro, Brazil,
[2]Hewlett-Packard Laboratories, Palo Alto, United States

F-08 **On Phase Importance in Parameter Estimation for Single-Channel Source Separation**

Pejman Mowlaee, Rainer Martin
Ruhr-University Bochum, Bochum, Germany

F-09 **Blind Matched Filtering for Speech Recording in Uncorrelated Noise**

Jürgen Freudenberger, Sebastian Stenzel
University of Applied Sciences Konstanz, Konstanz, Germany

F-10 **Blind Estimation of the Number of Speech Sources in Reverberant Multisource Scenarios Based on Binaural Signals**

Tobias May, Steven van de Par
University of Oldenburg, Oldenburg, Germany

10:30 – 11:00	Coffee Break

11:00 – 11:30	**Keynote Talk (Chair: Emanuël Habets)**

Optimized Directional Processing in Hearing Aids with Integrated Spatial Noise Reduction

Henning Puder, Eghart Fischer, Jens Hain
Siemens Audiologische Technik, Erlangen, Germany

11:30 – 12:30	**Poster Session G**

G-01 **An Update Algorithm for Frequency-Domain Correlation Model in a Nonlinear Echo Suppressor**

Osamu Hoshuyama
NEC Corporation, Saitama, Japan

G-02 **Optimized Spherical Sound Source for Room Reflection Analysis**

Martin Pollow, Johannes Klein, Pascal Dietrich, Gottfried Behler, Michael Vorländer
RWTH Aachen University, Aachen, Germany

G-03 **A Tradeoff Beamformer for Noise Reduction in the Spherical Harmonic Domain**

Daniel Jarrett[1,2], Emanuël Habets[2], Jacob Benesty[3], Patrick Naylor[1]
[1]Imperial College London, London, United Kingdom,
[2]International Audio Laboratories Erlangen, Erlangen, Germany,
[3]INRS-EMT, University of Quebec, Montreal, Canada

G-04 **Multi-Channel Noise Reduction in Hearing Aids with Wireless Access to an External Reference Signal**

Annelies Geusens[1], Alexander Bertrand[1,2], Bram Cornelis[1], Marc Moonen[1,2]
[1]KU Leuven, Leuven, Belgium,
[2]IBBT Future Health Department, Leuven, Belgium

G-05 **STFT Phase Improvement for Single Channel Speech Enhancement**

Martin Krawczyk, Timo Gerkmann
University of Oldenburg, Oldenburg, Germany

G-06 **A Psychoacoustic-Based Analysis of the Impact of Pre-Echoes and Post-Echoes in Soundfield Rendering Applications**

Lucio Bianchi, Fabio Antonacci, Antonio Canclini, Augusto Sarti, Stefano Tubaro
Politecnico di Milano, Milano, Italy

G-07 **Underdetermined Source Detection and Separation Using a Normalized Multichannel Spatial Dictionary**

Mahmoud Fakhry, Francesco Nesta
Fondazione Bruno Kessler, Trento, Italy

G-08 **Relaxed Multichannel Least Squares with Constrained Initial Taps for Multichannel Dereverberation**

Felicia Lim, Patrick Naylor
Imperial College London, London, United Kingdom

G-09 **ViSQOL: The Virtual Speech Quality Objective Listener**

Andrew Hines[1], Jan Skoglund[2], Anil Kokaram[2], Naomi Harte[1]
[1]Trinity College Dublin, Dublin, Irland,
[2]Google, Mountain View, United States

G-10 **Evaluation of a Speech Bandwidth Extension Algorithm Based on Vocal Tract Shape Estimation**

Itai Katsir, David Malah, Israel Cohen
Israel Institute of Technology, Haifa, Israel

12:30 – 14:00	Lunch

14:00 – 14:30	Keynote Talk (Chair: Christophe Beaugeant)

Multi-Microphone Speech Enhancement

Sharon Gannot
Bar-Ilan University, Ramat Gan, Israel

14:30 – 16:00	Poster Session H

H-01 **Posterior Residual Echo Canceling and its Complexity Reduction in the Wave Domain**

Satoru Emura, Shoichi Koyama, Ken'ichi Furuya, Yoichi Haneda
NTT Cyber Space Laboratories, Tokyo, Japan

H-02 **Connections Between Parallel and Serial Combinations of Comb Filters and Feedback Delay Networks**

Sebastian Schlecht, Emanuël Habets
International Audio Laboratories Erlangen, Erlangen, Germany

H-03 **Robustness Analysis of Speech Enhancement Using a Bone Conduction Microphone – Preliminary Results**

Sriram Srinivasan, Patrick Kechichian
Philips Research, Eindhoven, Netherlands

H-04 **Theoretical Performance Analysis of ANC-Motivated Noise Reduction Algorithms for Open-Fitting Hearing Aids**

Derya Dalga, Simon Doclo
University of Oldenburg, Oldenburg, Germany

H-05 **Distortionless-Response vs. Matched-Filter-Array Processing for Adaptive Binaural Noise Reduction**

Masoumeh Azarpour, Gerald Enzner, Rainer Martin
Ruhr-University Bochum, Bochum, Germany

H-06 **3D Spatial Soundfield Recording over Large Regions**

Prasanga Samarasinghe[1], Thushara Abhayapala[1], Mark Poletti[2]
[1]Australian National University, Canberra, Australia,
[2]Industrial Research Ltd, Lower Hutt, New Zealand

H-07 **Distributed MVDR Beamforming for (Wireless) Microphone Networks Using Message Passing**

Richard Heusdens, Guoqiang Zhang, Richard Hendriks, Yuan Zeng, W. Bastiaan Kleijn
Delft University of Technology, Delft, Netherlands

H-08 **A Multiple Hypothesis Gaussian Mixture Filter for Acoustic Source Localization and Tracking**

Youssef Oualil[1,2], Friedrich Faubel[1], Dietrich Klakow[1]
[1]Saarland University, Saarbrücken, Germany,
[2]Idiap Research Institute, Martigny, Switzerland

H-09 **Blind Separation of Infinitely Many Sparse Sources**

Hirokazu Kameoka[1,2], Misa Sato[1], Takuma Ono[1], Nobutaka Ono[3], Shigeki Sagayama[1]
[1]University of Tokyo, Tokyo, Japan,
[2]NTT Communication Science Laboratories, Kyoto, Japan,
[3]National Institute of Informatics, Tokyo, Japan

H-10 **Crosstalk Cancellation System Using a Head Tracker Based on Interaural Time Differences**

Yesenia Lacouture-Parodi, Emanuël Habets
International Audio Laboratories Erlangen, Erlangen, Germany

H-11 **Performance Comparison of Algorithms for Blind Reverberation Time Estimation from Speech**

Nikolay Gaubitch[1], Heinrich Löllmann[2], Marco Jeub[2], Tiago Falk[3], Patrick Naylor[1], Peter Vary[2], Mike Brookes[1]
[1]Imperial College London, London, United Kingdom,
[2]RWTH Aachen University, Aachen, Germany,
[3]Institute National de la Recherche Scientifique, Montreal, Canada

14:30 – 16:00	Demonstrator Session	(Room 5.31/5.32)

DH-01 **Demonstrator: Microphone Array Position Self-Calibration from Reverberant Speech Input**

Florian Jacob, Joerg Schmalenstroeer
University of Paderborn, Paderborn, Germany

DH-02 **Connected Visual Reality – High Quality Audio Visual Communication in Heterogeneous Networks**

Laurits Hamm[1], Tobias Engelbert[1], Jose Lausuch[1], Arturo Martin De Nicolas[1], Ramsundar Kandasamy[1], Martin Schink[2], Christian Feldmann[3], Christopher Bulla[3], Magnus Schäfer[3], Florian Heese[3], Thomas Schlien[3], Christiane Antweiler[3]
[1]Ericsson, Aachen, Germany,
[2]Rovi MainConcept, Aachen, Germany,
[3]RWTH Aachen University, Aachen, Germany

DH-03 **The Software Defined Terminal**

Hauke Krüger, Thomas Schlien, Matthias Rüngeler, Peter Vary
RWTH Aachen University, Aachen, Germany

16:00 – 16:30	Best Paper Award Closing Ceremonies	**(Chair: Peter Vary)**

Social Events

Welcome Reception: Monday, September 3rd, 19:00

All IWAENC 2012 attendees are invited to join us for the welcome reception at the 6th floor of the SuperC at RWTH Aachen University.

Drinks and light appetizers will be served.

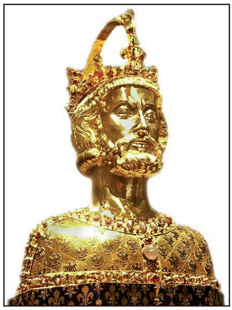
Sponsored by HEAD acoustics®

SuperC, RWTH Aachen University

City Tour: Wednesday, September 5th, 14:00

During the city tour you will have the opportunity to participate in different activities:

Tour of the Old Town
The historic old town of Aachen invites to go for a stroll. Let yourself be guided through narrow alleys and across historic squares through the 2000 year-old history of Aachen. Experience all facets of Aachen, a modern city with beautiful historic town houses, many old and new fountains, and innumerable stories all about the cathedral and the town hall.



Charlemagne

Treasury of the Cathedral of Aachen
The Cathedral Treasury in Aachen is regarded as the most important ecclesiastical treasury north of the Alps. It houses sacral art treasures from the late Antique, Carolingian, Ottonian and Staufian epoch, among them some unique exhibits like the „Cross of Lothair", the „Bust of Charlemagne", and the „Persephone sarcophagus". These and all the other exhibits document the importance of the Cathedral of Aachen as a place of the medieval treasury.

Printenbäckerei Klein
Printen are a type of gingerbread originating from the city of Aachen where they have been produced since the 15th century. Originally, the Printen were sold by pharmacists since some of their ingredients were considered to possess medical benefits. Printen are made from a variety of ingredients including cinnamon, aniseed, clove, cardamom, coriander, allspice, and also ginger. The exact mixture of these ingredients, however, is a close kept secret of the individual Printen bakeries. In the Printenbäckerei Klein visitors have the opportunity to peek behind the curtains of a real Printen bakery.

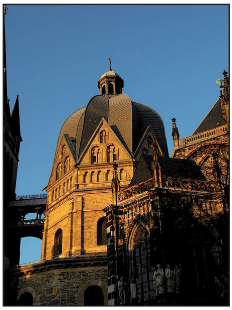
Aachener Printen

Aachen Cafe

During the city tour you have the opportunity to visit a typical Aachen cafe for a refreshment.

Cathedral Concert: Wednesday, September 5th, 19:00

The city tour will end with a small guided tour and a private concert in the Cathedral of Aachen.



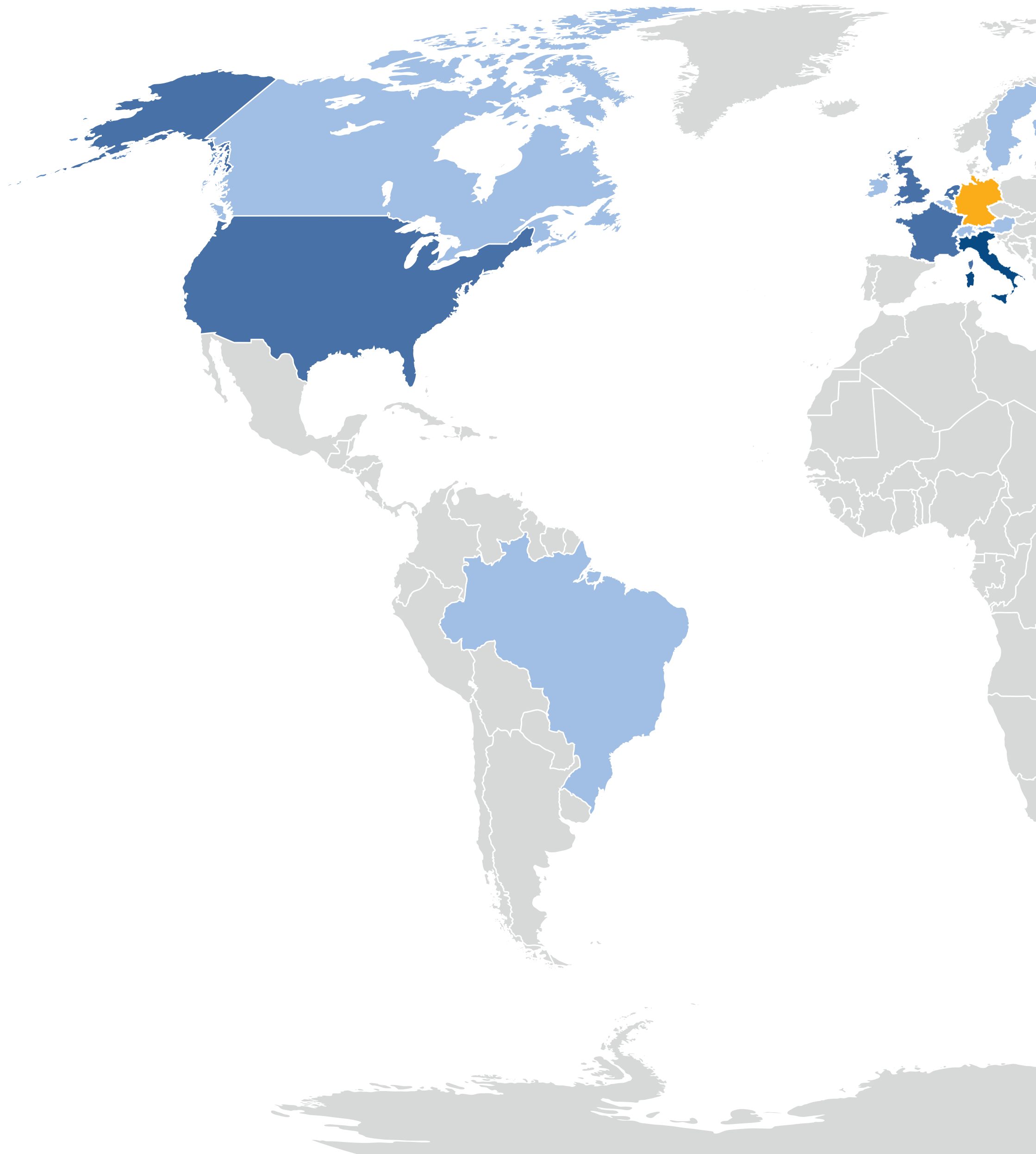
Cathedral of Aachen

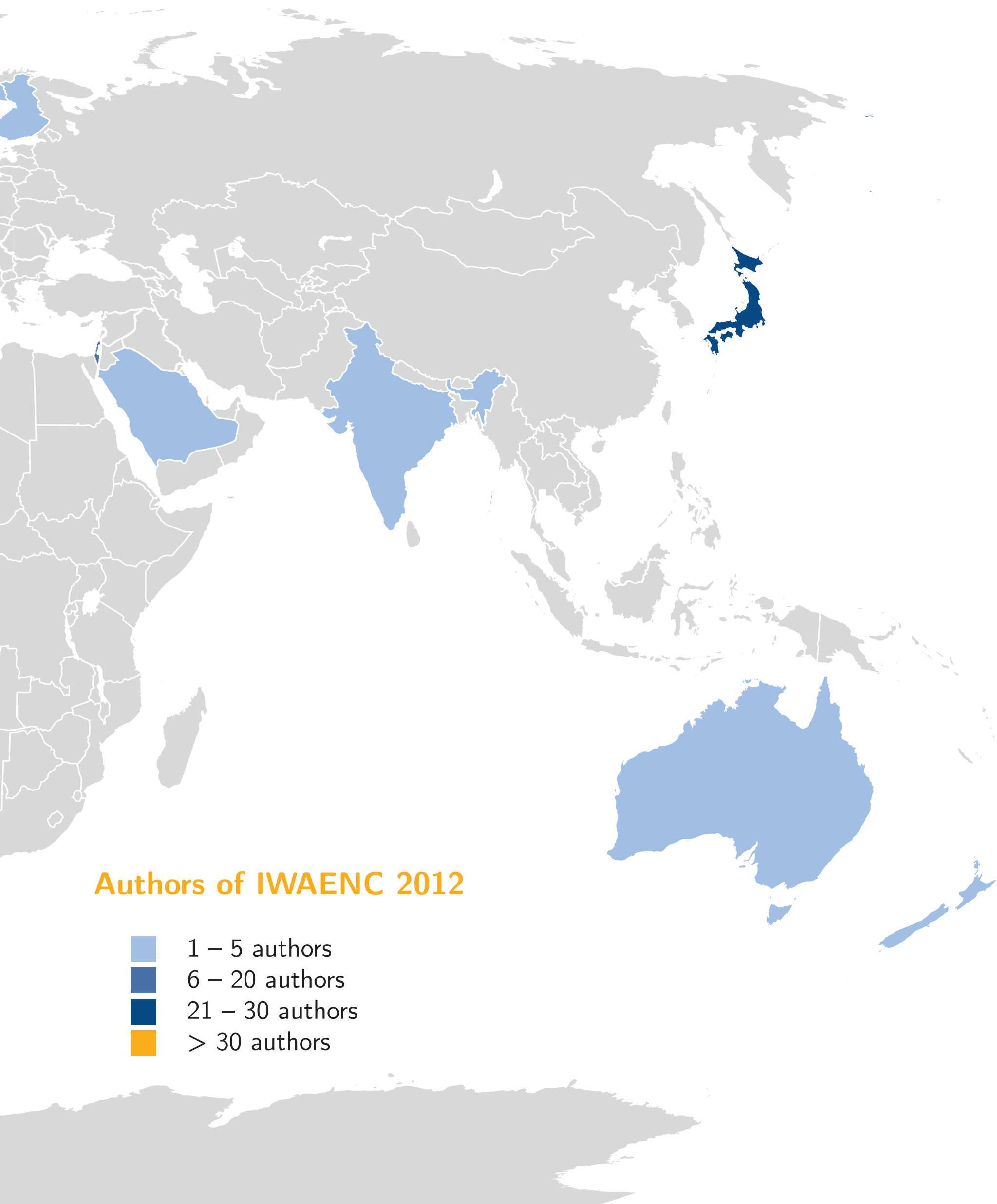
Banquet: Wednesday, September 5th, 20:00

After the concert, the banquet will be held at the Aula Carolina. The Aula Carolina, located in the historic center of Aachen, is a former monastery church and has been built in 1663.

Sponsored by intel

Author Index




Authors of IWAENC 2012
1 – 5 authors
6 – 20 authors
21 – 30 authors
> 30 authors

Imprint

Conference Secretariat

SuperC, RWTH Aachen University

Templergraben 57, 6th floor
52062 Aachen
Germany

Phone: +49 241 80-90799
Email: info@iwaenc2012.org

Office Hours

Monday,	September 3rd:	17:00 – 21:00
Tuesday,	September 4th:	08:00 – 18:00
Wednesday,	September 5th:	08:00 – 14:00
Thursday,	September 6th:	08:00 – 17:00

Picture Credits

p. 36: SuperC, Markus Klügel
p. 37: Aachener Printen, Elena Schevardo

The following pictures have been taken from Wikimedia Commons (http://commons.wikimedia.org/) and are licensed under the Creative Commons Attribution-ShareAlike 2.0 Generic License.

p. 36: Charlemagne, Lokilech
p. 37: Cathedral of Aachen, MBO163

Program at a Glance

Monday September 3rd	Tuesday September 4th	Wednesday September 5th	Thursday September 6th
	9:00 – 9:15 Welcome	**9:00 – 9:30** Keynote Talk Patrick Naylor	**9:00 – 9:30** Keynote Talk Richard Heusdens
	9:15 – 10:15 Keynote Talk Peter Kroon	**9:30 – 10:30** Session D	**9:30 – 10:30** Session F
	Coffee Break	Coffee Break	Coffee Break
	10:45 – 11:15 Keynote Talk Peter Jax	**11:00 – 13:00** Session E Selected Papers	**11:00 – 11:30** Keynote Talk Henning Puder
	11:15 – 12:30 Session A & Demonstrators		**11:30 – 12:30** Session G
	Lunch	Lunch	Lunch
	14:00 – 14:30 Keynote Talk Jan Skoglund	**14:00 – 19:00** Social Event	**14:00 – 14:30** Keynote Talk Sharon Gannot
	14:30 – 16:00 Session B		**14:30 – 16:00** Session H & Demonstrators
	Coffee Break		**16:00 – 16:30** Best Paper Award & Closing Ceremonies
	16:30 – 17:00 Keynote Talk Bernd Geiser		
17:00 – 19:00 Registration	**17:00 – 18:00** Session C	**19:00 – 20:00** Cathedral Concert	
19:00 – 21:00 Welcome Reception		**20:00** Banquet	

ind

RWTH AACHEN UNIVERSITY