Final program ARCHES Meeting / ICanHear Conference 2016

University of Zurich[UZH]

ICanHear
Improved Communication through Applied Hearing Research

ARCHES Meeting 2016
ICanHear Conference

Monday 21st – Wednesday 23rd November 2016

PROGRAM AND ABSTRACTS

Introduction

Dear Friends and Colleagues,

The next ARCHES meeting will take place in Zurich on Monday/Tuesday November 21/22 as promised last year during the ARCHES 2015 meeting in Groningen.

This year, there will be the ICanHear conference as a satellite meeting immediately following the ARCHES meeting. The ICanHear conference is open to all interested persons and we warmly welcome a combined registration for both events.

We are very pleased to welcoming you all together with your PostDocs and PhD students for a short but interesting workshop. We organized the meeting and conference within our University which is conveniently located in the center of Zurich and easily accessible from the airport or train station.

Following the discussions during the last meeting and your submissions we will stay with the general framework of the ARCHES meeting. There will be joint project talks and additional individual presentations. The poster sessions will be introduced by short (1 to 2 minutes) presentations.

The ICanHear conference includes plenary keynote lectures, short talks, poster presentations and demonstration examples of real-time implementation of signal processing algorithms.

Presentations and posters will be made accessible for registered participants after the meeting on a protected website page.

Hotel room block reservations have been made in a nearby hotel. Please make sure to book as soon as possible in order to take advantage of the special rates.

We wish you a productive and interesting meeting!

The Zurich ARCHES/ICanHear 2016 Team

Claudia Stenger
Sonia Tabibi
Dietmar Wohlbauer
Andrea Kegel
Wai Kong Lai
Norbert Dillier

The ICanHear conference is funded by the people Programme (Marie Curie Actions) of the European Union's Seventh Framework Programme FP7/2007-2013/ under REA grant agreement No. PITN-GA-2012-317521.

ICanHear
Improved Communication through Applied Hearing Research

We gratefullly acknowledge sponsorship of Cochlear, Sivantos and Speedgoat (ICanHear Conference) and Sonova (ARCHES meeting)

Cochlear™

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the hearing company

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real-time simulation and testing

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HEAR THE WORLD

ZIHP

Zurich Center for Integrative Human Physiology (ZIHP), University of Zurich (UZH)

Venue

The ARCHES meeting and the ICanHear Conference will both be held in the **RAA** (Alte Kantonsschule Zürich) building of the University of Zurich, **Rämistrasse 59** (about 100 m distance from main University building).

Oral presentations will be held in the **Aula RAA-G1**.

Poster walls are located in the Lichthof of the RAA building where also the welcome lunch and the coffee breaks will take place. The demonstration sessions of the ICanHear Conference will be held on the groundfloor location as well.

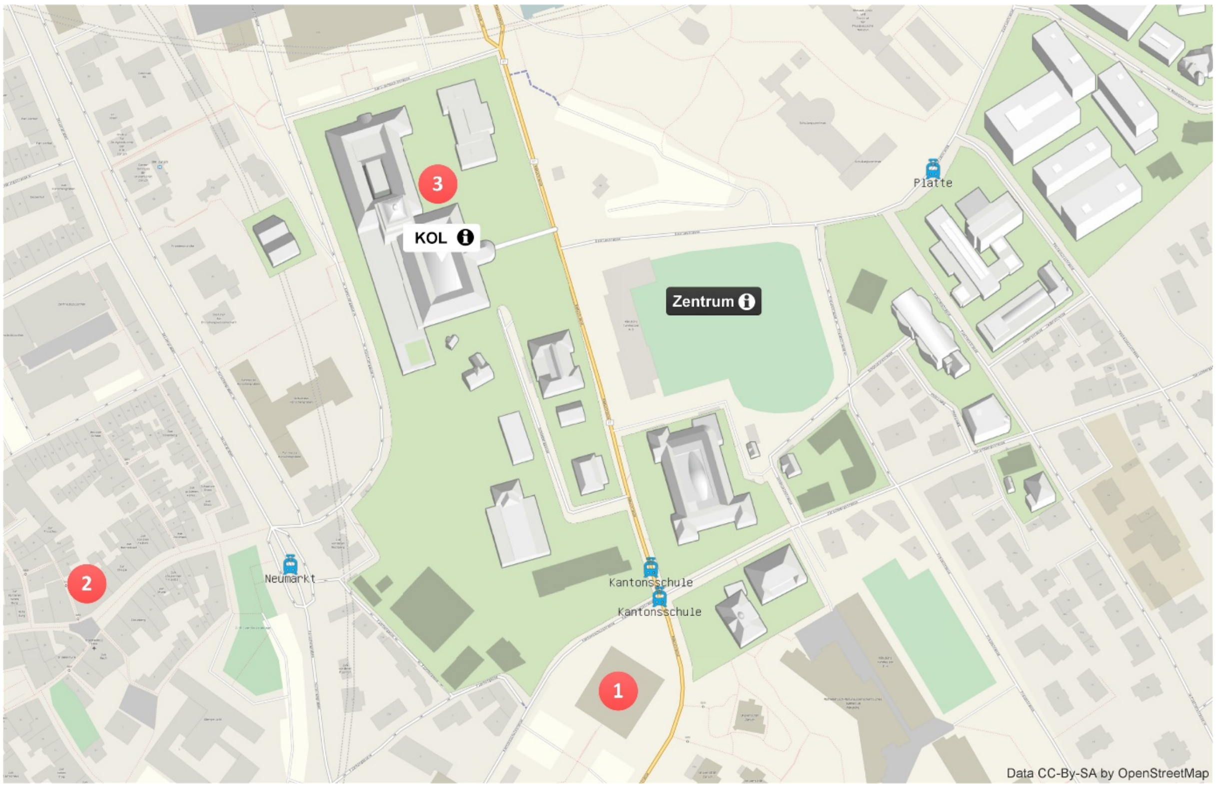
How to get there

Getting to the University from the airport or the main station of Zurich is easy. There is one tram line (number 10) which connects the airport with the railway main station. The nearest tram stop for the University main building is ETH/Universitätsspital (exit at tram stop Winkelriedstrasse to get to the Hotel Rigihof) From the Hotel Rigihof you can take **tram number 9** downhill and exit at the third stop (**Kantonsschule**) which is just in front of the venue (**Rämistrasse 59**).

Alternatively, you may take a fast train from the airport to the city (Zurich main station) and then either walk uphill to the University or take tram 10 or 6. If you arrive from the lake side (Bellevue)you may take tram number 5 or 9 and exit at the station "Kantonsschule".

Please note that tickets for public transport within Zurich and surroundings can be purchased at the tram or train station. Tickets are available for either a short/long ride or 24 hours.

To get to the Aula lecture hall from the tram stop, enter the RAA building (Alte Kantonsschule Zürich, Rämistrasse 59) and take the stairs to the G floor”



1. Aula Alte Kantonsschule Zürich, Rämistrasse 59 (AKSZ, RAA59)
2. Zunfthaus und Wirtschaft zum Neumarkt, Neumarkt 5
3. UniTurm Restaurant, University of Zurich, Rämistrasse 71

About ARCHES

ARCHES (Audiological Research Cores in Europe) is a European network of research groups focusing on hearing science, with the aim to stimulate networking, interaction, and scientific collaboration between researchers in the auditory field.

The following research groups take currently part in ARCHES:

- [BE] Katholieke Universiteit Leuven (Leuven, Belgium)
  Division of Experimental Otorhinolaryngology
  Department of Neurosciences
  Contact: Astrid van Wieringen, Jan Wouters
  Lab website: http://gilbert.med.kuleuven.be/
- [CH] University Hospital Zürich (Zürich, Switzerland)
  Laboratory of Experimental Audiology
  Department of Otorhinolaryngology, Head and Neck Surgery
  Contact: Norbert Dillier
  Lab website: http://www.uzh.ch/orl/lea/lea.html
- [DE] Carl von Ossietzky Universität (Oldenburg, Germany)
  + Hörzentrum Oldenburg
  Medical Physics Section
  Institute of Physics
  Contact: Birger Kollmeier
  Lab website: http://medi.uni-oldenburg.de/
- [DK] Technical University of Denmark (Kgs. Lyngby, Denmark)
  + Centre for Applied Hearing Research
  Department of Electrical Engineering
  Contact: Torsten Dau
  Lab website: http://www.cahr.elektro.dtu.dk/
- [FR] Université Paris Descartes (Paris, France)
  + Ecole Normale Supérieure
  + Centre National de la Recherche Scientifique
  Laboratoire Psychologie de la Perception
  Contact: Christian Lorenzi
  Lab website: http://lpp.psycho.univ-paris5.fr/
- [NL] Academic Medical Center (Amsterdam, The Netherlands)
  Department of Clinical and Experimental Audiology
  Contact: Wouter Dreschler
  Lab website: www.amc.nl/web/Research/Overview/Departments/Ear-Nose-and-Throat/Ear-Nose-and-Throat/Researchers.htm?p=401
- [NL] VU University Medical Center (Amsterdam, The Netherlands)
  Department of Otolaryngology, Head and Neck Surgery
  Contact: Theo Goverts, Cas Smits
  Lab website: http://www.vumc.com/branch/Otolaryngology/
- [NL] University Medical Center (Groningen, The Netherlands)
  Department of Otorhinolaryngology
  Contact: Pim van Dijk, Deniz Baskent
  Lab website: http://www.rug.nl/staff/d.baskent/
- [UK] Nottingham University (Nottingham, United Kingdom)
  MRC Institute of Hearing Research

Contact: Michael Akeroyd
Lab website: www.ihr.mrc.ac.uk/

Previous ARCHES meetings:

- 2007, Oldenburg
- 2008, Haarlem
- 2009, Nottingham
- 2010, Zurich
- 2011, Leuven
- 2012, Copenhagen
- 2013, Paris
- 2014, Oldenburg
- 2015, Groningen

About ICanHear

ICanHear (Improved Communication through Applied Hearing Research) is a Marie Curie Initial Training Network program which has received funding from the European Union's Seventh Framework Programme for research, technological development and demonstration under grant agreement no 317521.

ICanHear has developed models based on emerging knowledge about higher-level processing within the auditory pathway and exploits that knowledge to develop creative solutions that will improve the performance of hearing instruments.

New signal processing methods were implemented on a unified open signal processing platform and evaluated in terms of user benefits on common evaluation scenarios and data bases.

A distinctive feature of this project is the tight integration of academic and industrial partners. Fellows are trained to use rapid prototyping tools to implement and to test their newly developed solutions. Thus they become familiar with the various stages of product development and also contribute by providing suggestions for improvement of such tools and future product development platforms.

More information and project examples can be found on the ICanHear website: http://www.icanhear.eu

Program

Time	*Presentations*	*Authors*
12:00	**Monday November 21st 2016**	
12:00	**Lunch, registration**	
13:00	**Welcome**	Norbert Dillier ORL-LEA USZ
13:10	**Panoramic talk about projects at DTU**	Torsten Dau DTU
13:25	**Noise-robust neural tracking of attended talkers in real-world acoustic scenarios**	Søren Fuglsang, Torsten Dau, Jens Hjortkjaer Hearing Systems Group, Technical University of Denmark
13:40	**The effect of dynamic range compression on spatial perception in a reverberant environment**	Henrik Gert Hassager, Alan Wiinberg, Torsten Dau Hearing Systems group, Department of Electrical Engineering, Technical University of Denmark, DK-2800 Kgs. Lyngby, Denmark
13:55	**Aided Patient Performance Predictions with FADE**	Anna Warzybok, Marc René Schädler, Birger Kollmeier Universität Oldenburg
14:10	**Sharpening of Cortical Frequency Selectivity via Frequency Specific Adaptation**	Oscar Woolnough, Jessica de Boer, Katrin Krumbholz, Rob Mill and Chris Sumner MRC Institute of Hearing Research, University Park, Nottingham, NG7 2RD, UK
14:25	**Audiological research at Ecole normale superieure, Paris, France**	Christian Lorenzi Ecole normale superieure & CNRS
14:40	**Short presentations of posters**	
14:40	**Responses to sinusoidal frequency modulation in the guinea pig ventral cochlear nucleus**	Nihaad Paraouty (1,2); Arkadiusz Stasiak (1); Christian Lorenzi (2); Ian M. Winter (1) 1) Physiological Laboratory, University of Cambridge, United Kingdom 2) Ecole Normale Superieure Paris LSP UMR 8248, PSL, France
14:42	**Comparing the effects of age and sensorineural hearing loss on detection and temporal integration of amplitude and frequency modulation**	Nicolas Wallaert (1), Brian C. J. Moore (2), Stephan D. Ewert (3), Christian Lorenzi (1) 1) UMR CNRS LSP 8248, Institut d'Etude de la Cognition, Ecole normale supérieure, Paris Sciences et Lettres Research University, 29 rue d'Ulm, 75005 Paris, France 2) Department of Experimental Psychology, University of Cambridge,Downing street, Cambridge CB2 3EB, United Kingdom 3) Medizinische Physik and Cluster of

		Excellence Hearing4All, Universität Oldenburg, 26111 Oldenburg, Germany
14:44	**Influence of attention on speech-rhythm evoked potentials: first steps towards a speech driven brain-computer-interface**	Carlos da Silva Souto, Helge Lüddemann, Silvia Lipski, Mathias Dietz and Birger Kollmeier. Medizinische Physik and Cluster of Excellence Hearing4all, Universität Oldenburg, D-26111 Oldenburg, Germany
14:46	**Common Audiological Functional Parameters (CAFPAs): an abstract representation of audiological expert data**	Mareike Buhl, Marc René Schädler, Anna Warzybok, Birger KollmeierCvO Universität Oldenburg, Medical Physics and Cluster of Excellence Hearing4all (all authors)
14:48	**Frequency dependency of binaural masking level differences in NH and HI listeners**	Christopher F. Hauth, Thomas Brand and Birger Kollmeier Medizinische Physik and Cluster of Excellence "Hearing4All", Carl von Ossietzky (CvO) Universität Oldenburg, 26129 Oldenburg, Germany
14:50	**Functional activity in the right auditory cortex reflects individual pitch-discrimination abilities in musicians**	Federica Bianchi (1,2), Jens Hjortkjær (1,2), Sébastien Santurette (1,3), Hartwig R. Siebner (2,4), Robert J. Zatorre (5) and Torsten Dau (1) 1) Hearing Systems Group, Department of Electrical Engineering,Technical University of Denmark, Bygning 352,Ørsteds Plads, 2800 Kgs. Lyngby, Denmark 2) Danish Research Centre for Magnetic Resonance, Centre for Functional and Diagnostic Imaging and Research, Copenhagen University Hospital Hvidovre, 2650 Hvidovre, Denmark 3) Department of Otorhinolaryngology, Head and Neck Surgery & Audiology, Rigshospitalet, 2100 Copenhagen, Denmark 4) Department of Neurology, Copenhagen University Hospital Bispebjerg, 2400 Copenhagen, Denmark 5) Montreal Neurological Institute, McGill University, and BRAMS, Montreal, Canada
14:52	**Evaluating fast-acting compression in basic psychoacoustic and speech tasks**	Borys Kowalewski, Michal Fereczkowski, Ewen MacDonald, Olaf Strelcyk, Torsten DauHearing Systems group, Hearing Systems group, Hearing Systems group, Sonova U.S. Corporate Services, Hearing Systems group

14:54	**Objective assessment of voluntary stream segregation abilities of CI users**	Andreu Paredes Gallardo, Sara Miay Kim Madsen, Torsten Dau and Jeremy Marozeau Hearing Systems group, Department of Electrical Engineering, Technical University of Denmark, DK-2800, Kgs. Lyngby, Denmark
14:56	**A correlation metric in the envelope power spectrum domain for speech intelligibility prediction**	Helia Relaño Iborra, Tobias May, Johannes Zaar, Christoph Scheidiger, Torsten Dau Hearing Systems group, Department of Electrical Engineering, Technical University of Denmark, DK-2800, Kgs. Lyngby, Denmark
14:58	**Archetypal analysis of auditory profiling data towards a clinical test battery**	Raul H. Sanchez(1), Federica Bianchi(1), Sébastien Santurette(1,2) and Torsten Dau(1) 1) Hearing Systems group, Department of Electrical Engineering, Technical University of Denmark, DK-2800, Kgs. Lyngby, Denmark 2) Department of Otorhinolaryngology, Head and Neck Surgery and Audiology, Rigshospitalet, Copenhagen, Denmark
15:00	**Coffee break & Posters**	
16:00	**Overview of Auditory Research in Salamanca**	Enrique A. Lopez-Poveda University of Salamanca, Salamanca, Spain
16:10	**On the Value of Brief Sound Audiometry as a Diagnostic Tool for Cochlear Synaptopathy**	Enrique A. Lopez-Poveda (1), Peter T. Johannesen (1), Byanka C. Buzo (1), Filip Rønne (2), Niels H. Pontoppidan (2), James M. Harte (3) 1) University of Salamanca, Salamanca, Spain 2) Eriksholm Research Centre, Snekkersten, Denmark 3) Interacoustics Research Unit, Lyngby, Denmark
16:25	**Medial Olivocochlear Reflex Effects on Amplitude Modulation Detection**	Enrique A. Lopez-Poveda (1), Miriam Marrufo-Pérez (1), Luís E. López-Bascuas (2), Almudena Eustaquio-Martín (1) 1) University of Salamanca, Salamanca, Spain 2) Universidad Complutense de Madrid, Madrid, Spain.
16:40	**Synaptopathy with envelope following responses (EFRs): The off-frequency problem**	Gerard Encina-Llamas (1), Aravind Parthasarathy (2), James M. Harte (3), Torsten Dau (1), Sharon Kujawa (2), Barbara Shinn-Cunningham (4), Bastian Epp (1) 1) Hearing Systems group. Technical University of Denmark

		2) Eaton-Peabody Labs. Massachussets Eye and Ear Infirmary. Harvard Medical School 3) Interacoustics Research Unit 4) Boston University
16:55	**Panoramic talk on projects at Laboratory of Experimental Audiology, UZH/USZ**	Norbert Dillier Laboratory for Experimental Audiology, University & University Hospital Zurich
17:05	**Low speech intelligibility in older adults - The aging ear or brain?**	Andrea Kegel (1), Norbert Dillier (1), Nathalie Giroud (2), Martin Meyer (2) 1) Laboratory for Experimental Audiology, Department of Oto-Rhino-Laryngology, Head and Neck Surgery, University Hospital Zurich 2) Neuroplasticity and Learning in the Healthy Aging Brain, University of Zurich
17:20	**Neuroanatomical and intrinsic cortical oscillatory correlates of central hearing loss in older adults**	Nathalie Giroud (1), Sarah Hirsiger (2), Raphaela Muri (1), Andrea Kegel (3), Norbert Dillier (3), Martin Meyer (1) 1) University of Zurich 2) Psychiatric Hospital of the University of Zurich 3) University Hospital Zurich
17:35	**Short presentations of posters**	
17:35	**Improving rehabilitation of the hearing impaired based on profiling and COSI: results of a pilot with an improved protocol in the Netherlands.**	Wim Soede (1), Bert van Zanten (2) and Wouter A. Dreschler (3) 1) Leiden University Medical Center, The Netherlands 2) Dept of Audiology, Utrecht University Medical Center, The Netherlands 3) Academic Medical Center, The Netherlands
17:37	**Critical factors in hearing aid selection and evaluation: audiological characteristics, self-report profiles, and hearing aid properties.**	Wouter A. Dreschler (1), Inge de Ronde-Brons (1), Monique Boymans (1) and Wim Soede (2)1) Academic Medical Center, The Netherlands2) Leiden University Medical Center, The Netherlands
17:39	**Frequency-domain Reduced-rank Approximations of Music Signals for the Improvement of Music Perception in Cochlear Implant Listeners**	Anil Nagathil (1), Claus Weihs (2), Katrin Neumann (3), Rainer Martin (1) 1) Institute of Communication Acoustics (IKA), Faculty of Electrical Engineering and Information Technology, Ruhr-Universität Bochum, Germany 2) Chair of Computational Statistics, Faculty of Statistics, TU Dortmund, Germany 3) Abt. für Phoniatrie und Pädaudiologie, Klinik für Hals-, Nasen- und Ohrenheilkunde, Kopf- und

		Halschirurgie, St. Elisabeth-Hospital, Ruhr-Universität Bochum, Germany
17:41	**The eyes and brain reveal your hearing ability**	Adriana A. Zekveld (1,2), Dirk J. Heslenfeld (3), Niek J. Versfeld (1), Sophia E. Kramer (1) 1) Section Ear & Hearing, Dept. of Otolaryngology-Head and Neck Surgery and EMGO Institute for Health and Care Research, VU University medical center, De Boelelaan 1118, 1081 HZ, Amsterdam, The Netherlands 2) Department of Behavioural Sciences and Learning, Linnaeus Centre HEAD, The Swedish Institute for Disability Research, Linköping University, SE-581 83 Linköping, Sweden 3) Department of Psychology, VU University, Van der Boechorststraat 1, 1081 BT Amsterdam, The Netherlands
17:43	**Amplitude-modulation masking for frequency-modulation detection**	Andrew J King, Christian Lorenzi Laboratoire des systemes perceptifs, CNRS, Ecole normale superieure, Paris Sciences et Lettres Research University, 29 rue d'Ulm, 75005 Paris, France
17:45	**Salicylate-induced changes in brain activity in awake guinea pigs**	JI Berger, B Coomber, A Hockley, W Owen, MN Wallace and AR Palmer 1) Medical Research Council Institute of Hearing Research, Nottingham, NG7 2RD, UK 2) School of Medicine, University of Nottingham, Nottingham, NG7 2RD, UK
17:47	**Measuring tonotopic magnification in the human auditory cortex.**	Ben Gurer, Julien Besle & Katrin Krumbholz Medical Research Council Institute of Hearing Research, The University of Nottingham, University Park, Nottingham, NG7 2RD
17:49	**Posters**	
18:30	**Apéro & Dinner (Zunfthaus zum Neumarkt)**	

08:30	**Tuesday November 22nd 2016**	
08:30	**Panoramic talk about projects at AMC**	Wouter A. Dreschler Academic Medical Center, The Netherlands
08:40	**Panoramic talk about hearing Research in Oldenburg during the last year**	Birger Kollmeier Medizinische Physik, Universität Oldenburg & Cluster of Excellence "Hearing4all"
08:50	**Spectral and binaural loudness summation and the need for individual assessment in auditory rehabilitation.**	Mirjam van Geleuken, Monique Boymans, and Wouter A. Dreschler Academic Medical Center, The Netherlands
09:05	**Restoring loudness perception for hearing-impaired listeners**	Dirk Oetting (1), Volker Hohmann (2), Jens-E. Appell (1), Birger Kollmeier (2), Stephan D. Ewert (2) 1) Project Group Hearing, Speech and Audio Technology of the Fraunhofer IDMT and Cluster of Excellence Hearing4all, Oldenburg, Germany 2) Medizinische Physik and Cluster of Excellence Hearing4all, Universität Oldenburg,26111 Oldenburg, Germany
09:20	**Listener's preference for time constant settings in a noise-reduction algorithm**	Ilja Reinten, Inge de Ronde Brons, Wouter A. DreschlerAcademic Medical Center, The Netherlands
09:35	**Coffee break & Posters** **ARCHES group leader meeting**	
10:35	**Panoramic talk on projects at VUMC**	Theo Goverts VUMC
10:45	**Impact of stimulus-related factors and hearing status on listening effort as indicated by pupil dilation**	Adriana A. Zekveld (1,2,3), Barbara Ohlenforst (1,4), Thomas Lunner (3,4,5), Dorothea Wendt (4,6), Graham Naylor (7), Yang Wang (1,4), Niek J. Versfeld (1), Sophia E. Kramer (1) 1) Section Ear & Hearing, Dept. of Otolaryngology-Head and Neck Surgery, VU University Medical Center and EMGO Institute for Health Care Research, Amsterdam, The Netherlands 2) Department of Behavioral Sciences and Learning, Linköping University, inköping, Sweden 3) Linnaeus Centre HEAD, The Swedish Institute for Disability Research, Linköping and Örebro Universities, Linköping, Sweden 4) Eriksholm Research Centre, Oticon A/S, Snekkersten, Denmark 5) Department of Clinical and Experimental Medicine, Linköping University, Sweden

		6) Technical University of Denmark, Department of Electrical Engineering, Lyngby 7) MRC/CSO Institute of Hearing Research, Scottish Section, Glasgow, United Kingdom
11:00	**Panoramic talk about projects at KU Leuven**	Jan Wouters KU Leuven
11:10	**Can bimodal cochlear implant listeners use true binaural hearing to understand speech in noise?**	Dieudonné, B.; Francart, T.KU Leuven, ExpORL, Department of Neurosciences, Herestraat 49 bus 721, 3000 Leuven, Belgium.
11:25	**Auditory Steady-State Responses as a measure to evaluate temporal processing of speech envelope characteristics in adults with dyslexia**	Tilde Van Hirtum (1,2), Pol Ghesquière (2) & Jan Wouters (1) 1) ExpORL, Department of Neurosciences, KU Leuven, Belgium 2) Parenting and Special Education, KU Leuven, Belgium
11:40	**ECC neurophysiologically based coding strategy for cochlear implants: Real time implementation considerations**	Waikong Lai (1), Norbert Dillier (1), Matthijs Killian (2) 1) ORL Klinik UniversitätsSpital Zürich (CH), 2) Cochlear Technology Center Mechelen (BE)
11:55	**General discussion, next meeting**	All ARCHES members
12:30	**Lunch (Lichthof)**	
13:30	**Registration ICanHear Conference**	
13:45	**Welcome**	Rainer Martin Ruhr-Universität Bochum
14:00	**Models and mechanisms of temporal interactions in cochlear implant stimulation of the auditory nerve**	Ian C. Bruce Department of Electrical and Computer Engineering, McMaster University, Hamilton, Canada
14:45	**Computational modeling of the auditory periphery**	R. Meddis Department of Psychology, Essex University, UK
15:30	**Short presentations of posters and Demos**	
15:30	**Real-time demonstration of Neural Network based Speech Enhancement**	Tobias Goehring (1), Federico Bolner (2), and Stefan Bleeck (1) 1) University of Southampton, United Kingdom 2) Cochlear Technology Centre, Belgium
15:33	**A Noise Reduction Post-filter for Resolving Front-back Ambiguity in Single-Microphone Binaural Hearing Aids utilizing a Nearby External Microphone**	Dianna Yee (1), Homayoun Kamkar-Parsi (1), Rainer Martin (2), Henning Puder (1) 1) Sivantos GmbH 2) Ruhr Universität Bochum
15:36	**Adaptive Binaural Beamforming Based on Binaural Localization**	Mehdi Zohourian and Rainer Martin Institute of Communication Acoustics, Ruhr-Universität Bochum, Germany

15:39	**Implementation of a dereverberation algorithm for CI recipients on a realtime system**	Patricia Bleiker (1,2), Norbert Dillier (1), Andrea Kegel (1), Eleftheria Georganti (1,3), Dietmar Wohlbauer (1), Wai Kong Lai (1) 1) Laboratory of Experimental Audiology, ENT Department, University of Zurich, Switzerland 2) Department of Information Technology and Electrical Engineering, ETH Zurich, Switzerland 3) Sonova AG, Stäfa, Switzerland
15:42	**Speech Enhancement Based on Neural Networks Improves Speech Intelligibility in Noise for Cochlear Implant Users**	Tobias Goehring (1), Federico Bolner (2), Jessica Monaghan (1), Bas Van Dijk (2), Stefan Bleeck (1) 1) University of Southampton 2) Cochlear Technology Centre Belgium, KU Leuven
15:44	**Non-Intrusive Speech Intelligibility Prediction Using Hidden Markov Models**	Mahdie Karbasi and Dorothea Kolossa Cognitive Signal Processing Group, Ruhr-University Bochum, Germany
15:46	**Decision Device Comparison for Model-based Analysis of ITD Perception in Normal Hearing Listeners**	Arturo Moncada-Torres (1), Suyash N. Joshi (2), Bastian Epp (2), and Tom Francart (1) 1) ExpORL, Dept. of Neurosciences, KU Leuven, Herestraat 49, bus 721, 3000 Leuven, Belgium 2) Hearing Systems, Dept. of Electrical Engineering, Technical University of Denmark, Ørsteds Plads, building 352, DK-2800 Kgs. Lyngby, Denmark
15:48	**Functional modelling of interaural time difference discrimination in acoustical and electrical hearing**	Andreas Prokopiou, Arturo Moncada-Torres, Jan Wouters, Tom Francart ExpORL, Dept. Neurosciences, KU Leuven
15:50	**Modeling Speech Intelligibility based on Envelopes Derived from an Auditory Nerve Model**	Christoph Scheidiger, Johannes Zaar and Torsten Dau Hearing Systems group, Department of Electrical Engineering, Technical University of Denmark, DK-2800, Kgs. Lyngby, Denmark
15:52	**Speech perception and localisation with real-time SCORE in bimodal users**	Dimitar Spirrov (1), Bas van Dijk (2), Maaike Van Eeckhoutte (1), Tom Francart (1)1) ExpORL, Dept. Neurosciences, University of Leuven, Belgium2) Cochlear Technology Center, Mechelen, Belgium
15:54	**Towards a bio-inspired coding strategy for cochlear implants**	Sonia Tabibi (1,2), Andrea Kegel (2), Wai Kong Lai (2), Ian Bruce (3), Norbert Dillier (2) 1) Department of Information Technology and Electrical Engineering, ETH Zurich, Switzerland 2) Laboratory of Experimental

		Audiology, ENT Department, University Hospital and University of Zurich, Switzerland 3) Department of Electrical and Computer Engineering, McMaster University, Hamilton, Canada
15:56	**Spatial details in bilateral cochlear implants**	Dietmar Wohlbauer, WaiKong Lai, Norbert Dillier University & University Hospital Zurich
15:58	**Cross –correlation model of interaural time difference coding listeners with bilateral cochlear implants**	Suyash Narendra Joshi, Torsten Dau and Bastian Epp Hearing Systems group, Department of Electrical Engineering, Technical University of Denmark, Lyngby, Denmark
16:00	**Coffee break & Demos & Posters**	
17:00	**Machine listening for continuous improvement of speech intelligibility in hearing devices**	Bernd T. Meyer Center for Speech and Language Processing, Johns Hopkins University, Baltimore, USA
17:40	**Modelling of neural signal processing in the impaired auditory system**	Torsten Dau DTU
17:55	**Wrap-up ICanHear day 1**	Rainer Martin Ruhr-University Bochum
18:00	**Posters, Board meeting**	
18:30	**Apéro & Dinner (Restaurant Turm, UZH)**	

08:30	**Wednesday November 23rd 2016**	
08:30	**Models of hearing impairment and their consequences for signal processing algorithms**	Birger Kollmeier Medizinische Physik, Universität Oldenburg & Cluster of Excellence "Hearing4all"
09:10	**Innovative solutions for improved communication**	Dorothea Kolossa Ruhr-University Bochum
09:25	**Coffee break & Demos & Posters**	
10:25	**Evaluation and comparison of CI coding strategies regarding spectral and temporal cues**	Chris James (1,2), Mathieu Marx (2), Damir Kovacic (3) 1) Cochlear France SAS, Toulouse, France 2) Service ORL, CHU Toulouse, France 3) Department of Physics, University of Split, Croatia
11:05	**Signal processing algorithms for improved speech and music perception with cochlear implants**	Waldo Nogueira Medical University Hannover, Cluster of Excellence "Hearing4all", Hannover, Germany
11:45	**Evaluation - Consequences of hearing impairment and benefits from signal processing**	Jan Wouters KU Leuven
12:00	**Summary & Farewell**	Rainer Martin Ruhr-University Bochum
12:30	**Lunch (Mensa of University)**	

List of Posters (ARCHES)

Presentations	***Authors***	***Poster #***
Responses to sinusoidal frequency modulation in the guinea pig ventral cochlear nucleus	Nihaad Paraouty (1,2); Arkadiusz Stasiak (1); Christian Lorenzi (2); Ian M. Winter (1) 1) Physiological Laboratory, University of Cambridge, United Kingdom 2) Ecole Normale Superieure Paris LSP UMR 8248, PSL, France	AP-01
Comparing the effects of age and sensorineural hearing loss on detection and temporal integration of amplitude and frequency modulation	Nicolas Wallaert (1), Brian C. J. Moore (2), Stephan D. Ewert (3), Christian Lorenzi (1) 1) UMR CNRS LSP 8248, Institut d'Etude de la Cognition, Ecole normale supérieure, Paris Sciences et Lettres Research University, 29 rue d'Ulm, 75005 Paris, France 2) Department of Experimental Psychology, University of Cambridge,Downing street, Cambridge CB2 3EB, United Kingdom 3) Medizinische Physik and Cluster of Excellence Hearing4All, Universität Oldenburg, 26111 Oldenburg, Germany	AP-02
Influence of attention on speech-rhythm evoked potentials: first steps towards a speech driven brain-computer-interface	Carlos da Silva Souto, Helge Lüddemann, Silvia Lipski, Mathias Dietz and Birger Kollmeier. Medizinische Physik and Cluster of Excellence Hearing4all, Universität Oldenburg, D-26111 Oldenburg, Germany	AP-03
Common Audiological Functional Parameters (CAFPAs): an abstract representation of audiological expert data	Mareike Buhl, Marc René Schädler, Anna Warzybok, Birger KollmeierCvO Universität Oldenburg, Medical Physics and Cluster of Excellence Hearing4all (all authors)	AP-04
Frequency dependency of binaural masking level differences in NH and HI listeners	Christopher F. Hauth, Thomas Brand and Birger Kollmeier Medizinische Physik and Cluster of Excellence "Hearing4All", Carl von Ossietzky (CvO) Universität Oldenburg, 26129 Oldenburg, Germany	AP-05
Functional activity in the right auditory cortex reflects individual pitch-discrimination abilities in musicians	Federica Bianchi (1,2), Jens Hjortkjær (1,2), Sébastien Santurette (1,3), Hartwig R. Siebner (2,4), Robert J. Zatorre (5) and Torsten Dau (1) 1) Hearing Systems Group, Department of Electrical Engineering,Technical University of Denmark, Bygning 352,Ørsteds Plads, 2800 Kgs. Lyngby, Denmark 2) Danish Research Centre for Magnetic Resonance, Centre for Functional and Diagnostic Imaging and Research, Copenhagen University Hospital Hvidovre, 2650 Hvidovre, Denmark 3) Department of Otorhinolaryngology, Head and Neck Surgery & Audiology,	AP-06

	Rigshospitalet, 2100 Copenhagen, Denmark 4) Department of Neurology, Copenhagen University Hospital Bispebjerg, 2400 Copenhagen, Denmark 5) Montreal Neurological Institute, McGill University, and BRAMS, Montreal, Canada	
Evaluating fast-acting compression in basic psychoacoustic and speech tasks	Borys Kowalewski, Michal Fereczkowski, Ewen MacDonald, Olaf Strelcyk, Torsten DauHearing Systems group, Hearing Systems group, Hearing Systems group, Sonova U.S. Corporate Services, Hearing Systems group	AP-07
Objective assessment of voluntary stream segregation abilities of CI users	Andreu Paredes Gallardo, Sara Miay Kim Madsen, Torsten Dau and Jeremy Marozeau Hearing Systems group, Department of Electrical Engineering, Technical University of Denmark, DK-2800, Kgs. Lyngby, Denmark	AP-08
A correlation metric in the envelope power spectrum domain for speech intelligibility prediction	Helia Relaño Iborra, Tobias May, Johannes Zaar, Christoph Scheidiger, Torsten Dau Hearing Systems group, Department of Electrical Engineering, Technical University of Denmark, DK-2800, Kgs. Lyngby, Denmark	AP-09
Archetypal analysis of auditory profiling data towards a clinical test battery	Raul H. Sanchez(1), Federica Bianchi(1), Sébastien Santurette(1,2) and Torsten Dau(1) 1) Hearing Systems group, Department of Electrical Engineering, Technical University of Denmark, DK-2800, Kgs. Lyngby, Denmark 2) Department of Otorhinolaryngology, Head and Neck Surgery and Audiology, Rigshospitalet, Copenhagen, Denmark	AP-10
Improving rehabilitation of the hearing impaired based on profiling and COSI: results of a pilot with an improved protocol in the Netherlands.	Wim Soede (1), Bert van Zanten (2) and Wouter A. Dreschler (3) 1) Leiden University Medical Center, The Netherlands 2) Dept of Audiology, Utrecht University Medical Center, The Netherlands 3) Academic Medical Center, The Netherlands	AP-11
Critical factors in hearing aid selection and evaluation: audiological characteristics, self-report profiles, and hearing aid properties.	Wouter A. Dreschler (1), Inge de Ronde-Brons (1), Monique Boymans (1) and Wim Soede (2) 1) Academic Medical Center, The Netherlands 2) Leiden University Medical Center, The Netherlands	AP-12
Frequency-domain Reduced-rank Approximations of Music Signals for the Improvement of Music Perception in Cochlear Implant Listeners	Anil Nagathil (1), Claus Weihs (2), Katrin Neumann (3), Rainer Martin (1) 1) Institute of Communication Acoustics	AP-13

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The eyes and brain reveal your hearing ability	Adriana A. Zekveld (1,2), Dirk J. Heslenfeld (3), Niek J. Versfeld (1), Sophia E. Kramer (1) 1) Section Ear & Hearing, Dept. of Otolaryngology-Head and Neck Surgery and EMGO Institute for Health and Care Research, VU University medical center, De Boelelaan 1118, 1081 HZ, Amsterdam, The Netherlands 2) Department of Behavioural Sciences and Learning, Linnaeus Centre HEAD, The Swedish Institute for Disability Research, Linköping University, SE-581 83 Linköping, Sweden 3) Department of Psychology, VU University, Van der Boechorststraat 1, 1081 BT Amsterdam, The Netherlands	AP-14
Amplitude-modulation masking for frequency-modulation detection	Andrew J King, Christian Lorenzi Laboratoire des systemes perceptifs, CNRS, Ecole normale superieure, Paris Sciences et Lettres Research University, 29 rue d'Ulm, 75005 Paris, France	AP-15
Salicylate-induced changes in brain activity in awake guinea pigs	JI Berger, B Coomber, A Hockley, W Owen, MN Wallace and AR Palmer 1) Medical Research Council Institute of Hearing Research, Nottingham, NG7 2RD, UK 2) School of Medicine, University of Nottingham, Nottingham, NG7 2RD, UK	AP-16
Measuring tonotopic magnification in the human auditory cortex.	Ben Gurer, Julien Besle & Katrin Krumbholz Medical Research Council Institute of Hearing Research, The University of Nottingham, University Park, Nottingham, NG7 2RD	AP-17

List of Posters (ICanHear)

Real-time demonstration of Neural Network based Speech Enhancement	Tobias Goehring (1), Federico Bolner (2), and Stefan Bleeck (1) 1) University of Southampton, United Kingdom 2) Cochlear Technology Centre, Belgium	ID-01
A Noise Reduction Post-filter for Resolving Front-back Ambiguity in Single-Microphone Binaural Hearing Aids utilizing a Nearby External Microphone	Dianna Yee (1), Homayoun Kamkar-Parsi (1), Rainer Martin (2), Henning Puder (1) 1) Sivantos GmbH 2) Ruhr Universität Bochum	ID-02
Adaptive Binaural Beamforming Based on Binaural Localization	Mehdi Zohourian and Rainer Martin Institute of Communication Acoustics, Ruhr-Universität Bochum, Germany	ID-03
Implementation of a dereverberation algorithm for CI recipients on a realtime system	Patricia Bleiker (1,2), Norbert Dillier (1), Andrea Kegel (1), Eleftheria Georganti (1,3), Dietmar Wohlbauer (1), Wai Kong Lai (1) 1) Laboratory of Experimental Audiology, ENT Department, University of Zurich, Switzerland 2) Department of Information Technology and Electrical Engineering, ETH Zurich, Switzerland 3) Sonova AG, Stäfa, Switzerland	ID-04
Speech Enhancement Based on Neural Networks Improves Speech Intelligibility in Noise for Cochlear Implant Users	Tobias Goehring (1), Federico Bolner (2), Jessica Monaghan (1), Bas Van Dijk (2), Stefan Bleeck (1) 1) University of Southampton 2) Cochlear Technology Centre Belgium, KU Leuven	IP-01
Non-Intrusive Speech Intelligibility Prediction Using Hidden Markov Models	Mahdie Karbasi and Dorothea Kolossa Cognitive Signal Processing Group, Ruhr-University Bochum, Germany	IP-02
Decision Device Comparison for Model-based Analysis of ITD Perception in Normal Hearing Listeners	Arturo Moncada-Torres (1), Suyash N. Joshi (2), Bastian Epp (2), and Tom Francart (1) 1) ExpORL, Dept. of Neurosciences, KU Leuven, Herestraat 49, bus 721, 3000 Leuven, Belgium 2) Hearing Systems, Dept. of Electrical Engineering, Technical University of Denmark, Ørsteds Plads, building 352, DK-2800 Kgs. Lyngby, Denmark	IP-03
Functional modelling of interaural time difference discrimination in acoustical and electrical hearing	Andreas Prokopiou, Arturo Moncada-Torres, Jan Wouters, Tom Francart ExpORL, Dept. Neurosciences, KU Leuven	IP-04
Modeling Speech Intelligibility based on Envelopes Derived from an Auditory Nerve Model	Christoph Scheidiger, Johannes Zaar and Torsten Dau Hearing Systems group, Department of Electrical Engineering, Technical University of Denmark, DK-2800, Kgs. Lyngby, Denmark	IP-05
Speech perception and localisation with real-time SCORE in bimodal users	Dimitar Spirrov (1), Bas van Dijk (2), Maaike Van Eeckhoutte (1), Tom Francart (1)	IP-06

	1) ExpORL, Dept. Neurosciences, University of Leuven, Belgium 2) Cochlear Technology Center, Mechelen, Belgium	
Towards a bio-inspired coding strategy for cochlear implants	Sonia Tabibi (1,2), Andrea Kegel (2), Wai Kong Lai (2), Ian Bruce (3), Norbert Dillier (2) 1) Department of Information Technology and Electrical Engineering, ETH Zurich, Switzerland 2) Laboratory of Experimental Audiology, ENT Department, University Hospital and University of Zurich, Switzerland 3) Department of Electrical and Computer Engineering, McMaster University, Hamilton, Canada	IP-07
Spatial details in bilateral cochlear implants	Dietmar Wohlbauer, WaiKong Lai, Norbert Dillier University & University Hospital Zurich	IP-08
Cross –correlation model of interaural time difference coding listeners with bilateral cochlear implants	Suyash Narendra Joshi, Torsten Dau and Bastian Epp Hearing Systems group, Department of Electrical Engineering, Technical University of Denmark, Lyngby, Denmark	IP-09

ABSTRACTS

Short talk

Panoramic talk about projects at DTU

Torsten Dau
DTU

No Abstract

Noise-robust neural tracking of attended talkers in real-world acoustic scenarios

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Using selective auditory attention, normal hearing listeners can extract intelligible speech from complex multi-talker scenarios. In recent studies, reconstruction of speech envelopes from cortical responses has indicated enhanced neural representations of an attended speech stream in conditions with two competing talkers. This has been utilized to determine which of two talkers a listener is attending to from single-trial EEG responses. Here we used envelope reconstruction techniques to investigate the neural representation of attended speech in acoustically complex real-world scenes. We used loudspeaker arrays and room modeling to create virtual acoustic environments with varying degrees of reverberation and varying a number of competing talkers. Analyzing single-trial EEG, we found that decoding of attended speech streams was robust to different envelope distortions in the different acoustic scenarios. In highly reverberant environments, we found that the signal reconstructed from the cortical response represented the clean envelope more strongly than the distorted acoustic input. Our results suggest that selective auditory attention promotes the formation of noise-invariant cortical representations of speech.

The effect of dynamic range compression on spatial perception in a reverberant environment

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The present study investigated the effects of fast-acting hearing-aid compression on normal-hearing and hearing-impaired listeners' spatial perception in a reverberant environment. Three compression schemes - independent compression at each ear; linked compression between the two ears; and "spatially ideal" compression operating solely on the dry source signal - were considered using virtualized speech and noise bursts. Listeners indicated the location and distribution of their perceived sound images on the horizontal plane graphically on a touch screen. A linear amplification scheme was considered as the reference condition. The results showed that both independent and linked compression resulted in more diffuse and broader sound images as well as internalization and image splits, whereby more image splits were reported for the noise bursts than for speech. Only the spatial ideal compression provided the listeners with a spatial percept similar to that obtained with linear processing. The same general pattern was observed for both listener groups. An analysis of the interaural cross-correlation and direct-to-reverberant ratio suggested that the spatial distortions resulted from enlarged reverberant energy. Thus, modifictions of the relation between the direct and the reverberant part of the sound should be avoided in amplification strategies that attempt to preserve the natural sound scene around a listener while providing sufficient dynamic range compression to restore proper loudness cues.

Aided Patient Performance Predictions with FADE

Anna Warzybok, Marc René Schädler, Birger Kollmeier
Universität Oldenburg

Predicting the outcome of different acoustic experiments often requires the use of task-specific models.
The so-called Framework for Auditory Discrimination Experiments (FADE) uses a generalizing, automatic-speech-recognition-based (ASR) approach enabling outcome predictions of different acoustic experiments including speech recognition and classical psychoacoustic experiments.
To simulate the speech recognition process, ASR models are trained and tested on a broad range of signal-to-noise ratios on noisy (unprocessed or processed) speech recordings.
From the simulated recognition scores, the lowest achievable speech reception threshold is reported as the predicted performance. For hearing-impaired listeners, the simulations are individualized by incorporating the individual absolute hearing threshold into the front-end of the recognition system.
An overview about studies done so far will be given including validation of the model for speech intelligibility predictions in different languages and noise conditions, in combination of reverberation and noise as well as for aided conditions. The aided conditions will consider evaluation of different dereverberation algorithms in normal-hearing listeners and evaluation on different noise reduction schemes in normal-hearing and hearing-impaired listeners in realistic noise scenarios.

Sharpening of Cortical Frequency Selectivity via Frequency Specific Adaptation

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Background
Adaptation, the reduction in neural responses to repeated stimuli, is a ubiquitous phenomenon throughout sensory systems and is an important aspect of neural coding. Here we test the hypothesis that prolonged adaption increases the frequency selectivity of cortical responses in human EEG recordings, and present a neural model to explain the responses.
Method
Pure tone adapter-probe sequences were presented in which a single probe tone of one frequency was preceded by adapters of a second frequency (frequency difference: 0-, 600-, 1800- cents). Adapters consisted of either a train of 100ms tones (25ms gap), varying in number (1, 2, 3, 6, 9, 15), or a single adapter which varied in duration (100, 225, 350, 725, 1100, 1850-ms). Adaptation was characterized as a reduction of the probe response relative to the response to the probe alone.
Results
In all cases the response to the probe was most strongly reduced when the adapter frequency was closest to the probe frequency. However, the degree of adaptation depended on the temporal arrangement of adapters. At all frequency differences, adaptation of the probe initially grew with increasing numbers of adapters, or adapter duration. At long adapter durations (>400ms), regardless of frequency difference, adaptation was reduced relative to peak values. Adaptation also reduced with >3 repeated adapters, but only when there was a difference in frequency between the adapter and probe. Effectively, the tuning of adaptation became sharper with increasing numbers of adapters, but not with adapter duration.
EEG adaptation tuning was explained by an extension to a model proposed to explain frequency specific adaptation in single neurons (Mill et al. 2011). The model was a two-layer network with convergent inputs, independently adapting synapses and separate non-adapting inputs for onset responses. This model is able to quantitatively reproduce the observed non-monotonic adaptation and sharpening of tuning observed in our EEG responses, and the effects of repeated and prolonged adapters.
Conclusion
The results suggest that adaptation from repeated, but not prolonged stimulation leads to a sharpening of the frequency tuning in auditory cortex, which may serve to emphasize the representation of stimuli that are different in frequency to those that precede it. This may be a natural consequence of a hierarchical network of neurons with independently adapting inputs.

Mill R, Coath M, Wennekers T, Denham SL (2011) A neurocomputational model of stimulus-specific adaptation to oddball and Markov sequences. PLoS Comput Biol 7.

Audiological research at Ecole normale superieure, Paris, France

Christian Lorenzi
Ecole normale superieure & CNRS

Our team is conducting research on the perceptual consequences of aging and cochlear damage.

This team (http://www.iec-lsp.ens.fr/) is part of the Laboratoire des systemes perceptifs (LSP, IEC, ENS), a CNRS research unit of the Departement d'Etudes Cognitives (DEC, IEC) at Ecole normale superieure, in Paris, France.

We combine psychophysical, neurophysiological and modelling methods to investigate the perception of temporal modulations in sounds by the normal and impaired auditory system.

We currently conduct cross-linguistic studies to characterize temporal-modulation information (amplitude modulation, AM, and frequency modulation, FM) in speech.

We explore the neural coding of AM and FM information at the early stages of the auditory system and the central effects of aging and cochlear damage on AM and FM processing.This is achieved by recording neural responses to AM and FM in the low brainstem and auditory cortex of guinea pigs (collaboration: Univ Cambridge, UK; Univ Orsay, France), and by investigating interference and integration effects for AM and FM detection in normal-hearing and hearing-impaired humans (collaboration: Univ Cambridge, UK).

We also investigate the auditory development of modulation perception during infancy and childhood (collaboration: UCL, UK). A computational model of AM and FM processing using temporal-envelope and temporal fine-structure information is under development (collaboration: Hearing4all, Germany, Oldenburg).

Responses to sinusoidal frequency modulation in the guinea pig ventral cochlear nucleus

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Many psychophysical studies have investigated the detection of sinusoidal frequency-modulation (SFM) and on the type of sensory information it predominantly relies on: temporal-envelope (ENV) resulting from cochlear filtering, or temporal fine-structure (TFS) cues conveyed by changes in the neural phase-locking pattern over time, or a combination of both. Few neurophysiological studies have addressed this issue and data is still lacking regarding the coding of low-rate SFM in the early stages of the auditory pathway. This work aimed at characterizing the responses of ventral cochlear nucleus (VCN) neurons to low-rates SFM tones presented at various modulation depths and sound levels.

Single-units in normal-hearing anaesthetized pigmented guinea pigs were recorded extracellularly using tungsten-in-glass microelectrodes. Stimuli were 1-second SFM tones played at the unit's best frequency (BF), at modulation rates of 2, 5 and 10 Hz and modulation depths of 2, 4, 8, 16, and 32% relative to BF.

VCN responses to SFM varied as a function of sound level, bandwidth and unit type. Shuffled correlogram analyses were carried out in order to assess the relative strengths of ENV and TFS coding for the different unit types in the VCN. For small modulation rates (≤5Hz) and small modulation depths (≤16%), low-CF units showed weak temporal ENV coding but high phase-locking to TFS. In comparison, high-CF units generally followed the stimulus ENV for most conditions. The transition region over which temporal coding changes from being dominated by TFS coding to ENV coding was around 1-2 kHz.

The results provide physiological evidence that SFM is encoded via neural phase-locking to TFS cues for low carrier frequencies and modulation rates, and via neural phase-locking to ENV cues at high carrier frequencies. The results also suggest weaker phase-locking to TFS in guinea pigs compared to other species. Further work will be carried out in order to assess the strengths of these two coding mechanisms in the presence of a hearing deficit.

Acknowledgements
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Comparing the effects of age and sensorineural hearing loss on detection and temporal integration of amplitude and frequency modulation

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Background

The relative contribution of excitation-pattern and temporal fine-structure (TFS) information to modulation sensitivity is still a matter of debate. We addressed this issue by investigating the effects of age and sensorineural hearing loss on amplitude-modulation (AM) and frequency-modulation (FM) detection thresholds (AMDTs and FMDTs, respectively) and temporal integration for AM and FM detection.

Methods

AMDTs and FMDTs were measured at 40 dB SL for young and older normal-hearing listeners (NHy and NHo, respectively) and for older listeners with sensorineural hearing loss (HIo) using a 500-Hz sinusoidal carrier and modulation rates of 2 and 20 Hz. The number of modulation cycles, N, varied between 2 and 9. We also assessed whether a computational model of temporal-envelope processing based on the modulation-filterbank concept and a template-matching decision strategy could account for the data.

Results

Mean results for each group are shown in the figure. For all groups, AMDTs and FMDTs were always lower (better) for the 2-Hz rate than for the 20-Hz rate and decreased (improved) with increasing N. AMDTs were higher for the NHo than for the NHy group, and higher for the NHo group than for the HIo group. The effect of increasing N was similar for the NHy and NHo groups, but was greater for the HIo group. FMDTs were higher for the NHo group than for the NHy group for the 2-Hz rate only. FMDTs were higher for the HIo group than for the NHo group for both rates. The effect of increasing N was similar across the three groups. The computational model accounted relatively well for the AM data, but failed to account for the FM data, probably the model did not take account of the use of TFS information.

Conclusions

Taken together, the data show that the effects of age and hearing loss are different for AM and FM detection and for temporal integration in AM and FM detection, consistent with the notion that AM and FM detection rely on different information. In addition, the data suggest that:

i) Greater age reduces sensitivity to both excitation-pattern and TFS cues, but more so for the latter.

ii) Loss of amplitude compression in the impaired cochlea is responsible for the enhanced sensitivity and temporal integration of excitation-pattern cues found for hearing-impaired listeners.
iii) Ageing and hearing loss spare memory and decision factors responsible for temporal integration of excitation-pattern and TFS cues.

Influence of attention on speech-rhythm evoked potentials: first steps towards a speech driven brain-computer-interface.

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A Brain-computer interface (BCI) uses neuronal responses to control external systems. The majority of BCI systems are based on visual stimuli, few apply auditory input. Because auditory-based BCIs do not rely on visual skills or mobility of the body, they could be an alternative for visually or physically disabled people.

This study investigates the performance of an auditory paradigm using two competing streams of repeatedly presented speech syllables. The streams had different repetition rates of 2.3 and 3.1 Hz. Our auditory BCI approach uses the auditory steady-state response (ASSR) to automatically detect which stream a listener selectively attends to.
In a single trial classification ten healthy volunteers achieved a significant above chance accuracy of 61 % and an information transfer-rate (ITR) of 0.2 bit/min. Using the average over six random trails improved the average classification accuracy to 79 % by keeping the ITR comparable.

In conclusion it is possible to classify ASSR evoked from streams of spoken syllables. The performance of this auditory BCI is not yet effective enough for a real life application, but is a step towards the long term goal of using BCIs on natural speech features and eventually controlling the processing of hearing devices.

Common Audiological Functional Parameters (CAFPAs): an abstract representation of audiological expert data

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Correct diagnosis of hearing impairment and indication for individual treatment is a complex task influenced by many factors like the choice of audiological measurements, the most frequent cases at a clinic's location or the experience of the ENT doctor. The goal of the Common Audiological Functional Parameters (CAFPAs) approach is to build up a supporting tool uniting expert knowledge and providing it to the community of ENT doctors.

The CAFPAs are designed as an abstract representation of audiological measurements, thereby generalizing over different measurements and representing functional principles of the auditory system. First data for CAFPAs and measurement results, for different, predefined diagnostical cases and indications for treatment, was collected by the use of a survey among experts from Hanover and Oldenburg. With this data, it is possible to distinguish different diagnostical cases on the basis of audiological measurements as well as on the basis of the newly-introduced CAFPAs. Thus, the CAFPAs serve as an abstract and compact, but interpretable representation of the measurements that do not take into account every detail.

Furthermore, the survey results show that the degree of agreement of the experts on the measurement and CAFPA outcomes differed for different diagnostical cases.

This property supports the idea of using statistical methods, e.g. machine learning methods, for modeling the audiological diagnostical path.

Frequency dependency of binaural masking level differences in NH and HI listeners

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In binaural tone-in-noise experiments humans are able to achieve substantially lower thresholds if either the noise or the tone have interaural time/phase differences (ITDs/ IPDs) or level differences (ILDs) compared to a diotic presentation of tone and noise (same signals presented to both ears). This effect is named binaural masking level difference (BMLD) and was mainly investigated at 500 Hz for different interaural configurations of tone and noise, where a maximum BMLD of 15 dB can be achieved (Langford and Jeffress, 1964; Egan, 1965). Moreover, the results obtained in these experiments were used to fit binaural processing errors in the equalization cancellation (EC) mechanism (Durlach et al., 1963), which is an effective model of binaural processing. In this model, first, interaural differences in time and level between left and right ear signals are compensated (equalization step). Afterwards, the left ear signal is subtracted from the right ear signal (cancellation step). This leads to an improvement in the signal-to-noise ratio (SNR) by constructive and destructive interferences if target and interferer have differing ITDs or ILDs.

This EC mechanism is also applied in binaural speech intelligibility models, where it is combined with the speech intelligibility index (SII,ANSI 1997) (for example, Beutelmann et al 2010; Lavandier and Culling, 2010). In this study, the frequency dependency of BMLDs is investigated for listeners with normal hearing and listeners with high frequency hearing loss. Binaural tone in noise experiments are conducted for tone frequencies of 250, 500, 750, 1000, 1500, and 2000 Hz in a broadband noise ranging from 100-4000 Hz. In order to investigate the binaural release from masking, the ITD of the noise is varied depending on the tested tone frequency. The ITD is varied in steps of half of the period of the target tone from 0 (diotic presentation) to 5 times the period. In case of a 500 Hz tone, this means that the ITD of the noise is varied from 0 to 5 ms in steps of 0.5 ms. Furthermore, it is investigated whether the EC mechanism with processing errors derived at 500Hz can be used to predict the BMLD for the remaining frequencies and to which extent the data collected in this experiment can be used to individualize binaural processing errors in the EC model and, therefore, also to individualize binaural speech intelligibility predictions besides the audiogram.

Functional activity in the right auditory cortex reflects individual pitch-discrimination abilities in musicians

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Musicians have highly enhanced pitch-discrimination abilities compared to non-musicians. It is unclear whether this training-dependent ability can be ascribed to an enhanced neural representation of pitch at early or later stages of the auditory system. To address this question, we performed two experiments in 15 musicians and 15 non-musicians.

First, we estimated pitch-discrimination thresholds for harmonic complex tones with either spectrally resolved or unresolved harmonics in the auditory system. Musicians outperformed non-musicians, showing lower pitch-discrimination thresholds for all conditions. The behavioral results suggest an enhancement in musicians that was not limited to spectral resolvability and is likely to occur beyond peripheral stages of the auditory system.

The same participants underwent a task-related functional MRI at 3 tesla while they performed a similar pitch-discrimination task. To account for the between-group differences in pitch-discrimination, task difficulty was adjusted to the individual pitch-discrimination ability. Relative to non-musicians, musicians showed increased neural responses to complex tones with either resolved or unresolved harmonics in the right Heschl's gyrus, right inferior frontal gyrus, but also at the level of the inferior colliculus. Additionally, neural responses in the right auditory cortex were predictive of the individual pitch-discrimination performance in musicians but not in non-musicians, consistent with a hemispheric specialization of the right auditory cortex for processing fine spectral changes. These findings suggest a more robust neural representation of pitch in musicians both at the collicular and cortical level. However, only task-related activation of the right auditory cortex reflected individual differences in pitch-discrimination abilities in musicians.

Evaluating fast-acting compression in basic psychoacoustic and speech tasks

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Multi-channel wide dynamic-range compression (WDRC) is involved in the signal processing schemes of most modern hearing aids. Commonly used nonlinear-gain prescriptions (e.g., DSL v5.0, NAL) do not make specific recommendations for other parameters of compression amplification, such as time constants or the number of processing channels.

Currently, there is no consensus regarding optimal values of these parameters. Studies using speech recognition as an outcome measure have shown mixed results, partly due to the differences in testing conditions (e.g. overall input levels and signal-to-noise ratios), differences in individual cognitive abilities, and differences in available cues for speech recognition. The latter likely depend on the degree and configuration of the hearing loss.

Here, a model of auditory processing (Jepsen et al. 2011) was used to compare linear and various compression schemes in terms of individual listener's performance in basic psychoacoustic tasks. Specifically, time constants and the number of channels were chosen to provide the best possible improvement of the simulated aided performance of hearing-impaired listeners in tasks assessing spectral and temporal resolution. The simulation outcomes indicated that compression with a sufficiently large number of channels and short time constants can provide a benefit over linear amplification. These results were verified in aided psychoacoustic experiments.

Further, speech recognition performance of the same group of hearing-impaired listeners with compressive and linear amplification was investigated. Speech reception thresholds (SRTs) were measured using two different speech corpora and maskers. Dantale II (Wagener et al., 2003) sentences were presented with ICRA7 fluctuating noise and resulting SRTs were expected to be negative. Using the DAT corpus (Nielsen et al., 2014), the target talker was presented with two competing talkers and positive SRTs were expected in this condition. It was hypothesized that compressive processing may be more beneficial at negative signal-to-noise ratios (SNRs), hence the relative benefit of compressive over linear processing should be larger for the Dantale corpus.

Even though no significant interactions were observed, a trend was seen in the data suggesting that the relative benefits of compression and linear amplification might depend on the SNR. Studying the effect in more subjects and across a broader range of SNRs might help resolve these issues.

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Objective assessment of voluntary stream segregation abilities of CI users

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Cochlear implants (CI) significantly improve the ability to understand speech, allowing most CI recipients to achieve high levels of speech intelligibility in quiet situations. However, listening to music or a single voice in a crowded room is still challenging for most CI users. Both music perception and speech-in-noise understanding involve auditory streaming, a perceptual process by which the human auditory system organizes sounds from different sources into perceptually meaningful elements.

Despite its high relevance in many daily situations, the number of studies investigating segregation abilities of CI listeners is limited and their findings are contradictory. Most of the previous research assessed obligatory or primitive stream segregation, a bottom-up process driven exclusively by the acoustic characteristics of the stimuli. Little attention has been given to voluntary stream segregation, a top-down process where the listener actively tries to segregate the sounds and where attention influences perception. The present study objectively assesses voluntary stream segregation abilities of CI users as a function of electrode separation and aims to establish whether a two-stream percept can occur instantaneously or whether this needs time to be built up.

CI users participated in an objective rhythm detection task composed of a sequence of regularly presented bursts of pulses on a single electrode (A) interleaved with an irregular sequence presented on a different electrode (B). On half of the trials a small temporal delay was added to the last burst of the regular A sequence, and subjects were asked to indicate whether they could detect this delay. The period between consecutive bursts of the B sequence was jittered, making time judgments between the A and B sequences an unreliable cue to perform the task. Thus, segregation of the A and B sequences would improve performance. The electrode separation and the duration of the sequences were varied between trials. Results show that performance increases with electrode separation and sequence length, suggesting that CI users experience stream segregation as a function of electrode separation and that this percept builds-up over time.

A correlation metric in the envelope power spectrum domain for speech intelligibility prediction

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A speech intelligibility model, named sEPSMcorr, is presented, which uses a modulation-frequency selective processing based on the (multi-resolution) speech-based envelope power spectrum model (mr-sEPSM; Jørgensen et al. 2013) in combination with a cross-correlation based back end inspired by the short-time objective intelligibility measure (STOI; Taal et al., 2011). The model can accurately predict data obtained with normal-hearing (NH) listeners for a broad range of listening conditions, including effects of stationary and fluctuating additive interferers as well as effects of non-linear distortions, such as spectral subtraction, phase jitter and ideal binary mask (IBM) processing. The model has a larger predictive power than both the original mr-sEPSM (which fails in the phase-jitter and IBM conditions) and STOI (which fails to predict the influence of fluctuating interferers).

However the sEPSMcorr preprocessing does not provide a flexible framework to predict individual speech intelligibility data from hearing impaired listeners. Thus, the back end of the sEPSMcorr was combined with a more realistic auditory pre-processing front end adopted from the computational auditory signal processing and perception model (CASP; Jepsen et al., 2008). The preprocessing contains outer- and middle-ear filtering and a non-linear auditory filterbank (DRNL, López-Poveda and Meddis, 2001), followed by inner hair-cell transduction, adaptation and a modulation filterbank.

The predictions of the sEPSM-based and the CASP-based models were compared with respect to measured data (NH) in conditions of additive masking noise, phase jitter distortions, reverberation and noise-reduction algorithms. The effects of the back end as well as the different preprocessing stages on the predicted results were analyzed. The resulting modelling framework could be useful for the design and evaluation of, e.g. speech transmission algorithms or hearing-instrument algorithms.

Archetypal analysis of auditory profiling data towards a clinical test battery

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Nowadays, the pure-tone audiogram is the main tool used to characterize the degree of hearing loss and for hearing-aid fitting. However, the perceptual consequences of hearing loss are typically associated not only to a loss of sensitivity, but also to a loss of clarity (distortions) that is not captured by the audiogram. Here, we hypothesize that any listener's hearing can be characterized along two dimensions: audibility-related and non-audibility-related distortions. In this space, four profiles can be identified: normal-hearing, sensitivity loss, hearing loss with clarity loss and normal-hearing with clarity loss (hidden hearing loss). Recently, Thorup et al. (2016) proposed an extended auditory profile beyond the audiogram for hearing aid candidates. A new analysis of these data using archetypal analysis is presented here to evaluate our hypothesis. This technique uses unsupervised learning for identifying extreme patterns in the data, which would correspond to different profiles. Results provided consistent evidence of the existence of different "Auditory Profiles" in the data. The most sensitive tests for the classification of the hearing-impaired listeners were related to temporal processing, loudness, cognition, and speech perception. The current approach seems promising for analyzing other existing data sets in order to select the most relevant tests for auditory profiling.

Overview of Auditory Research in Salamanca

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University of Salamanca, Salamanca, Spain

I will provide an overview of the ongoing research projects at the Auditory Computation and Psychoacoustics Laboratory of the Neuroscience Institute of Castile and Leon, University of Salamanca, Spain.

On the Value of Brief Sound Audiometry as a Diagnostic Tool for Cochlear Synaptopathy

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Cochlear synaptopathy (or the loss of primary auditory synapses) is thought to be caused by noise exposure and/or aging and remains a subclinical condition. Based on a model, it has been suggested that synaptopathy leads to a relatively larger increase in detection thresholds for short than for long duration sounds [Marmel et al., 2015, Front. Aging Neurosci. 7:63]. Conversely, cochlear mechanical loss would tend to lead to larger threshold increases for long versus short duration sounds.

The present study investigates the use of brief-sound audiometry for the diagnosis of synaptopathy. On the assumption that for listeners with normal audiograms synaptopathy is correlated with age and noise exposure, we hypothesized that the threshold difference between short and long durations sounds increases with increasing age and noise exposure. To test this hypothesis, absolute thresholds for stimulus durations from 2 to 500 ms were measured for 24 listeners with normal audiograms (thresholds < 15 dB HL) and ages from 24 to 68 years. Stimuli included wideband noises (0.1-10 kHz) and pure tones with frequencies of 0.5, 1.5, 4.0, 8.0, and 12.0 kHz. Self-reported life-span exposure to loud sounds was assessed using a questionnaire. As expected, thresholds were higher for the shorter than for the longer stimuli. In contrast to the hypothesis, however, the threshold difference between the short and the long duration sounds decreased with increasing age and noise exposure.

The threshold difference also decreased with increasing the long-sound thresholds. Altogether, the pattern of results was more consistent with the effects of cochlear mechanical loss on thresholds than with synaptopathy, something surprising given that participants had normal audiograms. We conclude that either the modeled effects of synaptopathy on threshold-duration functions are nonexistent or washed out by the effects of subclinical (< 15 dB HL) mechanical losses, or that synaptopathy is less frequent than is thought to be.

[Work supported by the Oticon Foundation and CNPQ Brasil]

Medial Olivocochlear Reflex Effects on Amplitude Modulation Detection

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Medial olivocochlear (MOC) efferents may be activated in a reflexive manner by ipsilateral and contralateral sounds. Their activation linearizes basilar membrane input/output curves, an effect that could enhance the sound envelope as represented in the mechanical response of the cochlea, and thus facilitate the detection of amplitude modulation.

To investigate this possibility, amplitude modulation (AM) detection thresholds were measured monaurally in the presence and in the absence of a 60 dB SPL, 400-ms, wideband (0.1-10 kHz) noise. It was assumed that this noise activated the MOC reflex without activating the middle-ear muscle reflex. The probe was a 70 dB SPL, 1500 Hz tone modulated in amplitude at a rate of 40 Hz. MOC reflexes have an onset delay of about 25 ms and need about 300 ms to be completely activated. A short (50 ms) probe was used for the probe to minimally activate the MOC reflex by itself and AM thresholds were measured for the tone presented at the noise onset (early condition) and 300 ms after the noise onset (late condition). The difference in AM threshold for the late and early conditions was regarded as indicative of the MOC reflex effect on AM detection. To assess the effects of ipsilateral, contralateral, and bilateral MOC reflex elicitors on AM detection, AM thresholds were measured for ipsilateral, contralateral, and diotic noises, respectively, as well as in quiet. AM thresholds were worse in noise than in quiet. In noise, however, thresholds were better for the late than for the early condition, and the difference was comparable for ipsilateral, contralateral, and diotic noises. The results suggest that the MOC reflex facilitates the detection of amplitude modulation in noise, and that the magnitude of the benefit is comparable for ipsilateral, contralateral, and bilateral elicitors.

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Synaptopathy with envelope following responses (EFRs): The off-frequency problem

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2) Eaton-Peabody Labs. Massachussets Eye and Ear Infirmary. Harvard Medical School
3) Interacoustics Research Unit
4) Boston University

The ability to communicate in challenging situations with high levels of background noise is a fascinating property of the healthy auditory system. Despite normal sensitivity to pure-tones, many listeners complain about having difficulties in such situations. Recent animal studies have shown that noise over-exposure that produces temporary threshold shifts can cause the loss of auditory nerve (AN) fiber synapses. This neuronal degeneration has been termed "hidden hearing loss" or, more accurately, "synaptopathy", since it is not reflected in the traditional pure-tone threshold. The envelope following response (EFR) has been proposed as a potential objective method to assess synaptopathy in humans. In this study, an AN computational model was used to investigate the effects of off-frequency contributions (i.e. away from the characteristic place of the stimulus) and the differential loss of different AN fiber types on EFR level-growth functions. EFR results in human where lastly compared to EFR results in noise exposed (synaptopathic) and non-exposed mice.

EFRs in humans were measured using a 64-channel EEG system with active electrodes. EFR level-growth functions were recorded in young human listeners for SAM tones with a carrier frequency of 2 kHz and modulation frequency of 93 Hz. Modulations depths of 25% and 85% were tested at sound pressure levels (SPL) of 40 to 90 dB. Subjects were audiometrically normally hearing (NH). A humanized AN model (Zilany et al., 2014) was used to simulate the EFR level-growth functions at the tested modulation depths and levels for several degrees of synapatopathy.

A similar experiment was conducted in mice. EFR level-growth functions were recorded in mice for SAM tones with a carrier frequency of 12 kHz and 30 kHz and modulation frequency of 1024 Hz. Modulations depths of 25% and 85% were tested at sound pressure levels (SPL) of 40 to 80 dB.

The AN model can account for the general trends obtained from human EFR level-growth functions. The simulations reveal that on- vs off-frequency EFR level-growth functions show completely different shapes. Off-frequency contributions have a large impact in the overall EFR magnitudes, but particularly at higher stimulus intensities. The total EFR responses are strongly dominated by the high-SR fibers, especially at low and mid intensities.

EFR level-growth functions in mice show different trends in noise exposed versus non-exposed animals at synaptopathic frequencies.

Off-frequency contributions of high-SR fibers dominate the total model EFR responses, suggesting that the loss of low- and medium-SR fibers has only little impact on measured

EFRs. EFR level-growth functions in some NH human listeners show a similar trend to the one obtained at the synaptopatic frequency in the noise exposed mice group.

Funding
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Panoramic talk on projects at Laboratory of Experimental Audiology, UZH/USZ

Norbert Dillier
Laboratory for Experimental Audiology, University & University Hospital Zurich

The Laboratory of Experimental Audiology (LEA) is a research group within the Department of Otolaryngology, Head and Neck Surgery of the University Hospital Zurich. Since many years, fruitful collaborations have been established with institutes of the University and ETH Zurich and research groups at European Universities as well as with industrial partners.

The research goals are directed towards investigations of the functional properties of the auditory system, the application of digital signal processing technology for hearing instruments and cochlear implants and performance evaluation of hearing instruments and implantable hearing devices in realistic acoustic environments.

The following projects are currently under way:

- Coding strategies for CI users with better sound quality, especially for music
- Objective measures for improved programming and speech processor fitting and for coding strategy optimization
- Real life acoustic environments with noise and reverberation for evaluation of signal processing algorithms and systems
- Binaural algorithms for bilateral electrical or combined electrical acoustical stimulation to improve localization and speech recognition in noise
- Age related hearing loss, cognition and speech recognition

Low speech intelligibility in older adults - The aging ear or brain?

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2) Neuroplasticity and Learning in the Healthy Aging Brain, University of Zurich

Introduction:
Previous studies tried to investigate why older adults even in the absence of hearing impairment (per definition) typically experience increased difficulties understanding speech in noise. This study combines several perspectives on speech processing of young and elderly adults: an audiological, a neuroanatomical and a neurofunctional perspective. The current work focuses on the audiological aspects of speech processing.

Methods:
Auditory profiles were measured with MATLAB Auditory Periphery (MAP)-Software (Meddis) in young and elderly normal hearing adults. Absolute thresholds, measures for frequency selectivity and temporal masking were included in the measured profiles. Furthermore, a sentence recognition test in noise was performed.

Results:
Elderly participants showed increased absolute detection thresholds in comparison to the young listeners. Differences in frequency selectivity occurred between the two age groups. They were frequency dependent and only partly significant. Results for temporal masking were less meaningful because of high interindividual variance. Sentence recognition in noise was considerably worse for the elderly in comparison with young participants and could be identified as the variable which explained most of the variance for age differences.

Conclusions:
Elderly normal hearing listeners showed significant frequency-dependent differences in measures of threshold and frequency selectivity in comparison with younger participants.

Neuroanatomical and intrinsic cortical oscillatory correlates of central hearing loss in older adults

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Twenty-three older and thirteen younger adults were tested with an adaptive auditory test battery to measure not only traditional pure-tone thresholds, but also above-threshold temporal and spectral processing. Participants' speech recognition in noise (SiN) was also evaluated, and T1-weighted MRI images were obtained to determine cortical thickness (CT) and surface area (CSA) of auditory and higher speech-relevant regions. Further, we obtained resting state EEG to infer on the speech-relevant intrinsic theta and gamma power lateralization.

Behavioral results indicated that older adults performed worse in all tasks, despite normal peripheral hearing. These age-related distinctions were accompanied by lower CT in all regions. Age modulated the regressions in right auditory areas, where a thicker cortex was associated with better auditory performance in older adults. Moreover, a thicker right supratemporal sulcus predicted more rightward theta lateralization, indicating the functional relevance of the right auditory areas in older adults.

The question how age-related cortical thinning and intrinsic oscillatory architecture relates to central hearing loss has so far not been addressed. Here, we provide the first neuroanatomical and neurofunctional evidence that cortical thinning and lateralization of speech-relevant oscillatory organization relates to the extent of age-related central hearing loss in older adults.

Improving rehabilitation of the hearing impaired based on profiling and COSI: results of a pilot with an improved protocol in the Netherlands.

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Research questions:
In the Netherlands in 2013 a protocol started to make the selection process of a hearing aid more transparent. The protocol was based on a structured and partly validated framework to assess the individual needs for compensation based on the individual disabilities and individual fitting goals (Dreschler, IHCON 2014), including the selection of a hearing aid with an adequate compensation power.

The system in its origin was designed as a flexible, self-learning system, but In daily practice it turned out to be a tight inflexible system due to rigid rulings and pricing mainly set by contracts between dispensers and assurance companies. In 2015 it was concluded that a restart of the system with more attention for the personal needs of the hearing impaired was necessary, including attention for assistive listening devices.

Methods:
An improved protocol was designed to do human related profiling based on a basic questionnaire, addition of COSI and extra attention for the influences of disability on listening effort as well as work and social functioning. The protocol resulted in the need for compensation along six scales: detection, speech in quiet, speech in noise, localization, focus/ discrimination, and noise tolerance. Next to this the protocol results in an advice to the hearing aid dispenser for a hearing aid category (5 levels, simple to advanced). The dispenser could follow the pro-posed level or change the level based on his/her professional judgement. From September 2015 until April 2016 a large pilot study of the proto-col has been carried out with the help of 50 dispensers. A total of 1250 dossiers with answers of questionnaires, COSI, audiograms, selected levels of hearing aids and speech scores with/without hearing aids were gathered.

Results:
First results show that the new protocol is well accepted as a major improvement compared to the previous stricter protocol. Although the pilot is now based on a lot of questionnaires and paperwork, the improved protocol was judged positively by dispensers and clients. The special attention for the effect of the hearing problem on listening effort and influence on social situations and work was judged as beneficial. A thorough analysis of all data is now subject of investigation and will be presented at IHCON, including the personal experiences of clients and audiologists.

Conclusions:
Hearing impairment does have a severe influence on the daily life. A new protocol with attention for listening effort and impact on social functioning and work can be of help for clients and dispensers to find the best specific hearing aids and assistive listening devices for rehabilitation of the hearing-impaired individual.

Critical factors in hearing aid selection and evaluation: audiological characteristics, self-report profiles, and hearing aid properties.

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Research questions:
There is lack of standardization in hearing aid selection based on pre-fitting data. Knowledge about appropriate hearing aid selection is scarce and usually derived from laboratory studies or clinical trials with relatively low numbers of participants. There is need for a better method to collect and analyze data about user experiences in large groups of clients, and relate the beneficial effects to audiometric characteristics, user profiles based on self-report, and hearing aid properties.

The goals of this study are to investigate to what degree a-priori assessed compensation needs have been covered by the fitted hearing aids and whether the benefits reported in different domains can be related to audiometric characteristics and to hearing aid properties.

Methods:
For this purpose, we use the first results of the Dutch BRIDGE-approach: a well-structured framework to assess the individual needs and disabilities prior to fitting and the benefit obtained aid after fitting. Data for about 1000 hearing aid users are available now and they include questionnaire results summarized along six scales: detection, speech in quiet, speech in noise, localization, focus/ discrimination, and noise tolerance. Outcome values for these scales have been measured pre-fitting and post-fitting in the same subjects. This allows us to assess the benefit of the newly selected hearing aid for each of the six scales.

Results:
The results will indicate which compensation needs have not been covered yet. In addition, the study will provide valuable information about what to expect from hearing aids, given the individual audiological characteristics and the individual profiles of compensation needs. Possibly, this information can also be helpful in the selection of appropriate hearing aids.

Frequency-domain Reduced-rank Approximations of Music Signals for the Improvement of Music Perception in Cochlear Implant Listeners

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Currently available cochlear implants (CI) only provide a limited number of independent frequency channels leading to a reduced frequency selectivity as compared to normal-hearing listeners. Consequently, timbre and pitch cues are severely distorted such that music instruments cannot be distinguished and melodies are difficult to recognize. Motivated by studies which have shown that solo instrumental music or music remixed at higher signal-to-interference ratios are preferred by CI listeners over complex music ensembles or orchestras, we propose reduced-rank approximations of music signals in the constant-Q spectral domain as a means to reduce effects stemming from cochlear hearing loss and technical limitations of CIs. The rationale behind computing reduced-rank approximations is that they allow to reduce the spectral complexity of music signals. The reduced-rank approximations were obtained based on dimensionality reduction techniques, such as principal component analysis and partial least squares analysis, and were compared to a supervised source separation and remixing procedure. The strategies were evaluated in terms of their ability for mitigating effects of simulated reduced frequency selectivity and with respect to source signal distortions. Established instrumental measures and a newly developed measure indicate a considerable reduction of the auditory distortion resulting from cochlear hearing loss. Furthermore, a listening test with CI listeners reveals a significant preference rate of up to 75% for the reduced-rank approximations.

The eyes and brain reveal your hearing ability

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A relevant aspect of any listening condition is the degree of listening effort required by the listener. A widely used method to quantify listening effort is the task-evoked pupil dilation response. We assessed the influence of hearing loss on the neural correlates of the pupil dilation response. One group (n=17) included listeners (mean age 45.9 years) with normal hearing and the participants in the other group (n = 17; mean age 45.4 years) had mild-to-moderate hearing loss (mean PTA 46.9 dB HL). In two test sessions, participants repeated sentences pronounced by a female speaker that were degraded by noise-vocoding the speech, by imposing stationary noise, or by imposing interfering male speech. Speech Reception Thresholds for 50% intelligibility were adaptively estimated and pupil dilation responses were obtained in both test sessions. Functional magnetic resonance imaging data were acquired in session 2.

As expected, hearing impaired individuals had more difficulties perceiving degraded speech especially for the interfering speech masker. In line with previous studies, the pupil dilation response was smaller in listeners with hearing loss than in those with normal hearing. Depending on the type of degradation, hearing loss was associated with less activation in frontal and parietal areas, and with more activation in temporal areas. The results replicate and extend previous findings showing the effect of hearing acuity and degradation on speech perception and the pupil and brain response to speech perception. We will discuss the relevant processes underlying the results of these outcome measures.

Amplitude-modulation masking for frequency-modulation detection

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It is generally assumed that frequency modulation (FM) is detected as amplitude modulation (AM) for fast FM rates with low carrier frequencies, and for all FM rates with high carrier frequencies. If this is the case, then FM detection in the presence of an AM masker should exhibit the main features of AM masking: tuning, dependency on AM masker depth, negative masking, beating effects and phase effects.

We explored the masking effects produced by sinusoidal AM on detection thresholds of sinusoidal FM for normal-hearing listeners. FM rates ranged between 2 and 64 Hz. The carrier was either a 500-Hz or a 5000-Hz pure tone that was either unmodulated in amplitude or modulated in amplitude at 2 or 16 Hz. The masker AM depth was fixed to either 50% or 25%, and stimulus duration was set to either 500 ms or 1 sec.

Additionally, detection thresholds were tested as a function of the phase relationship between a 2-Hz target FM and a 2- or 4-Hz masking AM, and between a 16-Hz target FM and an 8-, 16-, or 32-Hz masking AM.

The data will be discussed in light of previous studies on AM masking and the modulation filter-bank concept.

Salicylate-induced changes in brain activity in awake guinea pigs

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Tinnitus - the perception of phantom sounds - chronically affects an estimated 10-15% of people (Baguley et al., 2013). Tinnitus is difficult to characterise and frequently intractable, and presents a significant burden to healthcare resources. Studies conducted in human subjects with tinnitus have shown altered patterns of resting-state oscillatory brain activity (e.g. Weisz et al., 2005; Adjamian et al., 2012). However, to date this has not been extensively explored using animal models, which allow more invasive examination of changes in neural activity and their correlation with objective behavioural measures of tinnitus.

Here, we describe an awake preparation for examining tinnitus-related changes in resting-state and auditory-evoked brain activity. Guinea pigs were implanted with electrocorticography (ECoG) electrode arrays, with electrodes positioned on the surface of the dura over left and right auditory cortex and over the cerebellum, to monitor auditory brainstem responses (ABRs). ECoG responses were subsequently compared before and two hours after tinnitus induction with sodium salicylate (350 mg kg-1; i.p.), a drug that reliably induces tinnitus in both humans and animals.

Subtle salicylate-induced changes in oscillatory activity were observed from electrodes over auditory cortex, predominantly at the low-mid frequency (6-10 Hz). Click-evoked cortical field potentials were dramatically enhanced (~100% increase) following salicylate treatment, compared with vehicle treatment, potentially indicating increased sensitivity to sound (hyperacusis). This was despite a significant reduction in the amplitudes of ABRs at the highest frequency (20 kHz). Salicylate-induced changes in spontaneous and auditory-evoked cortical potentials may be due to direct effects on neuronal excitability. Furthermore, in a separate group of animals, significant gap detection deficits in both behavioural and neural responses were observed following salicylate administration.

Tinnitus induction with salicylate is widely used as an acute tinnitus model. The next stage will involve a chronic tinnitus model (induced by acoustic over-exposure), that will allow us to track neural changes throughout tinnitus development in an awake-behaving model.

Acknowledgements
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Measuring tonotopic magnification in the human auditory cortex.

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In the auditory cortex is tonotopically organised, with neighbouring parts of the cortex responding to neighbouring frequencies. It has been shown in other sensory systems, that the cortical surface allocated to a given division of sensory space is proportional to its behavioural importance. For example, in the visual system more cortical surface is dedicated to the fovea than to the periphery. This cortical magnification function has been well characterised in visual cortex using fMRI. Tonotopic magnification in the human auditory cortex (the cortical distance spanned by a given frequency) is still unknown but we expect it to follow behavioural relevance as in the visual system.

Here we present the outline of a study aimed at measuring auditory cortical magnification in normal-hearing subjects using structural and functional MRI at 7T. We will use BOLD-fMRI to measure responses to narrow-band noise stimuli at 32 different centre frequencies. The characteristic frequencies and tuning widths of voxels will be estimated using population receptive field (pRF) modelling. We will use estimates of characteristic frequency and measures of distance on a 3D model of cortical surface to derive the cortical tonotopic magnification function in different auditory cortical areas. Cortical areas will be defined structurally using MR estimates of myelination (longitudinal relaxation time, R1) and functionally, using tonotopic gradient reversals.

Our first goal will be to determine whether cortical tonotopic magnification follows psychophysical measures of cochlear frequency tuning or frequency discrimination. In vision, cortical magnification is proportional to position acuity, a measure which is akin to frequency discrimination acuity in the auditory system. Visual fMRI studies also suggest that the cortical surface representing a receptive field (referred to as the point image) is constant within a given functional cortical area, i.e. the cortical magnification and tuning width function are inversely related. We will derive the auditory point image from our estimates of cortical magnification and population tuning width to determine if the same holds in auditory cortex. Finally, we will quantify how cortical magnification changes in different auditory areas and how this relates the areas' specialism.

We intend to use the measurements from this study to build a model allowing predictions of auditory cortical properties from cortical morphology. One other possible application of our research will be to measure the tonotopic reorganisation potentially associated with hearing loss and/or tinnitus, as measured in animal models but not yet detected by human imaging studies.

Panoramic talk about projects at AMC

Wouter A. Dreschler
Academic Medical Center, The Netherlands

No Abstract

Panoramic talk about hearing Research in Oldenburg during the last year

Birger Kollmeier
Medizinische Physik, Universität Oldenburg & Cluster of Excellence "Hearing4all"

Several dissertations were successfully finished in our group in the last year:

- Christoph Völker: „Instrumental and Perceptual Evaluation of Hearing Devices – Methods and Applications", 2016
- Steffen Kortlang: "Characterization and model-based compensation of suprathreshold auditory processing deficits", 2016
- Regina Baumgärtel: "Techniques for improving speech intelligibility and spatial perception in users of bilateral cochlear implants", 2016.
- Wiebke Schubotz, "Performance of Current Models of Speech Recognition and Resulting Challenges". 2015.
- Anne Schlüter, "Speech recognition tested at fixed, positive signal-to-noise ratios using time compression: Methods and applications". 2015.
- Sabine Hochmuth, "Assessment of language- and talker-specific factors influencing speech intelligibility in noise: A multilingual comparison". 2015
- Marc Rene Schädler, "Robust automatic speech recognition and modeling of auditory discrimination experiments with auditory spectro-temporal features". 2015.

The research building NeSSy in Oldenburg is fully operational since January 2016, since the major facilities (i.e., functional MRI, MEG and virtual reality lab) have been successfly put to work

Spectral and binaural loudness summation and the need for individual assessment in auditory rehabilitation.

Mirjam van Geleuken, Monique Boymans, and Wouter A. Dreschler
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Introduction:
Aversiveness of loud sounds is one of the most mentioned complaints about hearing aids. There is more aversiveness with two hearing aids than with one. Earlier research from Dirk Oetting et al. at the Oldenburg group indicated that the restoration of the narrowband loudness (NB) perception in hearing-impaired listeners may not be adequate for the perception of loud broadband (BB) signals, presented bilaterally. For hearing aid fitting it is important to know more about what the effect of different signals are on loudness growth, unilaterally as well as bilaterally, for different hearing losses.

Research questions:
Two aspects of loudness perception should be taken into account: spectral loudness summation and binaural loudness summation. This includes also the binaural loudness perception of broadband signals. The combined effect has to be considered because the hearing aid typically process binaural broadband signals later on. This presentation will focus on the diagnostic aspects.

More specifically:
1. What is the effect of individual characteristics of loudness perception in a wide range of typical candidates for bilateral hearing aids? For this purpose we use the classification of audiograms according to Bisgaard in order to study the effects of hearing loss.
2. What is the effect of spectral contents of the signals? We investigated this by using separate tests with high frequency noise and low frequency noise.
3. What technique is feasible for the clinical applicability? We investigated the pros and cons of loudness matching.

Methods:
To fulfil the purpose of this study we intend to objectify the hearing ability and loudness functions of hearing impaired listeners with a symmetric mild to moderate hearing impairment. We included participants with audiometric configurations N2, N3, N4, S2 and S3 according to Bisgaard. For the measurements we used a well-standardized technique ACALOS and also loudness matching to determine the loudness perception. Loudness matching will be used as a faster technique to get information about the individual loudness perception.

Results:
Preliminary results will be shown of ten subjects.

Restoring loudness perception for hearing-impaired listeners

Dirk Oetting (1), Volker Hohmann (2), Jens-E. Appell (1), Birger Kollmeier (2), Stephan D. Ewert (2)

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Loudness can be considered as one key factor during the fitting process of a hearing aid. Restauration of normal-hearing loudness perception in hearing-impaired listeners has been already mentioned in many gain adjustment methods in the literature. However, assessment of loudness perception is typically not used during fitting, and if focus is on restoring narrowband loudness perception in each ear. More natural, broadband and binaural signals are only used in the verification stage or for revision of the first fit. A possible reason for loudness complaints in aided listeners might thus be that loudness measurements are typically conducted with monaural narrow-band signals while binaural broadband signals as speech or environmental sounds are typical in daily life. Recently we showed large individual differences of the loudness summation for broadband signals in hearing-impaired listeners after gains were adjusted to restore monaural narrowband loudness perception. Some listeners showed a considerably higher-than-normal loudness summation, most prominent for binaural presentation, and the effect increased with increasing signal level. Thus, on an individual basis, loudness cannot be restored to normal for arbitrary input signals when using a fixed set of level and frequency dependent gains. Therefore, we developed a binaural bandwidth-adaptive dynamic compression (BBDC) algorithm that allows real-time changes of the gains suited for narrowband loudness compensation, depending on input signal properties like bandwidth, binaurality, and level of the signal. Channel gains were reduced based on the individual loudness function of signals with different bandwidth. Loudness functions in 15 HI listeners with BBDC compensation were similar to the average normal-hearing loudness function for narrow- and broadband signals. Average BBDC gains were similar to average gains prescribed by NAL-NL2, but large difference were observed when compared indiv idually. Normal loudness perception was also shown for 20 real-world signals with levels between 40 and 90 dB SPL. The results indicate that the proposed BBDC compensation method compensates for the individual loudness perception in a more appropriate way than typical threshold-based procedures.

Listener's preference for time constant settings in a noise-reduction algorithm

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Academic Medical Center, The Netherlands

Research questions:
Understanding speech in noise is a difficult task, especially for hearing aid users. For this reason, noise reduction algorithms have been implemented in hearing aids in order to improve listening comfort. Details of the algorithm's exact decision rules are essential for fitting the device to the preferences of the user, but they are often unknown to the clinician. Also, there is only little knowledge about the perceptual effects. Therefore, research is initiated in order to characterize the behavior of different noise reduction algorithms and investigate the perceptual effects.

Temporal aspects of speech and noisy speech have shown to be important in determining speech intelligibility and listening comfort. These aspects might also play an important part in successfully implementing a noise reduction algorithm. Therefore, we are interested in whether time constants in noise reduction algorithms are involved in determining listener's preference.

Methods:
In a currently running experiment we test whether listeners can detect differences in time constants in a noise reduction algorithm and if so, which time constant they prefer. Test signals consist of Dutch sentences in speech shaped noise processed with a minima controlled recursive noise reduction algorithm. Gain adjustment is being slowed down by applying an exponential smoother. Subjects listen to the different test signals and determine whether they can detect differences and select their preference for a particular signal in terms of speech quality, noise quality and overall quality.

Results:
Some preliminary results will be shown.

Short talk

Panoramic talk on projects at VUMC

Theo Goverts
VUMC

No Abstract

Impact of stimulus-related factors and hearing status on listening effort as indicated by pupil dilation

Adriana A. Zekveld (1,2,3), Barbara Ohlenforst (1,4), Thomas Lunner (3,4,5), Dorothea Wendt (4,6), Graham Naylor (7), Yang Wang (1,4), Niek J. Versfeld (1), Sophia E. Kramer (1)

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Previous research has reported effects of masker type and signal-to-noise ratio (SNR) on listening effort, as indicated by the peak pupil dilation relative to baseline (PPD) during speech recognition. The presence of a competing talker during speech recognition generally resulted in larger PPDs as compared to the presence of a fluctuating or stationary background noise. At about 50% correct sentence recognition performance, increasing SNRs generally results in declining PPDs, indicating reduced effort. However, the decline in PPD over SNRs has been observed to be less pronounced for hearing-impaired (HI) compared to normal-hearing (NH) listeners. One may expect further decrease in PPD across positive SNRs where speech recognition performance is high. In contrast, HI listeners usually report high listening effort during communication, even when speech intelligibility has been rated as good.

The aim of the present study was to examine the interplay between hearing-status, a broad range of SNRs corresponding to intelligibility levels varying from 0 to 100% correct sentence recognition, and different masker types on the PPD during speech perception.

Twenty-five HI and 32 age-matched NH participants listened to sentences across a broad range of SNRs, masked with speech from a single talker (-25 dB to +15 dB SNR) or with stationary noise (–12 dB to +16 dB). Correct sentence recognition scores and pupil responses were recorded during stimulus presentation.

For both groups and masker types, the relation between PPD and SNR described an inverse U-shaped function. The mean PPD increased with decreasing SNR until approximately 25% correct sentence recognition was reached and with further decreasing SNR, the mean PPD decreased. Mixed-model ANOVAs revealed significant interactions between hearing-status and SNR on the PPD for both masker types. NH listeners showed larger PPDs than HI listeners at low SNRs (-12 dB to -4 dB SNR), but NH listeners had smaller PPDs than HI listeners when SNRs were high (0 dB to +16 dB SNR).

HI listeners appear to have more listening effort across a relatively large range of positive SNR than NH listeners do. Our data show a different pattern of PPDs across SNRs for NH and HI listeners, which indicates that listening and the allocation of effort during listening in daily life environments may be different for NH and HI listeners.

Short talk

Panoramic talk about projects at KU Leuven

Jan Wouters
KU Leuven

No Abstract

Can bimodal cochlear implant listeners use true binaural hearing to understand speech in noise?

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Introduction. Spatial separation of speech and noise can improve speech intelligibility, due to the use of interaural time and level differences (ITDs and ILDs respectively). For normal-hearing listeners, different mechanisms are claimed to be responsible for spatial release from masking. Firstly, spatial separation yields one ear with a better signal-to-noise ratio (SNR) than the other: the "better ear" (i.e., a monaural mechanism). Secondly, listening with two ears (instead of the better ear only) yields an extra binaural benefit in speech intelligibility: squelch. Mostly, squelch is attributed to binaural decorrelation and/or spatial attention (i.e., binaural mechanisms). Spatial attention is said to be most important for informational maskers. For bimodal cochlear implant listeners, it is not clear which of these mechanisms are responsible for spatial release from masking

Hypotheses. We hypothesize that no binaural mechanisms are responsible for spatial release from masking in bimodal listeners. We hypothesize that speech intelligibility in noise is merely determined by the signal-to-noise ratios in the cochlear implant and the hearing aid microphones

Methods. Bimodal hearing was simulated with a noise vocoder at one ear, and a low-pass filter at the other ear. A Flemish matrix sentence test was used to measure the speech reception threshold. We investigated three different maskers, spectrally matched to target speech: speech-weighted noise (SWN), a competing talker and modulated SWN with the same temporal envelope as the competing talker. In a first experiment, we measured spatial release from masking for different spatial setups and compared monaural (cochlear implant only) with binaural (bimodal) situations. Spatial hearing was simulated with head-related transfer functions. In a second experiment, we varied the broadband noise level at both ears independently to investigate their relative importance in speech intelligibility.

Results. Spatial release from masking was mainly determined by the signal-to-noise ratio at the vocoded (cochlear implant) ear; the benefit of adding the low-pass filtered ear was determined by the signal-to-noise ratio at that ear.

Conclusion. Spatial release from masking for bimodal listeners was merely determined by monaural signal-to-noise ratios. As long as ITDs remain unperceptible for bimodal cochlear implant listeners, it is therefore not necessary to take into account binaural mechanisms in sound processing strategies to improve speech understanding; improving monaural signal-to-noise ratios should be the main objective.

Auditory Steady-State Responses as a measure to evaluate temporal processing of speech envelope characteristics in adults with dyslexia

Tilde Van Hirtum (1,2), Pol Ghesquière (2) & Jan Wouters (1)
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The temporal envelope of the speech signal is an important cue for speech intelligibility. Variations in amplitude of the envelope can be modeled and characterized by measures of amplitude modulation (AM) and rise time (RT). Firstly, the envelope consists of multiple AM-rates ranging from 2 to 50 Hz. Particularly important are the slower AM-rates because they comprise acoustic-phonological information about syllables (4 Hz) and phonemes (20 Hz). Secondly, RT provides information about syllable onset and prosody and is essential for phoneme discrimination. Neural temporal processing can be investigated using auditory steady-state-responses (ASSRs). ASSRs thus provide an objective measure to determine sensitivity to these speech-related cues.

Growing evidence exists that developmental dyslexia, a specific learning disorder characterized by severe reading and spelling difficulties, is related to a temporal processing deficit. Furthermore, decreased sensitivity to distinct speech envelope characteristics (AM, RT) influences speech perception and might interfere with the development of adequate phonological representations, resulting in the observed reading and spelling problems.

The present research project investigated neural processing of phonemic- and syllabic-rate modulations as well as rise time characteristics with ASSRs. ASSRs were evoked by amplitude-modulated noise band stimuli at 4, 10, 20 and 40 Hz along with different modulation envelopes with varying rise times. Preliminary analyses demonstrate deviant responses to short RT stimuli for dyslexics, mainly for syllabic-rate processing, which are related to phonological awareness and speech-in-noise perception. These results suggest an increased processing effort at syllable level for dyslexic adults for stimuli with shorter rise times.

ECC neurophysiologically based coding strategy for cochlear implants: Real time implementation considerations

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A cochlear implant coding strategy based on neurophysiological properties of neural excitation has been implemented using Matlab/Simulink. The Excitability Controlled Coding (ECC) strategy uses a model of the excitability state of the target neural population to determine its stimulus selection, with the aim of more efficient stimulation as well as reduced channel interaction. By taking into account the estimated refractory behavior of the stimulated neural populations, the excitability state is computed and updated after the presentation of each stimulus, and is then iteratively used in the selection of the next stimulus. Additionally, ECC regulates the frequency of stimulation on a given channel as a function of the corresponding input stimulus intensity. This was then implemented as a real-time xPC/Speedgoat Simulink model. Details of the model, implementation and results of benchtop tests are presented and discussed.

Welcome

Rainer Martin
Ruhr-Universität Bochum

Models and mechanisms of temporal interactions in cochlear implant stimulation of the auditory nerve

Ian C. Bruce
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In the cochlear implant (CI) literature, interpretation of psychophysical and evoked-potential data is often based on the assumption that auditory nerve (AN) fibers can very faithfully encode electrical pulse trains, with the only limitation being very short refractoriness (< 1 ms). However, recent physiological data suggest that a number of different types of temporal interactions can occur in AN fibers, as well as refractoriness that can extend out to at least 4 ms. This may at least partially explain why using very high pulse rates in CI speech processors does not improve speech intelligibility for most CI users. In the first part of this talk, I will discuss conceptual models of refractoriness, accommodation, adaptation and facilitation to describe the types of temporal interactions that are observed in the physiological data. Second, I will explore some potential biophysical mechanisms behind the different forms of temporal interaction and describe computational models th at are being developed to evaluate the possible biophysical causes. Simulation results suggest a number of different voltage-gated ion channels could contribute to the variety of temporal response properties measured in the physiological data. Furthermore, it appears that the ion channel locations and the site of action potential generation on the AN fiber may additionally contribute to the observed heterogeneity of temporal interactions in the data.

Computational modeling of the auditory periphery

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Hearing-related phenomena in general and hearing loss in particular are often described in terms of the underlying physiology and pathology of the hearer's auditory system. Computer models of the system are potentially valuable tools for making sense of psychophysical phenomena and exploring hypotheses about how hearing works. They can also be used to explore hearing loss as a consequence of peripheral pathology. Our ability to simulate auditory nerve firing patterns is now well developed and a number of models are already in common use. However, modelling of signal processing in the brainstem is largely restricted to small-scale models when it would be preferable to have larger integrated models of the signal processing that takes place at the neuronal level. This talk will present a software platform that can be used to synthesize the knowledge gathered by physiologists into a coherent framework represented at the cellular level. There are many cell types involved in auditory processing and each type responds differently to the same acoustic stimulus as a result of differences in anatomy, connectivity and expression of sub-cellular membrane channels. By adopting a modular approach the software allows different cell types and their connections to be introduced, modified and otherwise manipulated. Input to the brainstem model is from a simulation of the auditory nerve response and this allows the whole model to be driven by either acoustic stimulation, cochlear implant stimulation or voltage and current clamping of individual cells. The intention is to use the software in more general studies of psychophysics, hearing loss, medical measurements and cochlear implant technology. The software is freely available from the author including a graphical user interface for familiarization purposes.

Real-time demonstration of Neural Network based Speech Enhancement

Tobias Goehring (1), Federico Bolner (2), and Stefan Bleeck (1)
1) University of Southampton, United Kingdom
2) Cochlear Technology Centre, Belgium

We present a real-time demonstration of the Neural Network based Speech Enhancement (NNSE) algorithm framework. This algorithm has been developed for improving speech understanding in adverse listening conditions with high levels of background noise for application in hearing devices such as hearing aids and cochlear implants. The outcome of a listening study using this framework for improved speech comprehension in background noise for cochlear implant users will be presented as a poster at the ICanHear poster session. Further details about the implementation for the real-time setup will be given during the presentation. The listening demonstration comprises a comparison between NNSE, the unprocessed condition and a state-of-the-art Wiener Filter based Speech Enhancement algorithm.

A Noise Reduction Post-filter for Resolving Front-back Ambiguity in Single-Microphone Binaural Hearing Aids utilizing a Nearby External Microphone

Dianna Yee (1), Homayoun Kamkar-Parsi (1), Rainer Martin (2), Henning Puder (1)
1) Sivantos GmbH
2) Ruhr Universität Bochum

In this demonstration, we use a nearby external microphone for addressing front-back ambiguity in single-microphone hearing aid devices. Strategic placement of the external microphone is able to provide benefits from the body shielding back-directional noise and therefore information for discriminating between the frontal and back hemispheres. The scattering effects of the body is first analyzed to yield a placement strategy of the external microphone for maximizing the shielding effect of the body against back-directional noise sources while also optimizing for speech intelligibility. Assuming strategic placement, a probabilistic approach for a frontal target source presence probability (FTSPP) is derived by modeling the probability density function of the ratio of signal variance between microphone devices during frontal speech presence and absence. The probability density functions are parameterized using the empirically measured acoustic shielding of back-directional noise by a rigid body. Using the FTSPP estimator, a more comprehensive noise estimator is proposed that considers both stationary and non-stationary noise. The performance of the proposed noise estimator is evaluated in its application for post-filtering the binaural beamformer. A post-filter is then realized using the proposed noise estimator and also the state-of-the-art noise estimator. The performance of the post-filter computed using the state-of-the-art and the proposed noise estimator is compared to evaluate the performance of the noise estimators. The effect of post-filtering using the proposed noise estimator is to provide further reduction of directional noises from the lateral and back direction while preserving the frontal target signal in the look direction.

Adaptive Binaural Beamforming Based on Binaural Localization

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In this work we present a real time demonstration for binaural speaker localization and separation using hearing-aid microphones. Essentially, adaptive binaural beamforming requires information on the desired source locations relative to the current head position. The localization algorithm needs to be robust to unknown source/receiver con gurations and to adverse acoustic conditions. In this work we propose binaural localization algorithm based on optimizing cost functions derived from beamforming technique. Our approach uses joint IPD and ILD cues which are characterized in the form of HRTF. We use prototype HRTF that could be extracted either from a database or from a spherical head model. Our method is able to localize multiple source positions in the frontal azimuth plane without prior training of binaural cues. The method also enables the localization across a wide range of frequencies in the presence of reverberation. Next, we integrate the proposed localization algorithm in a generalized side-lobe canceller (GSC) to separate simultaneous speakers. The demonstration of the localization/separation framework is implemented on a real-time Speedgoat target machine.

Implementation of a dereverberation algorithm for CI recipients on a realtime system

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Reverberation and noise reduce speech intelligibility significantly and affect especially hearing impaired persons. Several denoising and dereverberation techniques have been developed in the past. The processing algorithm evaluated in this study consists of three steps: the denoising step, the removal of late reverberation parts and finally a general dereverberation stage based on computed coherence between the input signals at both ears. For the denoising part, a speech distortion weighted multi-channel Wiener filter (SDW-MWF) with an adaptable voice activity detector (VAD) is used in order to achieve an optimal trade-off between noise reduction and speech signal distortion.

In the second step a spectral subtraction filter is used in order to reduce late reverberation. Finally, a coherence filter is applied based on the assumption that the reverberated parts of a signal show a low coherence between the left and the right ear. In addition to the basic multi-channel Wiener filter approach which attenuates low coherent signal parts, an adaptation with a non-linear sigmoidal coherence to gain mapping is used.

The dereverberation algorithm was implemented on a Speedgoat xPC Target realtime system which processes two input signals and generates two output signals. The input signals are obtained from hearing instrument microphones placed at the two ears of the subject which pick up to speech sounds in a reverberant environment. The processed signals are presented to the two ears of a listener either via headphones (for NH subjects) or via direct input into the CI sound processors (for CI recipients).

Speech Enhancement Based on Neural Networks Improves Speech Intelligibility in Noise for Cochlear Implant Users

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Speech understanding in noisy environments is still one of the major challenges for cochlear implant (CI) users in everyday life. In this study, we propose a speech enhancement algorithm based on neural networks (NNSE) for improving speech intelligibility in noise for CI users. The algorithm decomposes the noisy speech signal into timefrequency units, extracts a set of auditory-inspired features and feeds them to the neural network to produce an estimation of which frequency channels contain more perceptually important information (higher signal-to-noise ratio, SNR). This estimate is used to attenuate noise-dominated and retain speech-dominated CI channels for electrical stimulation, as in traditional n-of-m CI coding strategies.

The proposed algorithm was evaluated by measuring the speech-in-noise performance of 14 CI users in three different types of background noise. Two distinct NNSE algorithms were compared in this experiment: a speaker-dependent algorithm, that was trained on the target speaker used for testing, and a speaker-independent algorithm, that was trained on different speakers. Significant improvements in speech intelligibility in stationary and fluctuating noises were found over the unprocessed condition for both speaker-dependent and speaker-independent algorithms, with the first algorithm providing bigger improvements. Results indicate that the proposed algorithm has the potential to improve speech intelligibility in noise for CI users and proves to generalize to a range of acoustic conditions, whilst meeting the requirements of low computational complexity and processing delay in CI devices.

Non-Intrusive Speech Intelligibility Prediction Using Hidden Markov Models

Mahdie Karbasi and Dorothea Kolossa
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Automatic speech intelligibility prediction has always been a challenging and important topic in the field of speech signal processing, since listening tests are expensive, time consuming and cannot be used online. Most of the available objective speech intelligibility measures are intrusive methods, as they require a clean reference signal in addition to the corresponding noisy/processed signal at hand. In order to overcome the problem of predicting the speech intelligibility in the absence of the clean reference signal, we have proposed two new approaches.

The first approach is based on a recognition/synthesis framework, the **twin hidden Markov model (THMM)**. We employ the THMM for synthesizing the clean features for an intrusive intelligibility prediction method. The THMM-based speech intelligibility measure performs equally well as well-known intrusive measures like the short-time objective intelligibility (STOI). Moreover, it is highly correlated with the human speech recognition results in different noise conditions.

In the second approach, an HMM-based automatic speech recognition system is used to extract a discriminative measure for predicting the speech intelligibility. The proposed discriminative measure, the **HMM-based log likelihood ratio (HLLR)**, only requires the noisy speech signal and its transcription as the input. It is shown that the HLLR can be used as an accurate predictor for the intelligibility of single words, with the added benefit of not requiring the clean version of the noisy signal.

Decision Device Comparison for Model-based Analysis of ITD Perception in Normal Hearing Listeners

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Interaural time differences (ITDs) play an important role in sound localization and speech understanding in noise. Previously (Moncada-Torres et al., 2016), we developed a framework capable of predicting ITD just noticeable differences (JNDs) based on a physiological model of the auditory nerve (AN). However, the decision device employed there did not take into account the variability of the used data in a straightforward manner.

In this work, we predicted ITD JNDs using two different decision devices in normal hearing listeners using information at the AN level. AN responses to acoustic stimuli from both ears (in the form of spikes) with and without introduced ITDs were simulated using the phenomenological model proposed by Zilany et al. (2009). Next, we used the shuffled cross-correlogram analysis (SCCs, Joris, et al., 2006) to quantify ITD encoding across the AN of both channels. Assuming that the auditory system is more sensitive to smaller ITDs, we corrected the SCC curves using the weighting function proposed by Stern and Shear (1996). Then, we predicted the imposed ITD by choosing the global maximum of the corrected curves. The distributions of the predicted reference and imposed ITDs were fed to two different decision modules: the receiver operating characteristic (ROC) and the detection index (d'). These allowed us to calculate the ITD JND as the 79.4% and 1.5 point, respectively, of the neurometric curve. Finally, we evaluated the performance of the decision devices' predictions by comparing them against literature behavioural data using pure tones with frequencies from 250 to 1400 Hz.

The proposed framework showed similar trends as in psychoacoustical data, with the d' metric being higher correlated with it. Future work will be focused in using the framework's improved pipeline to predict ITD discrimination performance in hearing impaired listeners and well as in optimizing hearing aids/cochlear implants signal processing.

Acknowledgments
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Functional modelling of interaural time difference discrimination in acoustical and electrical hearing

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Interaural time differences (ITDs) are important for sound source localisation. We present a model to predict the just noticeable differences (JNDs) in ITD discrimination for normal hearing and electric stimulation through a cochlear implant. We combined periphery models of acoustic and electric stimulation together with a novel JND in ITD estimation stage, which consists of a shuffled cross correlogram and a binary classifier characterisation method. Furthermore, an evaluation framework is presented which is based on a large behavioural dataset. The model correctly predicts behavioural observations for unmodulated stimuli, such as pure tones and electric pulse trains and modulated stimuli for modulation frequencies below 30 Hz. For higher modulation frequencies, the model predicts the observed behavioural trends, but tends to estimate higher ITD sensitivity. The presented model is useful for investigating the implications of modifying the stimulus waveform on ITD sensitivity.

Modeling Speech Intelligibility based on Envelopes Derived from an Auditory Nerve Model

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Speech intelligibility models aim to predict the human ability to understand speech in adverse listening conditions. Most previous speech intelligibility models do not capture the detailed physiological transformation the acoustic signal undergoes in the auditory periphery. The goal of this study was to combine an established speech intelligibility model with the auditory signal processing of an auditory nerve (AN) model. Specifically, the back-end processing of the multi-resolution speech-based envelope power spectrum model (mr-sEPSM; Jørgensen et al., 2013) was combined with the auditory nerve model by Zilany and Bruce (2014). The presented work calculated signal-to-noise-ratios in the envelope domain (SNRenv) for normal-hearing listeners based on different envelope representations derived from the AN model: (i) instantaneous firing rates (ii) peristimulus time histogram (PSTH) of auditory nerve spike trains, and (iii) shuffled-correlogram based analyses of AN spike times (Heinz and Swaminathan, 2009). The SNRenv patterns showed good agreements compared to the SNRenv patterns calculated from the acoustic (i.e. Hilbert) envelope (Heinz, 2016). Furthermore, speech intelligibility for normal-hearing listeners based on envelopes derived from PSTHs for CLUE sentences (CLUE; Nielsen and Dau, 2009) was predicted accurately in speech shaped noise (SSN), sinusoidally amplitude modulated noise (SAM) and speech-like noise (ISTS; Holube et al., 2010). Effects of hearing loss resulted in poorer speech intelligibility predictions. The work provides a foundation for quantitatively modeling individual effects of inner and outer hair cell loss on speech intelligibility.

Speech perception and localisation with real-time SCORE in bimodal users

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Introduction: In current clinical practice for bimodal stimulation a combination of an off-the-shelf hearing aid (HA) and a cochlear implant (CI) is often used. Sometimes, the HA and the CI are provided and fitted at different locations and by different clinicians. Additionally, the processing in CI and HA are often very different, because of the different stimulation modes and the fact that the devices were not developed with combined use in mind. This can be true even in similar processing blocks like compressors.
To address these problems, a loudness balancing strategy SCORE was created (Francart & McDermott, 2012), aimed at normalizing loudness perception and improving binaural loudness balance for bimodal users. It uses different loudness models. For each signal frame and each ear, a reference loudness is calculated, which is for example the loudness perceived by a normal hearing person. For the CI and for the HA side the loudness after processing is also calculated, taking into account the individual hearing impairment. The stimulation levels are then adjusted according to the loudness differences. This strategy has not been tested on a real-time platform before.

Methods: SCORE, the CI model of Cochlear (ACE strategy), and a linear HA were implemented as Simulink blocks on the real-time platform. Eight bimodal users participated in the study. For each participant the CI model was set according to the clinical MAP. The HA model was fitted according to the NAL-RP rule. All measurements were done in free field.
Speech understanding in quiet was measured with CVC words, in 3 listening conditions (HA-only, CI-only, and HA+CI) and 2 intensities (50 and 65 dBA). For speech understanding in noise sentences were presented in a competing talker background. For the localization test an array of 13 loudspeakers was used, placed in the horizontal plane from -90 to +90 degrees. For each test, intensity, and condition we compared between the SCORE algorithm switched on and off.

Results: Speech perception in quiet was in general better for HA-only, for CI-only at 50 dBA, and for HA+CI at 65 dBA. Due the inter-subject variability, however, the benefit was not significant. No benefits were observed for speech perception in noise. After training, three out of four participants had better sound localisation with SCORE.

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Towards a bio-inspired coding strategy for cochlear implants

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Research efforts to improve coding strategies for cochlear implants (CIs) were recently more frequently directed towards physiologically based approaches such as MP3000 strategy. However, the most widely used clinical speech processing strategies still do not regard the auditory nerve fibers' (ANF) capacity for conveying the acoustical information to the electrical stimulation.

In the first part of our study, frequency decomposition of an audio signal was done by a 4th order all-pole infinite impulse response (IIR) Gammatone filterbank which is more physiologically-based filterbank compared to the standard Fast Fourier Transform (FFT) filterbank that is being used in the Advanced Combination Encoder (ACE) strategy. Other conditions such as FFT filterbanks with higher frequency resolutions and an FFT filterbank with the Gammatone matched frequency mapping were considered for the comparison. Two experiments; melodic contour identification (MCI) and just noticeable difference (JND) were conducted with normal hearing (NH) and CI subjects. The results for this part will be shown and discussed as confusion matrices.

Exciting of ANF by an electrical pulse can change its response to the next pulse which is largely ignored in frame-based strategies such as ACE. Thus, in the second part of our study temporal response properties of ANFs such as refractoriness, facilitation, spike rate adaptation (SRA) and accommodation were considered in the selection of channels with the highest energy content. Apart from ANFs temporal considerations, spatial spread of the electric field which has a major impact on the spectral resolution of CI users was taken into account. The aforementioned phenomena can change the excitability of the ANFs in response to electrical stimulation. Preliminary results for this part will be shown as electrodograms.

Spatial details in bilateral cochlear implants

Dietmar Wohlbauer, WaiKong Lai, Norbert Dillier
University & University Hospital Zurich

The human auditory system has the ability to resolve complex auditory scenes by interpreting acoustic as well as visual cues in their surrounding. The acoustic cues are a compound of level and temporal differences between the ipsi- and the contralateral ear or the so called binaural cues.

Furthermore, the spectral information resulting from the shape of the pinnae, the so called monaural cues, are important for the individuals localization accuracy. Subjects which suffer from a severe hearing loss and who are equipped with cochlear implants (CIs) are disadvantaged in perception of their acoustic environment not only by their hearing loss but also by the used technology. The question arises if common CI signal processing chains preserve spatial details. We investigate the degradation of spectral information evoked by the speech processors signal collection with BTE microphones and further transduction via a bilateral vocoder simulation.

Cross –correlation model of interaural time difference coding listeners with bilateral cochlear implants

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A physiologically inspired phenomenological model of interaural time difference (ITD) coding in bilateral cochlear implant listeners is presented. The model performs a coincidence detection analysis between the spike-train inputs from the two ears and estimates the lateralization of binaural input signal relative to the midline. The model consists of three major components: a computational model of the auditory nerve fiber responses to electrical stimulation (Joshi et al., 2016), a binaural processing stage performing a neural cross-correlation analysis (Joris et al., 2003) and the decision device. The model was used to predict the effects of stimulation level, stimulation pulse rate, modulation frequency and jitter of inter-pulse-intervals on ITD discrimination thresholds and compared to the behavioral data from several studies. The model is shown to account for the main characteristics in the behavioral data. This model can serve as a useful framework to study the effect of various stimulation strategies on ITD sensitivity in bilateral cochlear implant listeners.

Machine listening for continuous improvement of speech intelligibility in hearing devices

Bernd T. Meyer
Center for Speech and Language Processing, Johns Hopkins University, Baltimore, USA

Current behind-the-ear hearing aids (HA) allow to perform spatial filtering to enhance localized sound sources; however, they often lack processing strategies that are tailored to spoken language. Hence, without a feedback about speech quality achieved by the system, spatial filtering potentially remains unused. In this talk I will present our approach to this problem, which is based on a combination of deep learning and estimates of direction of arrival (DOA). The DOA component is used to identify positions of localized sound sources in the acoustic scene. We apply a deep neural net (DNN) to determine which of these sound sources is a potential speech source. To this end, we apply performance measures (borrowed from automatic speech recognition) to phoneme posterior probabilities of a DNN. I will present results obtained with baseline performance measures such as entropy, and also show alternatives developed in our lab which support the idea that our approach can be used to optimize beam angles in multi-channel HAs and to identify speech sources in complex acoustic scenes.

Modelling of neural signal processing in the impaired auditory system

Torsten Dau
DTU

No Abstract

Invited talk

Models of hearing impairment and their consequences for signal processing algorithms

Birger Kollmeier
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No Abstract

Innovative solutions for improved communication

Dorothea Kolossa
Ruhr-University Bochum

Signal processing can be applied in a multitude of ways for improving the communication of hearing-impaired listeners, and a number of these have been addressed within the ICanHear project:

We have worked on cochlear implant algorithms to address some of the current system limitations, on advancing speech signal processing in difficult listening environments, and on integrating newly available sensors and smart phones to improve speech quality in noise.

This talk will give an overview of these recent achievements in applying signal processing to hearing aid and CI technology, and it will specifically focus on the value of integrating new sensors and devices for improving speech communication quality.

Evaluation and comparison of CI coding strategies regarding spectral and temporal cues

Chris James (1,2), Mathieu Marx (2), Damir Kovacic (3)
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Speech recognition performance by cochlear implanted subjects is largely influenced by demographic factors such as etiology and duration of hearing loss. Improvements in speech recognition performance, either in quiet or in competing background noise, have not been demonstrated using more than eight to twelve spectral channels. Similarly performance does not continue to increase for temporal envelope bandwidths greater than ~200 Hz.

Another important aspect of speech perception is voice pitch, or fundamental frequency F0, which is a string cue to speaker identity, emotional state and clarifies meaning via prosody in Western languages and tone identity in Oriental languages. We describe several studies of voice pitch perception using a novel experimental sound coder that allowed separate control of spectral and temporal resolution. In addition we report on a cohort of sixty-one cochlear implant subjects using clinical processors.

The conventional approach to improving voice pitch perception by enhancing temporal envelope modulation at F0 appears ineffective and limited to a similar extent as for speech recognition. Voice pitch perception varied widely across CI subjects; however both studies indicated that better spectral resolution in low frequency channels may be the key to improving performance.

Signal processing algorithms for improced speech and music perception with cochlear implants

Waldo Nogueira
Medical University Hannover, Cluster of Excellence "Hearing4all", Hannover, Germany

This presentation will review sound coding strategies and signal processing algorithms that try to improve speech and music perception with cochlear implants (CIs).

First, sound coding strategies that try to improve the spectral representation of the sounds and the transmission of low frequencies will be presented. The algorithms and its evaluation in the context of music and speech perception will be reviewed.

Second, a signal processing method based on source separation will be introduced that can be used to separate music components and remix them such that the music can be better appreciated by CI users. This source separation method can also be applied to separate speech signals from interferences.

Finally some future directions in signal processing and sound coding strategies will be outlined.

Evaluation - Consequences of hearing impairment and benefits from signal processing

Jan Wouters
KU Leuven

No Abstract

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