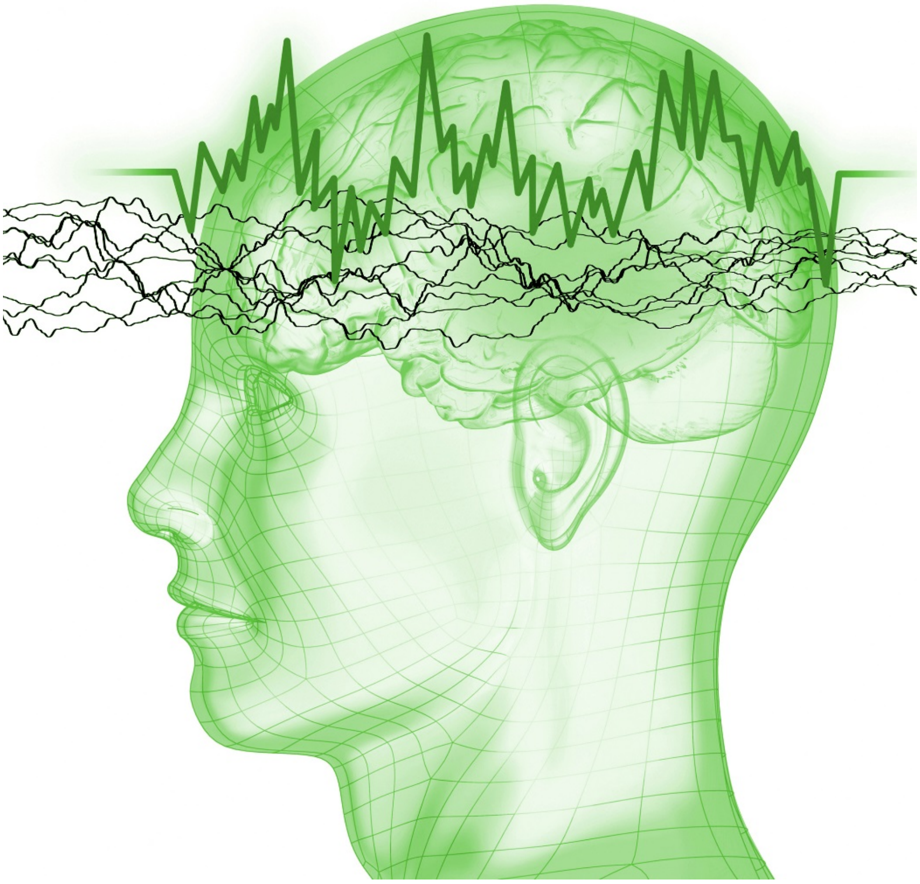
6th

International Symposium on Auditory and Audiological Research

ISAAR 2017

"Adaptive Processes in Hearing"



August 23-25, 2017
Hotel Nyborg Strand, Denmark

Programme and abstracts

About ISAAR

The “International Symposium on Auditory and Audiological Research” is formerly known as the “Danavox Symposium”. The 2017 edition corresponds to the 27th symposium in the series and the 6th symposium under the ISAAR name, adopted in 2007. The Danavox Jubilee Foundation was established in 1968 on the occasion of the 25th anniversary of GN Danavox. The aim of the foundation is to support and encourage audiological research and development.

Funds are donated by GN ReSound (formerly GN Danavox) and are managed by a board consisting of hearing science specialists who are entirely independent of GN ReSound. Since its establishment in 1968, the resources of the foundation have been used to support a series of symposia, at which a large number of outstanding scientists from all over the world have given lectures, presented posters, and participated in discussions on various audiological topics.

More information can be found at www.ISAAR.eu. Proceedings from past symposia can be found at www.audiological-library.gnresound.dk and proceedings.isaar.eu.

Cover illustration by Wet DesignerDog (www.wetdesignerdog.dk)
with thanks to Eva Helena Andersen

Welcome to ISAAR 2017

The general topic of the ISAAR 2017 symposium is "Adaptive processes in hearing". The concept is to consider this topic from different perspectives, including current physiological concepts, perceptual measures and models, as well as implications for new technical applications.

The programme consists of invited talks as well as contributed talks and posters. The symposium is divided into five sessions, to which the following speakers have been invited:

1. ***Adaptive behavior in complex listening environments***
   Virginia Best, Piotr Majdak, Owen Brimijoin

2. ***Neural mechanisms and modeling of adaptive auditory processes***
   Tobias Moser, Jan Schnupp, Elia Formisano, Mounya Elhilali

3. ***"Maladaptive" processes in hearing***
   Pim van Dijk, Roland Schaette, Martin Pienkowski

4. ***Electrophysiological correlates of auditory adaptation***
   Tobias Overath, Stefan Debener, Manuel Malmierca

5. ***Adaptive and learning processes with hearing devices***
   Monita Chatterjee, Nikolai Bisgaard, Tom Francart, Andrew Sabin

In addition to these scientific presentations, one of the objectives of ISAAR is to promote networking and create contacts between researchers from different institutions in the fields of audiology and auditory research. ISAAR is a great opportunity for young scientists to approach more experienced researchers and vice-versa.

After the symposium, written versions of the presentations and posters will be published in a proceedings book and in an online version. All participants will receive a copy of the ISAAR 2017 proceedings.

The organizing committee and the Danavox Jubilee Foundation wish you an interesting and fruitful symposium. Happy networking!

Wednesday 23 August

09:00-10:30	*Registration and hanging of posters*
10:45-11:00	Torsten Dau: Welcome and introduction to the symposium

Session 1: Adaptive behavior in complex listening environments

11:00-11:30	Owen Brimijoin: Parametric measurements of natural conversation behaviour reveal effects of background noise level on speech, movement, and gaze
11:30-12:00	Piotr Majdak: The role of listener-specific factors in binaural sound reproduction
12:00-13:30	*Lunch*
13:30-13:50	Karen Banai: Rapid adaptation to rapid speech: A detriment to speech perception in populations with speech-perception deficits?
13:50-14:10	Andrew Oxenham: Adaptation and context effects in auditory and speech perception: Effects of hearing loss and cochlear implants

Wednesday 23 August

Session 1: Adaptive behavior in complex listening environments (cont'd)

14:10-14:30	Jacques Grange: "Turn an ear to hear": CI users can use head orientation to improve their intelligibility of speech in noisy social settings
14:30-14:50	Andreu Paredes-Gallardo: The role of temporal cues on voluntary stream segregation in cochlear implant users
14:50-15:20	*Coffee break*

Session 2: Neural mechanisms and modeling of adaptive auditory processes

15:20-15:50	Jan Schnupp: What can we learn from statistical models of auditory cortex neurons?
15:50-16:20	Elia Formisano: Task-dependent encoding of real-life sounds in human auditory cortex
16:20-16:40	Birger Kollmeier: Aided patient performance prediction: Machine learning vs. auditory modelling?
16:40-17:00	Xin Zhou: Using fNIRS to study audio-visual speech integration in post-lingually deafened cochlear implant users
17:00-19:00	**Poster session I**
19:00-20:30	*Dinner*
20:30-23:00	*Drinks in the poster area*

Thursday 24 August

Session 2: Neural mechanisms and modeling of adaptive auditory processes (cont'd)

08:30-09:00	Mounya Elhilali: Speech processing using adaptive auditory receptive fields
09:00-09:20	Laurel Carney: Aural contrast and speech-on-speech masking: Model midbrain responses to simultaneous speech
09:20-09:40	*Coffee break*
09:40-10:10	Tobias Moser: Hearing the light: Optogenetic stimulation of the auditory pathway
10:10-10:30	Vani Rajendran: Midbrain adaptation may set the stage for the perception of musical beat
10:30-10:50	*Coffee break*

Session 3: "Maladaptive" processes in hearing

10:50-11:20	Pim Van Dijk: Tinnitus-related activity in cortical and sub-cortical brain areas
11:20-11:40	Marlin Johansson: Children with congenital unilateral sensorineural hearing loss: Effects of late hearing aid amplification
11:40-12:00	Stephan Ewert: Physiologically motivated binaural loudness model for normal hearing and hearing impaired
12:00-13:30	*Lunch*

Thursday 24 August

Session 3: “Maladaptive” processes in hearing (cont’d)

13:30-14:00	Roland Schaette: Adaptation deficits through hidden hearing loss reveal interplay between threshold and gain adaptation
14:00-14:30	Martin Pienkowski: Can long-term exposure to non-damaging noise lead to tinnitus and hyperacusis?
14:30-14:50	*Coffee break*

Session 4: Electrophysiological correlates of auditory adaptation

14:50-15:20	Tobias Overath: Electrophysiological correlates of acoustic and linguistic analyses of temporal speech structure
15:20-15:50	Stefan Debener: Intra-modal and cross-modal cortical reorganization in postlingually deaf cochlear implant users
15:50-16:10	*Coffee break*
16:10-16:40	Manuel Malmierca: Emergence of deviance detection along the auditory neuroaxis: The neuronal basis of predictive coding
16:40-17:00	Jonatan Märcher-Rørsted: Dynamics of cortical oscillations during an auditory N-back task
17:00-19:00	**Poster Session II**
19:00-20:30	*Dinner*
20:30-23:00	*Drinks in the poster area*

Friday 25 August

Session 5: Adaptive and learning processes with hearing devices

08:30-09:00	Monita Chatterjee: Processing of fundamental frequency changes, emotional prosody and lexical tones by pediatric CI recipients
09:00-09:30	Tom Francart: Neuro-steered adaptation of hearing aid and cochlear implant processing
09:30-09:50	*Coffee break*
09:50-10:10	Julie Arenberg: Adaptation to listener-tailor strategies in adult cochlear implant listeners
10:10-10:30	Lars Bramsløw: Data-driven hearing care with time stamped logging
10:30-10:50	Gabriel Gomez: A pinna-cues preserving beamforming algorithm for hearing aids
10:50-11:10	*Coffee break*
11:10-11:40	Andrew Sabin: Real World Self Adjustment of a Hearing Assistance Device
11:40-12:10	Nikolai Bisgaard: Adapting to hearing aid use to achieve full benefit
12:10-12:30	Torben Poulsen: Closing remarks
12:30-14:00	*Lunch and departure*

Venue and Travel Information

Venue

The symposium venue is Hotel Nyborg Strand, Østerøvej 2, 5800 Nyborg, Denmark. The hotel is situated in the middle of Denmark (GPS coordinates: Lat: N 55º 19' 5.74", Long: E 10º 48' 43.88"). The distance from Copenhagen Airport (CPH) is about 134 km, about 1½ hour by rail or road. For more information, visit www.nyborgstrand.dk. You may contact the hotel by phone (+45 65 31 31 31) or e-mail (nyborgstrand@nyborgstrand.dk).

Travel information

Air travel

The nearest airports are Copenhagen Airport "Kastrup Lufthavn" (CPH, see www.cph.dk) and Billund Airport (BLL, see www.bll.dk).

From Copenhagen or Billund airport to Nyborg by rail

From CPH airport you will find trains to Copenhagen Central Station (København H), where you can change for trains to Nyborg. There are also some direct trains between CPH airport and Nyborg. One-way standard fare: DKK 244 (approx. EUR 33, USD 35, fare may vary depending on ticket type). Duration: ca. 1h45m.

From BLL airport, take bus 34 or 134 to Vejle station, where you can change for trains to Nyborg. One-way standard fare: DKK 244 (bus: DKK 60 + train: DKK 184, fare may vary depending on ticket type). Duration: ca. 1h45m.

Please always check www.journeyplanner.dk for updated timetable information and use www.dsb.dk/en/ for online ticket reservations.

From Copenhagen or Billund airport to Nyborg by road

Road travel from CPH airport to Hotel Nyborg Strand takes about 1½ hour (134 km or 83 miles). Note a one-way toll charge of DKK 240 or EUR 34 per vehicle for crossing the Great Belt Bridge. Travel from BLL airport to Hotel Nyborg Strand takes about 1½ hour (135 km).

From Nyborg station to the hotel

Nyborg railway station is about a 5-minute drive from Hotel Nyborg Strand. Taxi: DKK 60 (approx. EUR 8, USD 9). If you like walking, there is a 15-minute “Nature Path” between the railway station and the hotel. Use www.journeyplanner.dk to assist your planning of local transportation.

Planning ahead

On planning your return, prepare 2 hours for transport to Copenhagen Airport and another 2 hours for check-in and security check at the airport. The scientific programme starts on August 23 at 10am and end on August 25 at 12:30pm. Please plan your journey accordingly.

About the weather

The weather in Denmark is unpredictable. Day temperatures between 15 and 25 degrees centigrade. Frequent showers and often windy. See www.dmi.dk for the current forecast.

Practical Information

Posters

Hanging of posters: Wed 23 Aug 09:00-10:30

Presenters of odd-numbered posters are encouraged to be present at their poster during the first dedicated poster session (Wed 17-19), presenters of even-numbered posters during the second dedicated poster session (Thu 17-19). Posters will remain on display throughout the symposium to allow further interaction outside these dedicated sessions.

Talks

Dedicated time with assistance for slide upload and technical tests in the auditorium:

Wed 23 Aug	10:00-10:30 and 17:00-17:15
Thu 24 Aug	17:00-17:15

A PC with PowerPoint software will be available in the auditorium.

Contributed oral presentations should not exceed 15 min. in length (25 min. for invited talks), in order to leave at least 5 min. after each talk for questions and discussion.

Meals and drinks

The ISAAR registration fee includes all meals and social activities during the symposium and a copy of the symposium proceedings. Two glasses of wine will be served free of charge at dinner. Complimentary beer, wine, and soft drinks will also be available in the evenings in the poster area. Other drinks may be purchased at the hotel bar.

Contact information

For any questions concerning the programme or manuscripts, please contact:
webmaster@isaar.eu

For registration or venue information, please contact Hotel Nyborg Strand directly at:
nyborgstrand@nyborgstrand.dk

For general information about ISAAR, or to contact the scientific committee, please write to:
isaar@isaar.eu

Manuscript Information

Manuscripts for ISAAR proceedings

Authors are encouraged to submit a manuscript for their ISAAR contribution. Manuscripts from both oral and poster presentations will be published in the proceedings book and distributed to all participants after the symposium. Proceedings will also be accessible to all participants at proceedings.isaar.eu as well as via the GN ReSound audiological library (www.audiological-library.gnresound.dk).

All manuscripts must be submitted electronically at www.isaar.eu. Authors are requested to follow the manuscript guidelines and to use the templates available at www.isaar.eu. Manuscripts are limited to a maximum length of 8 pages for contributed papers and 12 pages for invited papers.

The deadline for receipt of manuscripts is 15 September 2017.

Special issue of Trends in Hearing

Authors of accepted proceedings manuscripts will be given the opportunity to submit a full journal paper based on their ISAAR contribution to a special issue of open-access journal Trends in Hearing (see http://tia.sagepub.com/).

Trends in Hearing remains the only fully open-access journal to specialize in topics related to human hearing and hearing loss, and is currently the top-ranked journal in the area of Audiology and Speech-Language Pathology, based on impact factor.

All manuscripts should be submitted by 15 December 2017. Please see the journal website for online submission and guidelines.

When submitting the manuscript, please indicate in the cover letter that the manuscript is intended for the ISAAR special issue. Overlap with material in the ISAAR book manuscript is permitted. All manuscripts will undergo rigorous peer review and authors should receive an initial decision on their manuscript by February 2018. Articles will be published and added to the special issue shortly after acceptance.

A special discount on publication fees will be applied for submissions to this special issue (invited papers: covered by the Danavox Jubilee Foundation; contributed papers: $750; normal publication fee: $1,000). In cases where funds are not available to the authors, a fee waiver may be granted.

Session 1:

Adaptive behavior in complex listening environments

Chairs: Brian Moore & Ewen MacDonald

Wed 23 Aug, 11:00-14:50

S1.1 – Wed 23 Aug, 11:00-11:30

Parametric measurements of natural conversation behaviour reveal effects of background noise level on speech, movement, and gaze

W. Owen Brimijoin*, Lauren V. Hadley, Graham M. Naylor, William M. Whitmer *- MRC/CSO Institute of Hearing Research - Scottish Section, Glasgow, UK*

When conversing in a noisy place there are a number of ways that speakers and listeners can improve understandability. They can adjust how loudly they talk, they can move closer to their partner, or they can vary where they direct their head and gaze. Critically, these behaviors are likely to interact with the behavior of their conversation partner, the way their hearing aids operate, and the type and level of background noise. To assess this we measured speech, head movement, and eye movement of seated pairs and triplets of hearing impaired people (N=65) holding semi-structured conversations. In each conversation the background noise level was altered every 15-25 seconds. Results showed an effect of increasing noise level on all measured behaviors: participants spoke louder, moved closer together, and pointed their head and eyes toward their partner with less variability. These conversation strategies help but do not fully compensate for the changes in noise level. For example, the acoustic changes due to moving closer together were small (<1 dB), as were changes in speaking level (1-2 dB), and the sum of these did not offset the changes in noise level (6 dB steps). Although the observed behaviors may not fully counteract changes in noise, we suggest they are stereotypical enough to be exploited in current hearing rehabilitation and future multi-modal hearing devices.

Corresponding author: Owen Brimijoin (owen.brimijoin@nottingham.ac.uk)

S1.2 – CANCELLED

The timecourse of segregation in multitalker mixtures

Virginia Best* *- Department of Speech, Language and Hearing Sciences, Boston University, Boston, USA*

Speech is a temporal signal in which information is distributed across time, and it is well known that auditory scene analysis evolves over time. However, little is known about how speech perception changes over time in complex listening situations (e.g., when multiple talkers compete for attention). There are some examples in the literature suggesting that, in multitalker mixtures, speech intelligibility improves from word to word when some defining feature of the target talker (e.g., location, voice) remains fixed. The results are consistent with the idea that listeners refine their selectivity over time when stable cues are available. Here, the behavioral evidence for this "buildup" effect is reviewed, along with some relevant physiological data. In addition, some new work is presented that explores the timecourse of the buildup, and the conditions under which it occurs, in listeners with normal and impaired hearing.

Corresponding author: Virginia Best (ginbest@bu.edu)

S1.3 – Wed 23 Aug, 11:30-12:00

The role of listener-specific factors in binaural sound reproduction

Piotr Majdak* - *Acoustics Research Institute, Austrian Academy of Sciences, Wien, Austria*
Claudia Jenny - *Department of Musicology, University of Vienna, Wien, Austria; Acoustics Research Institute, Austrian Academy of Sciences, Wien, Austria*
Robert Baumgartner - *Acoustics Research Institute, Austrian Academy of Sciences, Wien, Austria*

Millions of people use headphones every day for listening to music, watching movies, or communicating with others. Nevertheless, sounds presented via headphones are usually perceived inside the head and not at their actual natural spatial position. This unnatural perception is an inherent limitation as it may unconsciously require more cognitive load and result in faster fatigue and feeling uncomfortable in social interactions. Under precisely controlled laboratory conditions of headphone listening, especially involving listener-specific head-related transfer functions (HRTFs), virtual sounds can be indistinguishable from natural sounds. Listener-specific HRTFs are required for a perfect auditory illusion but other factors like room reverberation, head and source movements, attention, perceptual adaptation, or input from other modalities like vision and proprioception may interact with the need for such HRTFs. In this contribution, we discuss recent findings and open research questions related to the degree of realism achieved by sound reproduction via headphones with a particular focus on the role of listener-specific HRTFs.

Corresponding author: Piotr Majdak (piotr@majdak.com)

S1.4 – Wed 23 Aug, 13:30-13:50

Rapid adaptation to rapid speech: A detriment to speech perception in populations with speech-perception deficits?

Karen Banai*, Limor Lavie - *Communication Sciences and Disorders, University of Haifa, Haifa, Israel*

Rapid perceptual learning plays a role in adapting to different forms of hard-to-perceive (e.g., rapid, accented) speech in the general population. Whether this learning is impaired in clinical populations with speech perception difficulties is unknown. In this talk data from a series of studies on the perceptual learning of time-compressed speech will be presented. These suggest that older adults (ONH), older adults with age-related hearing loss (OHI) and nonnative listeners (NN) which are known to have difficulties in the perception of rapid speech, also benefit to a lesser extent from the opportunity to rapidly adapt to such speech as well as from more extended training. Furthermore, in normal-hearing young adults as well as in ONH and OHI adults, rapid learning of time-compressed speech was strongly correlated with the perception of naturally-fast speech even after controlling for initial performance. It thus seems that perceptual deficits due to either predominantly sensory factors (OHI listeners), or due to non-sensory factors (NN listeners) interfere with the adaptation to the rapidly changing auditory circumstances associates with ecological listening conditions. This should be considered when exploring perceptual training for hearing rehabilitation because there is currently little data to suggest that such training contributes to the rapid learning of speech.

Corresponding author: Karen Banai (kbanai@research.haifa.ac.il)

S1.5 – Wed 23 Aug, 13:50-14:10

Adaptation and context effects in auditory and speech perception: Effects of hearing loss and cochlear implants

Andrew Oxenham*, Lei Feng - *Department of Psychology, University of Minnesota, Minneapolis, MN, USA*

Auditory enhancement and speech context effects are both examples of adaptive mechanisms in auditory and speech perception. They both play important roles in perception, assisting in the adaptation to different talkers, different room environments, and different background noises. Although, they have generally been studied as separate phenomena, it is possible that they both reflect the same underlying auditory mechanisms. Here we review recent studies from our lab that have attempted to elucidate the mechanisms underlying auditory enhancement and speech context effects in people with normal and impaired hearing and with cochlear implants. The data suggest that similar mechanisms may underlie both effects and demonstrate interesting similarities and differences between the three populations. Differences between the groups can be used to design signal-processing algorithms that can restore adaptation and context effects in clinical populations. [Supported by NIH grant R01 DC012262.]

Corresponding author: Andrew Oxenham (oxenham@umn.edu)

S1.6 – Wed 23 Aug, 14:10-14:30

“Turn an ear to hear”: CI users can use head orientation to improve their intelligibility of speech in noisy social settings

Jacques A. Grange*, John F. Culling *- Cardiff University, Cardiff, UK*
Sarah Hughes *- Bridgend PoW Hospital, ABM-ULHB-CI program, Bridgend, UK*
Steven Backhouse *- Bridgend PoW Hospital, ABM-ULHB-ENT, UK*
Laura Mackinney *- Cardiff University, Cardiff, UK*

Cochlear implant (CI) users struggle to follow conversations in noisy settings. Surveys of 98 CI users and 37 clinical professionals, as well as web-based listening strategy advice, showed that 90% of the time, CI users are advised to face a conversation partner. Yet, testing CI users in a sound-treated room for head-orientation benefit (HOB) to speech intelligibility in noise, Grange and Culling (2016) found that CI users reached a HOB at 30° as high (5 dB) as normal-hearing listeners' (NHs), with no detrimental effect on lip-reading. Attending to a talker in noise, CI users spontaneously kept facing the talker, but could follow the speech to much lower SNRs when they used their HOB. The real-restaurant, virtual simulation from Grange & Culling was combined with an advanced CI simulation (SPIRAL vocoder) to test for CI users' HOB in a real-life setting. Speech reception thresholds (SRTs) in noise were 13 dB higher than for NHs. SRTs with interfering voices were higher again by 5 dB, as SPIRAL removed access to the cues NHs can use to compensate for 'informational' masking. 30° HOBs went from 1.6 dB in noise (same as NHs') to 2.8 dB with interfering voices (interaction not found with NHs), suggesting that modulation masking by interfering voices varies faster with signal-to-noise ratio at the better ear than does energetic masking by steady-state noises. The listening advice given to CI users should reflect our HOB findings.

Corresponding author: Jacques Grange (grangeja@cardiff.ac.uk)

S1.7 – Wed 23 Aug, 14:30-14:50

The role of temporal cues on voluntary stream segregation in cochlear implant users

Andreu Paredes-Gallardo*, Sara M. K. Madsen, Torsten Dau, Jeremy Marozeau *- Hearing Systems, Department of Electrical Engineering, Technical University of Denmark, Kgs. Lyngby, Denmark*

Most of the previous research investigating segregation abilities of cochlear implant (CI) users assessed the temporal coherence boundary, the largest difference between sounds at which they still can be integrated. Little attention has been given to the fission boundary (FB), the smallest difference between sounds needed to segregate them. The present study investigates the FB as a function of temporal cues in CI users and aims to establish whether a two-stream percept occurs instantaneously or needs time to build up. CI users participated in a detection task composed of a sequence of regularly presented bursts of pulses ("A") on a single electrode interleaved with an irregular sequence ("B") presented on the same electrode with a different pulse rate. The pulse rate difference and the duration of the sequences were varied between trials. In half of the trials, a delay was added to the last burst of the regular A sequence and the listeners were asked to detect this delay. As the period between consecutive B bursts was jittered, time judgments between the A and B sequences provided an unreliable cue to perform the task. Thus, segregation of A and B should improve performance. Results showed that performance increased with rate differences and sequence length, suggesting that CI users can rely on temporal cues to segregate sounds and that this percept builds up over time.

Corresponding author: Andreu Paredes Gallardo (apaga@elektro.dtu.dk)

Session 2:

Neural mechanisms and modeling of adaptive auditory processes

Chairs: Jens Hjortkjær & Jakob Christensen-Dalsgaard

Wed 23 Aug, 15:20-17:00
Thu 24 Aug, 08:30-10:30

S2.1 – Wed 23 Aug, 15:20-15:50

What can we learn from statistical models of auditory cortex neurons?

Jan W. H. Schnupp* *- City University of Hong Kong, Hong Kong*
Nicol S. Harper, Benjamin Willmore *- University of Oxford, Oxford, UK*
Neil Rabinowitz *- Google Deep Mind, London, UK*
Andrew King *- University of Oxford, Oxford, UK*

Understanding the input-output transformations of neurons in the auditory system is perhaps the key problem in auditory neuroscience. Given the stochastic nature of neural discharges, this problem is usually framed as a statistical one: one attempts to predict the firing probability, or equivalently, the average firing rate, for arbitrary inputs. Reverse correlation methods have been applied to this problem as long ago as the 1960s, but they "measure only the linear part" of a neuron's input-output function, and given the well known non-linearities of the auditory system, this type of approach was thought to be unlikely to be useful for cortex until the late 90s, when spectro-temporal receptive field models (STRFs) of cortical neurons first emerged. Recent years have seen useful extensions to a purely linear framework with so called linear-nonlinear (LN), or even non-linear-linear-nonlinear (NLN) models. In my talk I will use examples from work that I have been involved in over the last 20 years to illustrate both the strengths and the limitations of STRF approaches, briefly discuss conceptual and methodological background, and I will provide examples of how STRF and derived models have contributed to our understanding of such diverse topics as binaural spatial processing or adaptation to changing auditory contrast or to background noise in auditory cortex.

Corresponding author: Jan Schnupp (wschnupp@cityu.edu.hk)

S2.2 – Wed 23 Aug, 15:50-16:20

Task-dependent encoding of real-life sounds in human auditory cortex

Elia Formisano* *- Maastricht Brain Imaging Center, Maastricht University, Maastricht, The Netherlands*

Electrophysiological studies in animals have established that the spectro-temporal receptive fields of neurons in auditory cortex are optimized to process behaviorally relevant sounds. Furthermore, it has been established that neurons can rapidly adapt their spectro-temporal receptive fields to meet the demands of a current behavioral task. These changes are specific to the task-relevant acoustic features and lead to enhanced processing of that attended features and suppressed processing of unattended features. Here I will present research that combines high resolution fMRI with computational modelling to investigate these neural processing mechanisms in human auditory cortex. First, I will describe how the embedding of sound representation models in the fMRI data analysis enables assessing the encoding of real-life sounds in primary and non-primary auditory areas. Second, I will present new experimental evidence showing that this encoding is highly adaptive and task-dependent. In fact, listening to the same sounds (or scenes) while performing different tasks that rely on distinctive sets of acoustic features results in different cortical representations, where the task-relevant features are significantly enhanced. This work provides insights into the neuro-computational mechanisms enabling the efficient processing of sounds in a rich and dynamic auditory environment.

Corresponding author: Elia Formisano (e.formisano@maastrichtuniversity.nl)

S2.3 – Wed 23 Aug, 16:20-16:40

Aided patient performance prediction: Machine learning vs. auditory modelling?

Birger Kollmeier*, Anna Warzybok, David Hülsmeier, Constantin Spille, Stephan Ewert, Bernd T. Meyer, Marc-René Schädler *- Medizinische Physik & Cluster of Excellence Hearing4All, Universität Oldenburg, Oldenburg, Germany*

Several attempts are reviewed to predict speech recognition performance of hearing-impaired listeners in noise with and without a hearing aid using the audiogram and suprathreshold performance data. While the “classical” auditory-model-based approach has the advantage of relating to several auditory functions but makes strong assumptions about the underlying detection process (such as, e.g., an “optimum detector”), machine learning tools employed in automatic speech recognition (ASR) make much less strict assumptions and better exploit the statistical properties of the presented input signals. Here the Framework for Auditory Discrimination Experiments (FADE, Schädler et al., JASA, 2016) is employed for predicting patient performance employing the German Matrix sentence test (available for 16 major languages, see Kollmeier et al., Int. J. Audiol., 2015). It is compared with a DNN-based ASR system utilizing an open-set sentence recognition test. FADE can well predict the average individual performance with different (binaural) noise reduction algorithms using a cafeteria noise in comparison to individual empirical data from Völker et al. (2015) with R^2 of about 0.9. The results of a follow-up experiment with a set of 19 hearing-impaired listeners shows a reduction of the average prediction error down to 4.6 dB if individual suprathreshold processing properties are accounted for.

Corresponding author: Birger Kollmeier (birger.kollmeier@uni-oldenburg.de)

S2.4 – Wed 23 Aug, 16:40-17:00

Using fNIRS to study audio-visual speech integration in post-lingually deafened cochlear implant users

Xin Zhou*[,S], Hamish Innes-Brown, Colette M. McKay *- The Bionics Institute of Australia, Department of Medical Bionics, University of Melbourne, Melbourne, Australia*

The aim of this experiment was to investigate differences in audiovisual (AV) speech integration between cochlear implant (CI) users and normally hearing (NH) listeners using behavioral measures and near infrared spectroscopy (fNIRS). Participants were 14 post-lingually deafened adult CI users and 13 age-matched NH listeners. Participants' response accuracy in audio-alone (A), visual-alone (V) and AV modalities were measured for consonant perception with close-set /aCa/ non-words and for open-set CNC words. AV integration was estimated as the difference between the actual AV score and the score predicted from a probabilistic model assuming no integration. Using fNIRS, brain activation was measured when listening to or watching A, V, or AV speech with or without multi-speaker babble. For fNIRS, evidence of AV integration was measured using the "inverse effectiveness" model (comparing the increase in activation in each measurement channel from A to AV listening in quiet and noise conditions). Behavioral AV integration was similar in the two groups for CNC words but poorer in the CI group compared to NH group for consonant perception. From fNIRS data, brain regions were identified where activity levels were consistent with the "inverse effectiveness" model. Results showed that CI users were recruiting occipital areas for AV integration more strongly than NH listeners.

Corresponding author: Xin Zhou (xzhou@bionicsinstitute.org)

S2.5 – Thu 24 Aug, 08:30-09:00

Speech processing using adaptive auditory receptive fields

Mounya Elhilali*, Aswhin Bellur *- Johns Hopkins University, Whiting School of Engineering, Baltimore, MD, USA*

The auditory system exhibits a remarkable ability to adapt to its listening environment, driven both by sensory-based cues and goal-directed processes. Here, we focus on the role of attentional feedback in facilitating processing of speech sounds in presence of nonstationary noises. We examine a theoretical formulation for retuning of cortical-like receptive fields to enable robust detection of speech sounds in presence of interference. The framework employs 2-D Gabor filters whose parameters are retuned based on goal-directed feedback to enhance separability between the feature representation of speech and nonspeech sounds. This retuning process, driven by feedback from statistical models of speech and nonspeech classes, attempts to minimize the misclassification risk of mismatched data, with respect to the original statistical models. We hypothesize that this risk minimization procedure results in an emphasis of unique speech and nonspeech modulations in a high- dimensional space. We show that such an adapted system is indeed robust to other novel conditions, with a marked reduction in equal error rates for a variety of databases with additive and convolutive noise distortions. We discuss the lessons learned from biology with regard to adapting to an ever-changing acoustic environment and the impact on building truly intelligent audio processing systems.

Corresponding author: Mounya Elhilali (mounya@jhu.edu)

S2.6 – Thu 24 Aug, 09:00-09:20

Aural contrast and speech-on-speech masking: Model midbrain responses to simultaneous speech

Laurel H. Carney* *- Biomedical Engineering & Neuroscience, University of Rochester, Rochester, NY, USA*

At the level of the auditory midbrain, low-frequency fluctuations within each frequency channel drive neurons with band-pass modulation transfer functions (MTFs). The amplitude of low-frequency fluctuations in ascending neural signals is affected by stimulus amplitude due to the gradual saturation of the inner hair cells (IHCs) beginning at moderate sound levels. This level-dependence of low-frequency fluctuation amplitudes results in aural contrast cues at the level of the midbrain: spectral peaks result in lower responses of cells with bandpass-MTFs, whereas spectral valleys result in higher responses. Here, we focus on model population midbrain responses with different best-modulation frequencies (BMFs) to simultaneous speech. Midbrain responses were simulated for single HINT sentences and for simultaneous sentences, spoken by a male and a female. As expected, correlations between population responses to individual male (or female) sentences and responses to simultaneous sentences are strongest for models with BMFs in the range of the male (or female) fundamental frequencies. The pattern of aural contrast in the midbrain representation provides a framework for studying speech-on-speech masking for listeners with normal hearing and sensorineural hearing loss, as well as effects of adaptive processes at different levels of the auditory pathway on aural contrast cues.

Corresponding author: Laurel Carney (laurel.carney@rochester.edu)

S2.7 – Thu 24 Aug, 09:40-10:10

Hearing the light: Optogenetic stimulation of the auditory pathway

Tobias Moser* *- The Göttingen Cochlear Optogenetics Program, University Medical Center Göttingen, Göttingen, Germany; German Primate Center, Göttingen, Germany; Max-Planck-Institute, Göttingen, Germany*

Sound coding with current cochlear implant (CI), based on electrical stimulation of auditory neurons, has limited frequency resolution due to broad current spread. We aim to improve frequency and intensity resolution of CI coding by establishing spatially confined optical stimulation of spiral ganglion neurons (SGNs). We have established optogenetic stimulation of the auditory pathway in rodents using virus-mediated expression of channelrhodopsins to render SGNs light-sensitive. Optogenetic stimulation of spiral ganglion neurons activated the auditory pathway, as demonstrated by recordings of single neuron and neuronal population responses at various stages of the auditory system. Fast opsins enabled SGN firing at near physiological rates (hundreds per second). We approximated the spatial spread of cochlear excitation by recording local field potentials in the inferior colliculus in response to suprathreshold optical and electrical stimuli, which suggested a better frequency resolution for optogenetic than for electrical stimulation. Towards characterizing the percept induced by cochlear optogenetics we studied activation of neurons in primary auditory cortex and performed a behavioral response in virus-injected gerbils. In summary, optogenetic stimulation of the auditory nerve is feasible and bears substantial potential for improved hearing restoration by CIs.

Corresponding author: Tobias Moser (tmoser@gwdg.de)

S2.8 – Thu 24 Aug, 10:10-10:30

Midbrain adaptation may set the stage for the perception of musical beat

Vani G. Rajendran* *- University of Oxford, Oxford, UK*
Jose A. Garcia-Lazaro *- UCL Ear Institute, London, UK*
Nicol S. Harper *- University of Oxford, Oxford, UK*
Nick A. Lesica *- UCL Ear Institute, London, UK*
Jan W. H. Schnupp *- City University of Hong Kong, Hong Kong*

The ability to spontaneously feel a beat in music is a curious phenomenon widely believed to be unique to humans. Though beat perception involves the coordinated engagement of sensory, motor, and cognitive processes in humans, the contribution of low-level auditory processing to the activation of these networks in a beat-specific manner is poorly understood. Here, we present evidence from a rodent model that midbrain pre-processing of sounds may already be shaping where the beat is ultimately felt. For simple musical rhythms, on-beat sounds on average evoked higher firing rates than off-beat sounds, and this difference was a defining feature of the set of beat interpretations most commonly perceived by listeners over others. Basic firing rate adaptation provided a sufficient explanation for these results. Our findings suggest that midbrain adaptation, by encoding the temporal context of incoming sounds, creates points of neural emphasis that may influence the perceptual emergence of musical beat.

Corresponding author: Jan Schnupp (wschnupp@cityu.edu.hk)

Session 3:

“Maladaptive” processes in hearing

Chairs: Inga Holube & Torsten Dau

Thu 24 Aug, 10:50-14:30

S3.1 – Thu 24 Aug, 10:50-11:20

Tinnitus-related activity in cortical and sub-cortical brain areas

Pim van Dijk*, Elouise A. Koops, Cris P. Lanting, Emile de Kleine *- Department of Otorhinolaryngology, Head & Neck Surgery, University Medical Center Groningen, University of Groningen, Groningen, The Netherlands*

Tinnitus is a phantom sound percept that mainly occurs in combination with peripheral hearing loss. Studies in humans and lab animals have shown that hearing loss correlates with abnormal brain activity, which presumably corresponds to tinnitus. Functional imaging studies in humans identified changes in the central auditory system and beyond. Tinnitus corresponds to increased amplitude of responses in auditory cortex, but probably not in the brainstem. However, in response to sound, the correlation between brainstem and auditory cortex activity is reduced in tinnitus patients. In a group of subjects with gaze-modulated tinnitus, an increase of tinnitus loudness corresponded to an increase of activity in the brainstem and auditory cortex, and to inhibition of thalamus activity. These findings have been attributed to either a change in the thalamo-cortical interactions or a change in thalamic gating in tinnitus. Studies of cortical networks showed that the default mode network is less extensively activated during rest in tinnitus patients, presumably due to the saliency of the tinnitus. Further studies have shown that the activity and extent of cortical networks correlates with tinnitus handicap. Together, these results suggest that tinnitus corresponds to changes in cortical and subcortical brain areas, where the subcortical areas presumably correspond to the tinnitus percept itself, while cortical networks reflect the reactions to tinnitus and hence reflect the severity of the handicap caused by tinnitus.

Corresponding author: Pim van Dijk (p.van.dijk@umcg.nl)

S3.2 – Thu 24 Aug, 11:20-11:40

Children with congenital unilateral sensorineural hearing loss: Effects of late hearing aid amplification

Marlin Johansson,S* - *Department of Clinical Science, Intervention and Technology, Karolinska Institutet, Stockholm, Sweden*

Filip Asp - *Department of ENT, Section of Cochlear Implants, Karolinska University Hospital, Stockholm, Sweden; 4Department of Signals and Systems, Chalmers University of Technology, Gothenburg, Sweden; Department of Clinical Science, Intervention and Technology, Karolinska Institutet, Stockholm, Sweden*

Erik Berninger - *Department of Audiology, Karolinska University Hospital, Stockholm, Sweden; Department of Clinical Science, Intervention and Technology, Karolinska Institutet, Stockholm, Sweden*

Maladaptive neural reorganization might be avoided with timely unrestricted aural input along both auditory pathways. We investigate whether children with congenital unilateral sensorineural hearing loss (9.7-10.8 yrs of age; n=6) benefit from late introduction of hearing aid amplification (after 4.8-8.9 yrs of age). Electrophysiological and psychoacoustic measurements and questionnaires were used. The mean pure tone average was 45 dB HL and 6 dB HL in the impaired and normal ear, respectively (n=6). Subjective aided benefit in ease of communication was found, but no aided improvement in two psychoacoustic tests simulating demanding listening situations. Sound localization accuracy (objective eye-tracking test) was poorer in the aided compared to unaided condition ($p<0.05$). A distinct correlation between aided sound localization accuracy and the auditory brainstem response interpeak I-V interval from the worse ear was also found ($r=0.98$; $p<0.05$). This may indicate that different degrees of adaptation to the asymmetric condition had occurred, resulting in different degrees of disrupted localization accuracy when audibility was restored to the impaired side. In sum, hearing aids fitted late can disrupt binaural abilities, such as localization of sound source, but help the child with congenital unilateral sensorineural hearing loss in everyday communication.

Corresponding author: Marlin Johansson (marlin.johansson@ki.se)

S3.3 – Thu 24 Aug, 11:40-12:00

Physiologically motivated binaural loudness model for normal hearing and hearing impaired

Iko Pieper, Manfred Mauermann *- Medizinische Physik and Cluster of Excellence Hearing4All, Universität Oldenburg, Oldenburg, Germany*
Dirk Oetting *- Project Group Hearing, Speech and Audio Technology of the Fraunhofer IDMT and Cluster of Excellence Hearing4all, Oldenburg, Germany*
Birger Kollmeier, Stephan Ewert* *- Medizinische Physik and Cluster of Excellence Hearing4All, Universität Oldenburg, Oldenburg, Germany*

A steeper progression of loudness as a function of stimulus level (loudness recruitment) is one consequence of sensorineural hearing loss. Existing loudness models aim to explain altered loudness functions effectively using an attenuation and compression component. Binaural loudness is typically derived by a simple weighted summation of monaural loudness. However, Oetting et al. (Hear. Res. 335, 2016) demonstrated strong individual differences in binaural loudness perception of hearing impaired (HI) which might indicate adaptive binaural processes. Here the physiologically motivated loudness model of Pieper et al. (J. Acoust. Soc. Am. 139, 2016) was extended to help distinguishing the role of peripheral factors and higher stages of auditory processing on monaural and binaural loudness perception. Individual hearing thresholds were simulated by cochlear gain reduction and linear attenuation (damage of inner hair cells). Hearing threshold and cochlear gain loss were estimated from individual monaural and binaural loudness scaling data for narrow band noise. While existing loudness models fail to predict individual loudness functions for HI, the current model accounted for individual loudness functions in HI and normal hearing using a linear gain above threshold (referred to as post gain). It was investigated how the post gain is adapted in binaural conditions.

Corresponding author: Stephan Ewert (stephan.ewert@uni-oldenburg.de)

S3.4 – Thu 24 Aug, 13:30-14:00

Adaptation deficits through hidden hearing loss reveal interplay between threshold and gain adaptation

Warren Bakay *- University of Manchester, Manchester, UK*
Lucy Anderson, Jose Garcia-Lazaro *- UCL Ear Institute, London, UK*
David McAlpine *- Macquarie University, Sydney, Australia*
Roland Schaette* *- UCL Ear Institute, London, UK*

Auditory neurons adapt their responses to changes in the prevailing acoustic environment. In recordings from the mouse auditory midbrain, we demonstrate an interplay of two different kinds of adaptation, threshold and gain adaptation. For dynamically varying auditory environments containing high-probability regions of sound intensities centred on 44, 56, 68, and 80 dB SPL, threshold adaptation was able to adjust neural responses up to the loudest environment in control mice, but it was strongly reduced in mice subjected to mild noise exposure. In contrast, gain adaptation was unaffected, keeping neural activity constant across stimuli with different mean sound intensities in both groups. However, due to threshold adaptation deficits, gain adaptation substantially decreased neural gain in noise-exposed mice, substantially reducing neural information about variations in sound level for loud environments. These findings reveal a cascade of adaptation mechanisms, with gain adaptation downstream of threshold adaptation, and show that central mechanism might even aggravate deficits caused by damage to the periphery.

Corresponding author: Roland Schaette (r.schaette@ucl.ac.uk)

S3.5 – Thu 24 Aug, 14:00-14:30

Can long-term exposure to non-damaging noise lead to tinnitus and hyperacusis?

Martin Pienkowski* *- Osborne College of Audiology, Salus University, Elkins Park, PA, USA*

Hearing loss triggers changes in the central auditory system. A region of primary auditory cortex (A1) deprived of input responds more strongly and synchronously to unaffected lesion-edge frequencies, and its spontaneous firing (SF) increases. These changes have been correlated with tinnitus and hyperacusis, but models remain speculative. Reorganization of A1 tonotopic maps and regional increases in SF and synchrony are also observed after long-term exposure to non-damaging noise (NN). Adult cats exposed to bands of NN had suppressed evoked activity and SF in the A1 region mapped to the noise band, but enhanced activity and SF in A1 regions above and below the band. We hypothesized that frequencies within the band should be perceived as softer than normal (hypoacusis) and frequencies outside the band as louder (hyperacusis). We also wondered, given the increases in SF, whether NN could lead to tinnitus. Both short-term auditory deprivation and brief exposure to loud sound can lead to transient tinnitus, so perhaps NN-induced hyperactivity can as well. I will present the results of acoustic startle-based tests for hypo/hyperacusis and tinnitus after long-term exposure to 8-16 kHz noise at 70 dB SPL in CBA/Ca mice (interestingly, 75 dB SPL did cause synaptopathic damage). Testing mice without any hearing loss avoided a major confound of startle-based methods for detecting tinnitus and hyperacusis.

Corresponding author: Martin Pienkowski (mpienkowski@salus.edu)

Session 4:

Electrophysiological correlates of auditory adaptation

Chairs: Laurel Carney & Olaf Strelcyk

Thu 24 Aug, 14:50-17:00

S4.1 – Thu 24 Aug, 14:50-15:20

Electrophysiological correlates of acoustic and linguistic analyses of temporal speech structure

Tobias Overath* *- Duke Institute for Brain Sciences, Duke University, Durham, NC, USA*

Speech perception entails the mapping of the acoustic waveform to its linguistic representation. For this transformation to succeed, the speech signal needs to be tracked across a large temporal range in order to decode linguistic units from phonemes to sentences. I will present data on how cortical processing of temporal speech structure is modulated by higher-order linguistic analysis, which is conceived here as a form of adaptation to acquired statistical rules. To obtain control over the temporal scale of analysis, we use a novel sound-quilting algorithm that controls acoustic structure at different temporal scales (Overath et al., 2015). To obtain control over the linguistic content, independent of the temporal acoustic structure, we construct speech quilts from both familiar (English) and foreign (Korean) languages. This approach ensures that any changes at the signal-acoustics level affect both languages identically, while manipulating the linguistic percept differently. Thus, neural responses that vary as a function of segment length but are shared across languages suggest analysis at the signal-acoustics level, whereas neural responses that differ based on language familiarity imply the presence of linguistic processing. The results suggest neural entrainment in the theta and delta bands differentially encodes acoustic and linguistic speech structure, respectively.

Corresponding author: Tobias Overath (t.overath@duke.edu)

S4.2 – Thu 24 Aug, 15:20-15:50

Intra-modal and cross-modal cortical reorganization in postlingually deaf cochlear implant users

Stefan Debener* - *Department of Psychology, University of Oldenburg, Oldenburg, Germany; Cluster of Excellence Hearing4All, University of Oldenburg, Oldenburg, Germany*

In sensory and motor systems, the lack of experience or use results in a shrinkage of the corresponding cortical representations. Shrinkage often goes along with a redistribution of the remaining representations and the invasion of abandoned regions by the remaining modalities. A fundamental question in the field of neurorehabilitation is how system deprivation-induced cortical reorganization impacts on the capacity of the adult brain to adapt to restored input. This interaction can be studied in deaf adults receiving cochlear implants (CI). Different types of adaptation can be distinguished here, adaptation to extended periods of deafness and adaptation to restored (albeit artificial) sensory input as provided by the CI. Unfortunately little is known about how deprivation-adaptation predicts restoration-adaptation. We use high-density electroencephalography (EEG) and functional near infrared spectroscopy (fNIRS) to capture the dynamics of cortical reorganization in CI users, with the overall aim to identify maladaptive and adaptive patterns of cortical reorganization. I will summarize cross-sectional and longitudinal studies that have identified intra-modal and cross-modal cortical adaptation in CI users, and discuss the methodological and conceptual challenges that have to be overcome to better understand how brain reorganization can promote hearing with a CI.

Corresponding author: Stefan Debener (stefan.debener@uni-oldenburg.de)

S4.3 – Thu 24 Aug, 16:10-16:40

Emergence of deviance detection along the auditory neuroaxis: The neuronal basis of predictive coding

Manuel S. Malmierca*, Javier Nieto-Diego, Gloria G. Parras, Guillermo V. Carbajal, Catalina Valdes-Baizabal *- Auditory Neuroscience Laboratory, Institute of Neuroscience of Castilla y León (INCYL), University of Salamanca, Spain*

Carles Escera *- Brainlab-Cognitive Neuroscience Research Group, University of Barcelona, Spain*

Stimulus-specific adaptation (SSA) is the reduction in the responses to a common sound relative to the same sound when rare. It was originally described in the primary auditory cortex (A1) as the neuronal correlate of the mismatch negativity (MMN). However, the relationship between SSA and the MMN is still a subject of debate. The MMN is a mid-late potential (~150-200 ms in humans), and its neural sources have been located mainly within non-primary auditory cortex in humans and animal models. Moreover, SSA is also present as early as in the auditory midbrain and thalamus (IC and MGB). In this talk, we will show our recent findings from the IC, MGB and auditory cortex (AC) to unravel a hierarchical emergence of prediction error signals along the central auditory system. These error signals are detectable already at subcortical levels and correlated with large-scale mismatch responses in auditory cortex. Thus we demonstrate that deviance detection can be tracked down to the neuronal level and highlight the role of subcortical structures in cognition. Our results unify three coexisting views of perceptual deviance detection at different levels of description: neuronal physiology, cognitive neuroscience and the theoretical predictive coding framework. Financial support was provided by the Spanish MINECO (BFU2013-43608-P).

Corresponding author: Manuel Malmierca (msm@usal.es)

S4.4 – Thu 24 Aug, 16:40-17:00

Dynamics of cortical oscillations during an auditory N-back task

Jonatan Märcher-Rørsted*, Søren Asp Fuglsang[S], Torsten Dau *- Hearing Systems, Department of Electrical Engineering, Technical University of Denmark, Kgs. Lyngby, Denmark*

Jens Hjortkjær *- Hearing Systems, Department of Electrical Engineering, Technical University of Denmark, Kgs. Lyngby, Denmark; Danish Research Centre for Magnetic Resonance, Copenhagen University Hospital Hvidovre, Hvidovre, Denmark*

The N-back working memory task is a widely used experimental paradigm for probing the neural correlates of working memory load. To date, it remains largely unknown whether working memory processes interact with the way auditory stimuli are represented in human cortical activity, and how the dynamics of load-driven cortical activity evolve over time. Here, we recorded 64-channel scalp EEG from 15 subjects listening to 50-s long sequences of spoken digits while performing a continuous N-back task on the speech stimulus. The subjects were asked to report if each digit in the sequence was the same as the digit that appeared N digits earlier. We show that burdening working memory load for higher N caused and initial enhancement of parietal alpha power consistent with N-back tasks for visual stimuli and suggesting a general working memory mechanism. However, when tracked over longer periods time we also found considerable variations in alpha power which were related to behavioral performance. These results suggest that oscillatory power changes associated with persistent working memory demands may have a more complex dynamic than what is revealed when considering short task-evoked responses.

Corresponding author: Jonatan Märcher-Rørsted (jonmarc@elektro.dtu.dk)

Session 5:

Adaptive and learning processes with hearing devices

Chair: Jeremy Marozeau & Tobias Neher

Fri 25 Aug, 08:30-12:10

S5.1 – Fri 25 Aug, 08:30-09:00

Processing of fundamental frequency changes, emotional prosody and lexical tones by pediatric CI recipients

Monita Chatterjee* *- Auditory Prostheses & Perception Lab, Center for Hearing Research, Boys Town National Research Hospital, Omaha, NE, USA*

Mickael L. D. Deroche *- Center for Research on Brain, Language & Music, McGill University, Montreal, QC, Canada*

Shu-Chen Peng *- Center for Devices & Radiological Health, United States Food & Drug Administration, Silver Spring, MD, USA*

Hui-Ping Lu *- Dept. of Otolaryngology-Head & Neck Surgery, Chi-Mei Medical Center, Tainan, Taiwan*

Nelson Lu *- Center for Devices & Radiological Health, United States Food & Drug Administration, Silver Spring, MD, USA*

Yung-Song Lin *- Dept of Otolaryngology-Head & Neck Surgery, Chi-Mei Medical Center, Tainan, Taiwan; Taipei Medical University, Taipei, Taiwan*

Charles Limb *- Dept. of Otolaryngology-Head & Neck Surgery, University of California San Francisco School of Medicine, San Francisco, CA, USA*

As cochlear implants (CIs) do not provide an adequate representation of the harmonic structure of complex sounds, the perception of the voice fundamental frequency (F0) is severely limited in CI users. As F0 plays an important role in speech prosody and in lexical tones, this deficit has a negative impact on communication. This presentation will focus on the pediatric CI population, most of whom were prelingually deaf and were implanted before three years of age, within the most adaptive period of the brain's development. Our results suggest that, relative to their normally-hearing peers, school-age children with CIs have significant deficits in their sensitivity to F0-changes. In addition, children with CIs also have deficits in their identification of emotional prosody and in lexical-tone recogniton. As children develop in age and experience, performance improves on all tasks. We will review these recent findings in relation to the relevant acoustic cues available to CI recipients, and discuss implications for future designs of clinical intervention and device development.

Corresponding author: Monita Chatterjee (monita.chatterjee@boystown.org)

S5.2 – Fri 25 Aug, 09:00-09:30

Neuro-steered adaptation of hearing aid and cochlear implant processing

Tom Francart*, Jonas Vanthornhout *- KU Leuven - University of Leuven, Department of Neurosciences, ExpORL, Leuven, Belgium*
Neetha Das *- KU Leuven - University of Leuven, Department of Neurosciences, ExpORL; Department of Electrical Engineering, STADIUS, Leuven, Belgium*
Ben Somers, Lien Decruy, Eline Verschueren, Jan Wouters - *KU Leuven - University of Leuven, Department of Neurosciences, ExpORL, Leuven, Belgium*
Alexander Bertrand *- KU Leuven - University of Leuven, Department of Electrical Engineering, STADIUS, Leuven, Belgium*

The normal auditory system uses a number of efferent feedback mechanisms. When aiding or even replacing the auditory periphery with hearing aids or cochlear implants, some of these mechanisms are disabled. With the advent of wearable EEG recording systems, it is becoming technically possible to integrate similar feedback mechanisms in the devices, leading to closed-loop auditory prostheses. One approach is to inform a noise suppression algorithm of the attended speaker from a mixture. In this case, auditory attention detection is performed by measuring to which of the speech signals in the mixture there is most neural entrainment. We review a number of approaches to do this without access to the clean speech signals, and show that it is possible to get high accuracies at ecologically relevant signal to noise ratios. Another approach is to adapt the fitting of auditory prostheses in real-time according to the listener and listening environment. In this case, an EEG-based measure of hearing acuity is needed to serve as an input to a fitting parameter optimisation algorithm. We propose a measure based on neural coding of the speech envelope, and show that it is well-correlated with behaviourally measured speech intelligibility.

Corresponding author: Tom Francart (tom.francart@med.kuleuven.be)

S5.3 – Fri 25 Aug, 09:50-10:10

Adaptation to listener-tailor strategies in adult cochlear implant listeners

Julie G. Arenberg*, Wendy S. Parkinson *- University of Washington, Department of Speech and Hearing Sciences, Seattle, WA, USA*
Leonid Litvak *- Advanced Bionics Corporation, Hearing Performance Research, Valencia, CA, USA*

Cochlear implants are designed to deliver spectral and temporal information to auditory neurons near electrodes arrayed along the tonotopic axis of the cochlea. Unwanted channel interaction can reduce the effectiveness with which this information is transmitted. The goal of this study is to reduce channel interaction and to improve speech perception performance. Dynamic focusing, a novel method for stimulating implants, is tested which employs a highly focused configuration for threshold inputs and a broader configuration for input levels near most comfortable listening to reduce channel interaction and better mimic the acoustic activation of the normal hearing cochlea. To examine the improvements possible with listening experience, three, adults, implanted with the Advanced Bionics device, have taken home research processors with such experimental strategies for up to one month. Testing was performed before and after take-home with sentences in quiet at 60 dB-A and with additional four-talker babble (Auditek) at a +10 signal-to-noise ratio. Testing in noise was only performed if scores in quiet exceeded 60% correct. In all cases performance on speech perception improved over time or was within ~5% of the initial scores. For one listener, performance met and exceeded the level of the baseline testing, when listeners used their everyday program settings.

Corresponding author: Julie Arenberg (jbierer@uw.edu)

S5.4 – Fri 25 Aug, 10:10-10:30

Data-driven hearing care with time stamped logging

Niels Henrik Pontoppidan, Xi Li, Lars Bramsløw* - *Eriksholm Research Centre, Oticon A/S, Snekkersten, Denmark*
Benjamin Johansen - *DTU Compute, Technical University of Denmark, Kgs. Lyngby, Denmark*
Claus Nielsen, Atefeh Hafez - *Eriksholm Research Centre, Oticon A/S, Snekkersten, Denmark*
Michael Kai Petersen - *Eriksholm Research Centre, Oticon A/S, Snekkersten, Denmark; DTU Compute, Technical University of Denmark, Kgs. Lyngby, Denmark*

Modern hearing devices holds significant personalization potentials while the processes associated with the administration do not fully accommodate the dialogue for finding the optimized and personalized settings. The hearing devices presented here use a connected smartphone to log a snapshot of 21 sound environment parameters every minute, e.g. sound pressure level in low, mid, and high frequencies and broad band, the estimate of the signal-to-noise ratio in the same 4 bands, the sound environment detector, etc. This data stream shows the sound environments which the user of the hearing devices experience. The continuous stream of sound environment data is supplemented by the users operation of the hearing device, e.g., which program is chosen when, and how is the volume control adjusted as well. Whenever the user changes program or volume, the change is logged with the time stamp. Together, the continuous and event based data logging will reveal in which situations the user prefers a given program and on the bigger time scale which program that should be the default program. The close integration of the hearing device, the mobile phone, and cloud services turning it into Internet of Things device not only enable the learning and adaptation but also supplementing the dialogue between user and audiologist with objective data about the actual use of the hearing devices.

Corresponding author: Niels Henrik Pontoppidan (npon@eriksholm.com)

S5.5 – Fri 25 Aug, 10:30-10:50

A pinna-cues preserving beamforming algorithm for hearing aids

Gabriel Gomez*, Bernhard U. Seeber - *Audio-Information Processing, Technical University of Munich, Munich, Germany*

A new pinna-cues preserving beamforming filter algorithm for hearing aids is presented, which combines the advantages of preserving pinna cues (naturalness, good spatial sound quality, externalization) with SNR gain benefits of traditional beamforming. The algorithm was validated with 8 normal hearing participants in different experiments against the conditions behind-the-ear (BTE), in-the-ear (ITE) and static delay-and-subtract beamformer (BF) with attenuation at 180°. In a localization experiment with front and back sound presentation, the new algorithm performed equally well as the ITE condition, with a mean localization error between -+30° of 4.6° (vs. 42.9° BTE, 35.8° BF). In an externalization rating (ER) experiment (0: internalized, 3: at the loudspeaker), the new algorithm performed equally well as the ITE condition with a mean ER of 2.8 (vs. 2.3 BTE, 1.9 BF). In a spatial sound quality experiment with complex acoustic scenes, the new algorithm was rated best for preserving the spatial quality of frontal sources in the dimensions externalization, diffuseness, locatability and source separation. Finally, in an OLSA speech understanding experiment with either one disturber at 180° (1N) or two babble disturbers (2N) at 7.5° and 165°, the new algorithm performed equal to the BF condition, with a signal-to-noise ratio advantage of 8 dB for the 1N condition and 4.5 dB for the 2N condition over the ITE/BTE average speech reception thresholds.

Corresponding author: Gabriel Gomez (gabriel.gomez@tum.de)

S5.6 – Fri 25 Aug, 11:10-11:40

Real world self adjustment of a hearing assistance device

Andrew T. Sabin*, Dianne J. Van Tasell *- Bose Corporation, Framingham, MA, USA*

In common practice, the selection of hearing assistance signal processing parameters is done by a clinician. Here, we present and evaluate a method for fitting hearing assistance devices in which users select their own parameters through an application on a mobile device. The user interface of this application (Ear Machine) provides two simple controls that allow the user to select a full set of wide dynamic range compression parameters across 12 audio frequency bands. We will report on how this interface is used in everyday life by persons with mild to moderate hearing losses. Participants were given mobile devices (Apple iPod Touch) containing an application (Ear Machine) that wirelessly controlled a Bluetooth® headset performing wide dynamic range compression (Bose Hearphones™). For all participants, an initial fit and fine tuning was performed by an audiologist. Over the next month, while wearing the device in their everyday lives participants either (a) selected their own parameters using the Ear Machine interface or (b) used the parameters selected by the clinician. Data collection is ongoing; early results are consistent with our laboratory studies indicating that, on average, the user-selected parameters were more-preferred than the clinician selected ones and were within a range appropriate for participants' hearing losses.

Corresponding author: Andrew Sabin (andrew_sabin@bose.com)

S5.7 – Fri 25 Aug, 11:40-12:10

Adapting to hearing aid use to achieve full benefit

Nikolai Bisgaard* *- GN Hearing A/S, Ballerup, Denmark*

People who have lived with a hearing loss for several years have adapted to the changes in their hearing abilities as the hearing loss increased over time. When they are fitted with hearing aids, they experience a sudden change in auditory input that require them to adapt to the new situation to gain full benefit of their hearing aids. The outcome of this process varies substantially across users. Recent research indicates that user satisfaction is positively influenced by the efforts exercised by professionals during the adaptation period, but even after such efforts, some users have a very low use time of their hearing aids. Over that last 8 years, thousands of hearing aid users have been responding to questionnaires in the EuroTrak program. The results of these studies show an increasing adoption of hearing aids in general and in particular of bilateral solutions. Satisfaction scores with the different factors contributing to user satisfaction are also analyzed and will be presented. Distributions of use time and satisfaction will be presented. Adoption and use patterns vary across countries and these differences and their possible origin will also be presented and discussed.

Corresponding author: Nikolai Bisgaard (nbisgaard@gnresound.com)

Poster Sessions I and II

Posters will remain on display throughout the symposium.

Presenters will be at their posters:

Wed 23 Aug, 17:00-19:00 (odd-numbered posters)
Thu 24 Aug, 17:00-19:00 (even-numbered posters)

P.1 – Wed 23 Aug, 17:00-19:00

Relation between speech intelligibility and perceived width of virtual sound sources

Axel Ahrens*, Torsten Dau, Marton Marschall *- Hearing Systems, Department of Electrical Engineering, Technical University of Denmark, Kgs. Lyngby, Denmark*

Loudspeaker-based virtual sound environments (VSEs) are emerging as a versatile tool for studying human auditory perception. Higher-order ambisonics (HOA) is a spatial audio reproduction technique that allows placing sound sources in arbitrary directions while keeping the spatial spread of sound energy constant. The energy spread increases with decreasing ambisonics order and has been shown to lead to a larger perceived source width for low orders. However, it is unclear whether the broadening of the reproduced sound source also leads to a reduced ability to separate closely-spaced target and interfering speech. This study investigated the effect of ambisonics order on speech intelligibility and perceived source width. Speech reception thresholds (SRTs) were measured with two interferers separated by ±15°, in an anechoic and a simulated reverberant environment, presented at various HOA orders. Listeners were also asked to estimate the location and perceived width of the sound sources. Results showed that while perceived source width varied with ambisonics order, speech intelligibility remained constant. This suggests that speech intelligibility is robust to a degradation of spatial information as represented by low ambisonic orders, despite spatial perception being affected.

Corresponding author: Axel Ahrens (aahr@elektro.dtu.dk)

P.2 – Thu 24 Aug, 17:00-19:00

Cortical auditory evoked potentials to speech stimulus in children with normal hearing and hearing aid users

Mohammad Shamim Ansari*,S, Shivraj Bhimte *- Ali Yavar Jung National Institute of Speech and Hearing Disabilities (Divyangjan), Mumbai, India*

Speech perception ability is a developmental indicator of speech and language skills in children with sensory and communication disorders. Recently, cortical auditory evoked potentials to speech stimulus have been developed to measure speech perception abilities in children with auditory disorders that can provide an objective indicator of functional capacity of auditory cortex. The study aims to explore the correlation between speech perception scores and aided cortical evoked potentials in children using hearing aids. Speech perception scores and cortical evoked potentials were obtained with the speech stimulus |ba| in 42 hearing-aid users and 42 typical children. The speech perception scores and auditory evoked potentials differed significantly between the groups. The typically developing children showed a robust wave P1 compared to hearing-aid users. The hearing-aid users were further categorized into two groups, good performers and poor performers based on their speech perception scores. The data revealed that the mean amplitude of P1 of the poor performers was significantly lower than those of the good performers and the typically developing group of children. Speech perception scores showed a positive correlation with P1 amplitude but did not show a significant correlation with P1 latency. Overall, the findings indicate that speech perception scores reflected in the P1 component are related to the behavioral auditory capacity of children with normal hearing and hearing-aid users.

Corresponding author: Mohammad Shamim Ansari (msansari5000@yahoo.com)

P.3 – Wed 23 Aug, 17:00-19:00

Comparing DPOAE and behavioral measures of the basilar-membrane input-output function

Konstantinos Anyfantakis*, Michal Fereczkowski, Bastian Epp, Ewen N. MacDonald *- Hearing Systems, Department of Electrical Engineering, Technical University of Denmark, Kgs. Lyngby, Denmark*

The cochlea is one of the first stages of the auditory pathway and yet its functioning cannot be measured in a direct way. Among several behavioural methods designed for determining basilar membrane input/output (BMI/O) function, the temporal masking curve (TMC) paradigm is the most popular. However, otoacoustic emissions (OAEs) provide an objective method for evaluating the cochlear nonlinearity and do not require active participation of the listener. While a correlation between compression knee-point estimates of BMI/O from TMC and distortion-product (DP) OAEs has been found, no correlation between corresponding estimates of compression rate has been reported, particularly in hearing-impaired listeners. We hypothesize that the lack of correlation could be due to the interplay between source and reflection components in the recorded DPOAEs. In this study, we compare basilar-membrane I/O function estimates from the TMC experiment and the corresponding DPOAE-I/O-gram at three audiometric frequencies (1, 2, and 4 kHz) in 10 normal-hearing listeners. The DPOAEs are evoked using sweeps, which allows unmixing of the source and reflection components and estimation of the I/O gram based on the source-component alone. The clinical feasibility of both paradigms is discussed.

Corresponding author: Michal Fereczkowski (mfer@elektro.dtu.dk)

P.4 – Thu 24 Aug, 17:00-19:00

Effect of musical training on pitch discrimination performance in older normal-hearing and hearing-impaired listeners

Federica Bianchi* *- Hearing Systems, Department of Electrical Engineering, Technical University of Denmark, Kgs. Lyngby, Denmark*

Sébastien Santurette *- Hearing Systems, Department of Electrical Engineering, Technical University of Denmark, Kgs. Lyngby, Denmark; Department of Otorhinolaryngology, Head and Neck Surgery & Audiology, Rigshospitalet, Copenhagen, Denmark*

Torsten Dau *- Hearing Systems, Department of Electrical Engineering, Technical University of Denmark, Kgs. Lyngby, Denmark*

Hearing-impaired (HI) listeners, as well as elderly listeners, typically have a reduced ability to discriminate the pitch of complex tones as compared to young normal-hearing (NH) listeners. This perceptual deficit may be ascribed to a variety of factors, such as reduced frequency selectivity, degraded temporal fine structure cues and decreased neural synchrony. Several studies have shown that musical training, on the other hand, leads to an improved pitch-discrimination performance in NH listeners. However, it is unclear whether a comparable effect of musical training can be observed in listeners whose cues for pitch perception are degraded. To address this question, pitch discrimination was investigated in three groups of listeners (young NH, elderly NH and HI listeners), each including musicians and non-musicians, using complex tones that differed in harmonic content. For the young NH listeners, musical training significantly increased pitch-discrimination performance in all conditions. For the elderly NH and HI listeners, improved pitch discrimination was observed only for the high-numbered harmonics, but not for the low-numbered harmonics, for which pitch is retrieved via place and/or temporal fine structure cues. These findings suggest an interaction between training-induced plasticity and the availability of peripheral cues.

Corresponding author: Federica Bianchi (fbia@elektro.dtu.dk)

P.5 – Wed 23 Aug, 17:00-19:00

Spectral and binaural loudness summation in order to fit hearing aids bilaterally

Monique Boymans* *- Department of Clinical and Experimental Audiology, Academic Medical Center, Amsterdam, The Netherlands*
Dirk Oetting *- HörTech, Oldenburg, Germany*
Mirjam van Geleuken, Wouter A. Dreschler *- Department of Clinical and Experimental Audiology, Academic Medical Center, Amsterdam, The Netherlands*

Aversiveness of loud sounds is a frequent complaint by hearing-aid users, especially when fitted bilaterally. This study investigates whether loudness summation can be held responsible for this finding. Two aspects of loudness perception should be taken into account: spectral loudness summation and binaural loudness summation for narrow and broadband signals. In this study, two main aspects were investigated: (i) the effect of different symmetrical hearing losses according the classification (N2, N3, N4, S2 and S3) of Bisgaard et al (2010) and (ii) the effect of spectral contents of the signals, by using high frequency noise and low frequency noise. For the assessment of loudnes perception we used Adaptive Categorical LOudness Scaling (ACALOS), as developed by Brand and Hohmann (2001). Also loudness matching was applied as a potentially faster technique to use in a clinical setting, to get information about the individual loudness perception in order to fine tune hearing aids in the future. We found large inter-individual differences with unexpectedly high - but reproducible - levels of binaural loudness summation in some of the subjects. The general trend of the findings was the same for loudness scaling and loudness matching. Our results suggest that individual loudness perception deserves a more prominent role in prescriptive fitting rules. The pros and cons for using such an approach in the clinic will be discussed.

Corresponding author: Monique Boymans (m.boymans@amc.nl)

P.6 – Thu 24 Aug, 17:00-19:00

An improved competing voices test for test of attention

Lars Bramsløw*, Marianna Vatti, Rikke Rossing, Niels Henrik Pontoppidan *-*
Eriksholm Research Centre, Oticon A/S, Snekkersten, Denmark

People with hearing impairment find competing voices scenarios to be challenging for their ability to switch attention and adapt to the situation. With the Competing Voices Test (CVT) we can explore how they can adapt and change their attention between voices. We have earlier presented results from this newly developed listening test using one male and one female speaker, presenting pairs of sentences from the Danish HINT test. We have now recorded five additional speakers, for a total of three male and three female speakers, and administered the updated test on 14 moderate-severely hearing impaired listeners. Three methods of cueing the listener to the target speaker were tested: the original male/female cue (for male+female sentence pairs), an audio voice cue and a text cue using one word from the target sentence. The cue was presented either before or after the sentence pair playback. Four spatial conditions were tested: summed (diotic), separate (dichotic) plus two types of ideal masks for separating the two speakers. The results show that the added talkers and test methods have made the test more sensitive, when using the text cue and regardless of the chosen speaker pair. This cue has the further advantage that it can be used for e.g. male+male speaker pairs as well. Furthermore, the applied ideal masks show test scores very close to the ideal separate spatial condition.

Corresponding author: Lars Bramsløw (labw@eriksholm.com)

P.7 – Wed 23 Aug, 17:00-19:00

Predicting the benefit of binaural cue preservation in bilateral directional processing schemes for listeners with impaired hearing

Thomas Brand*, Christopher Hauth, Tobias Neher *- Medizinische Physik and Cluster of Excellence "Hearing4All", Universität Oldenburg, Oldenburg, Germany*

Linked pairs of hearing aids offer various possibilities for directional processing providing adjustable tradeoff between signal-to-noise ratio improvement and binaural cue preservation. The potential benefit of directional processing depends on the processing scheme, the acoustic scenario, and the listener's ability to exploit binaural cues (cf. Neher et al, ISAAR 2017). Here, a binaural speech intelligibility model (BSIM) was used to predict the benefit of five processing schemes in three acoustic scenarios individually for elderly listeners with symmetric (N = 20) or asymmetric (N = 19) hearing thresholds below 2 kHz. The acoustic scenarios consisted of a frontal target talker presented against two speech maskers from ±60° azimuth or spatially diffuse cafeteria noise. In (Neher et al, ISAAR 2017), N0Sπ detection performance at 500 Hz was found to be a good predictor of individual benefit from low-frequency binaural cues. This is consistent with predictions of an extended BSIM, which takes individual binaural processing accuracy into account. Furthermore, the binaural frontend of BSIM was extended to predict binaural unmasking of speech based on mixed target and interferer signals. This may enable BSIM, in principle, to be used for adapting bilateral directional processing in hearing aids to the individual listener's abilities and the current acoustical situation.

Corresponding author: Thomas Brand (thomas.brand@uni-oldenburg.de)

P.8 – Thu 24 Aug, 17:00-19:00

Sensitivity of encoding models for auditory fMRI

Mette L. V. Carstensen [*,S] *- Hearing Systems, Department of Electrical Engineering, Technical University of Denmark, Kgs. Lyngby, Denmark; Danish Research Centre for Magnetic Resonance, Copenhagen University Hospital Hvidovre, Hvidovre, Denmark*

Søren Asp Fuglsang [S]***, Torsten Dau*** *- Hearing Systems, Department of Electrical Engineering, Technical University of Denmark, Kgs. Lyngby, Denmark*

Jens Hjortkjær *- Hearing Systems, Department of Electrical Engineering, Technical University of Denmark, Kgs. Lyngby, Denmark; Danish Research Centre for Magnetic Resonance, Copenhagen University Hospital Hvidovre, Hvidovre, Denmark*

Functional MRI provides spatially accurate information about auditory processing. However, generalizability of fMRI results across stimuli, experimental protocols, scanner hardware, etc. is a recurring problem, e.g., with traditional statistical parametric mapping or multi-voxel pattern classification. Voxel-wise encoding models propose an alternative approach that is driven by explicit model predictions about the auditory representations that give rise to the measured functional activity. Encoding models, however, also pose new challenges in terms of experimental designs and choice of imaging parameters. Model accuracy is usually computed via cross-validation, and 'stimulus-rich' designs can be used to test model generalizability on unseen data. Due to the poor signal-to-noise ratio in fMRI, a trade-off exists between stimulus variability, which enables generalizability, and signal averaging, which improves SNR to allow for more accurate model fitting. Here, we evaluate the effects of different stimulus-related parameters (stimulus variability, inter-stimulus interval, stimulation length) and imaging parameters (scan repetition time, multiband acquisition techniques, spatial resolution). We show that for optimized parameter settings, encoding models are sensitive to frequency tuning in early auditory cortex and to higher-order selectivity for phoneme categories in the superior temporal gyrus.

Corresponding author: Jens Hjortkjær (jhjort@elektro.dtu.dk)

P.9 – Wed 23 Aug, 17:00-19:00

Innovative methods and technologies for spatial listening and speech intelligibility using hearing implants

Anja Chilian *- Fraunhofer Institute for Digital Media Technology, Ilmenau, Germany*
Maria Gadyuchko *- Mechatronics Group, Technische Universität Ilmenau, Ilmenau, Germany*
András Katai *- Fraunhofer Institute for Digital Media Technology, Ilmenau, Germany*
Ralf Keilig *- Mechatronics Group, Technische Universität Ilmenau, Ilmenau, Germany*
Florian Klein *- Electronic Media Technology Group, Technische Universität Ilmenau, Ilmenau, Germany*
Verena Skuk *- Research Unit Person Perception, Institute for Psychology, Friedrich Schiller University of Jena, Jena, Germany*
Stephan Werner* *- Electronic Media Technology Group, Technische Universität Ilmenau, Ilmenau, Germany*

The proportion of population with reduced hearing abilities is increasing worldwide. Specific types of hearing diseases require the treatment with hearing implants. Cochlear implants and bone conduction implants are two examples. The present contribution and underlying project addresses new methods and technologies that improve spatial hearing with such implants. The methods are adjusted specifically for both types of hearing implants. For cochlear implants bio-inspired signal processing methods are applied. For bone conduction implants new operating principles for mechanical stimulation based on piezo-electric transducers are investigated. To evaluate the developments perceptional experiments are conducted, which investigate spatial listening and speech intelligibility with normal hearing and hearing impaired persons. For this purpose a virtual listening environment is applied to synthesize different room acoustics, source positions, audio signals, and acoustic scenes with different complexity. Cochlear implants and a custom-made bone conduction device are used as playback systems. The bone conduction device generates the mechanical input and transmits mechanical oscillations via the temporal bone to the cochlea. Listening tests assess speech intelligibility in noisy environments and localization abilities.

Corresponding author: Stephan Werner (stephan.werner@tu-ilmenau.de)

P.10 – Thu 24 Aug, 17:00-19:00

Measuring hearing instrument sound modification using integrated ear-EEG

Florian Denk*, Marleen Grzybowski *- Medizinische Physik and Cluster of Excellence Hearing4all, University of Oldenburg, Germany*

Stephan M.A. Ernst *- Medizinische Physik and Cluster of Excellence Hearing4all, University of Oldenburg, Germany; Present address: ENT Clinic, University Hospital Gießen und Marburg, Gießen, Germany*

Stefan Debener, Martin Bleichner *- Neuropsychology Lab and Cluster of Excellence Hearing4all, University of Oldenburg, Germany*

We integrated ear-centred electrodes into a live hearing system and evaluate the feasibility of extracting valid EEG features in this configuration by means of an auditory discrimination experiment. The long-term goal is to construct a closed-loop brain-computer-interface that is integrated in a mobile research hearing system. The EEG setup comprises 3 electrodes embedded in the earmoulds of an experimental hearing system and 10 electrodes positioned around each ear, which are all connected to one wireless amplifier. Four consecutive identical broadband stimuli were played on headphones. However, the spectral profile of sounds arriving at the eardrum was altered by switching the signal processing setting of the hearing system. Such switches were made between presentation of the third and the fourth tone, in half of all epochs. 17 normal hearing subjects participated. To assure their attendance, they were instructed to indicate whether the last tone sounded different. The behavioural data also verified clear audibility of the switches. The EEG analysis revealed a significant difference in the event related potential in response to the fourth tone between switch and non-switch epochs. Furthermore, we evaluated the discrimination performance with the ear-electrodes when utilizing different electrodes and referencing configurations, as well as single-subject and single-trial data.

Corresponding author: Florian Denk (florian.denk@uni-oldenburg.de)

P.11 – Wed 23 Aug, 17:00-19:00

Auditory disabilities, individual fitting targets and the compensation power of hearing aids: Successes and the gap still to be bridged

Wouter A. Dreschler*, Inge de Ronde-Brons, Monique Boymans *- Clinical & Experimental Audiology, AMC, Amsterdam, The Netherlands*

At ISAAR 2015 we presented a systematic approach based on individual user profiles for Human Related Intended Use (HRIU) of hearing instruments. HRIU is based on self-report data: a modified version of the AIADH, combined with a COSI-approach. AIADH results determine the profile of disabilities and COSI results determine the profile of targets. The differences between these profiles can be interpreted as the profile of compensation needs. Until now, we collected data of more than 2500 hearing aid users. In the pre-fitting data the effect of audiometric factors and amnamnestic factors have been studied. Post-fitting results show improvements in each of the 6 dimensions and determine the degree of compensation provided by the hearing aid. But there are striking differences between the effective compensation provided by hearing aids. Again, these differences can be related to audiometric and amnamnestic factors, but there are also related to the kind of hearing aid fitting and the type of the hearing aid fitted. The results also indicate the limitations of current technology.

Corresponding author: Wouter Dreschler (w.a.dreschler@amc.uva.nl)

P.12 – Thu 24 Aug, 17:00-19:00

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

Gerard Encina-Llamas* *- Hearing Systems, Department of Electrical Engineering, Technical University of Denmark, Kgs. Lyngby, Denmark*
Aravind Parthasarathy *- Massachusetts Eye and Ear Infirmary, Harvard Medical School, Boston, MA, USA*
James M. Harte *- Interacoustics Research Unit, Kgs. Lyngby, Denmark*
Torsten Dau *- Hearing Systems, Department of Electrical Engineering, Technical University of Denmark, Kgs. Lyngby, Denmark*
Sharon Kujawa *- Massachusetts Eye and Ear Infirmary, Harvard Medical School, Boston, MA, USA*
Barbara Shinn-Cunningham *- Boston University, Boston, MA, USA*
Bastian Epp *- Hearing Systems, Department of Electrical Engineering, Technical University of Denmark, Kgs. Lyngby, Denmark*

The ability to communicate is a fascinating property of the healthy auditory system. Despite normal sensitivity to pure-tones, many listeners complain about having difficulties in challenging situations with high levels of background noise. Recent animal studies have shown that noise over-exposure that produces temporary threshold shifts can cause the loss of auditory nerve (AN) fiber synapses (synaptopathy). 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. The AN model can account for the general trends obtained from human EFR level-growth functions. Off-frequency contributions of high-spontaneous-rate (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. High-SR fibers dominate the total response in the simulated EFRs. The model simulation were compared to EFR level-growth functions recorded in exposed and non-exposed mice. The simulated EFR level-growth functions agree very well with EFR level-growth functions recorded in mice, where synaptopathy can be quantified.

Corresponding author: Gerard Encina-Llamas (encina@elektro.dtu.dk)

P.13 – Wed 23 Aug, 17:00-19:00

Optimizing the microphone array size for a virtual artificial head

Mina Fallahi*, Matthias Blau, Martin Hansen *- Institut für Hörtechnik und Audiologie, Jade Hochschule Oldenburg, Oldenburg, Germany*

Simon Doclo, Steven van de Par *- Department of Medical Physics and Acoustics and Cluster of Excellence Hearing4All, Carl von Ossietzky Universität Oldenburg, Oldenburg, Germany*

Dirk Püschel *- Akustik Technologie Göttingen, Göttingen, Germany*

Rasumow et al. (IEEE TASLP, 24, 2016) showed that spatial directivity patterns of individual binaural head-related transfer functions (HRTFs) can be accurately synthesized in the horizontal plane using a microphone array with 24 microphones and individual digital filtering. To calculate the filter coefficients, a narrow-band least squares cost function was minimized for a limited number of directions and the robustness of the synthesis at intermediate directions not included in the optimization was enhanced by imposing a constraint on the average white noise gain. Fallahi et al. (DAGA, 2017) improved the performance of this virtual artificial head by imposing additional constraints on the synthesis error for a large number of directions, showing that the synthesis error can be kept below 2 dB up to 8 kHz for a dense resolution grid in the horizontal plane. Using the same method we investigate here the effect of different microphone array sizes on the synthesis accuracy. To this end, individual HRTFs were synthesized with simulated arrays of different sizes and were compared to the original HRTFs, both objectively in terms of spectral distortion and interaural-level-difference error metrics and subjectively in a listening test with 10 participants. The results confirm that array size is a major factor and that the synthesis accuracy can be improved by carefully choosing an appropriate array size.

Corresponding author: Mina Fallahi (mina.fallahi@jade-hs.de)

P.14 – Thu 24 Aug, 17:00-19:00

Steering of audio input in hearing aids by eye gaze through in-ear electrodes

Antoine Favre-Felix*, Carina Graversen *- Eriksholm Research Centre, Snekkersten, Denmark*
Torsten Dau *- Hearing Systems, Department of Electrical Engineering, Technical University of Denmark, Kgs. Lyngby, Denmark*
Thomas Lunner *- Eriksholm Research Centre, Snekkersten, Denmark*

The behavior of a person during a conversation typically involves both auditory and visual attention. Visual attention implies that the person directs their eye gaze towards the sound target of interest, and hence detection of the gaze may provide a steering signal for future hearing aids. Identification of the sound target of interest could be used to steer a beamformer or select a specific audio stream from a set of remote microphones. We have previously shown that in-ear electrodes can be used to identify eye gaze through electrooculography (EOG) in offline recordings. However, additional studies are needed to explore the precision and real-time feasibility of the methodology. To evaluate the methodology we performed a test with hearing-impaired subjects seated with their head fixed in front of three targets positioned at -30°, 0°, and +30°. Each target presented speech from the Danish DAT material, which was available for direct input to the hearing aid using head related transfer functions. Speech intelligibility was measured in three conditions: a reference condition without any steering, an ideal condition with steering based on an eye-tracking camera, and a condition where EarEOG were measured to select the desired audio stream. The capabilities and limitations of the methods are discussed.

Corresponding author: Antoine Favre-Felix (afav@eriksholm.com)

P.15 – Wed 23 Aug, 17:00-19:00

Evaluation of the noise reduction of hearing aids in situations with multiple signals

Marlitt Frenz* *- German Institute of Hearing Aids, Lübeck, Germany*
Alfred Mertens *- Universität zu Lübeck, Lübeck, Germany*
Hendrik Husstedt *- German Institute of Hearing Aids, Lübeck, Germany*

In 2004 Hagerman and Olofsson presented a method to objectively evaluate the noise reduction of hearing aids. This method provides the signal-to-noise ratio (SNR) at the output of the hearing aid while speech and noise are presented simultaneously. If two signals shall be distinguished, a comprehensive evaluation of the noise reduction is possible. However, in many hearing situations more than two signals are simultaneously present, e.g. multiple noise sources and/or multiple speakers. Therefore, in this study, an extension of the method of Hagerman and Olofsson for more than two signals is introduced. To proof the concept, a setup with 8 speakers is used where all speakers are equally distributed on a circle around the hearing aid. The radius of this circle is 1 m and the angular distance between the speakers is 45°. In this study, speech is presented from an angle of 0° and 8 different noise signals are presented from all 8 directions. With the extended method, the speech signal and all 8 noise signals can be separated. As a result, the individual SNR of each direction is depicted in polar diagrams for different settings. These settings include four hearing aid configurations, where the noise reduction and/or the directional microphones are turned on or off.

Corresponding author: Marlitt Frenz (m.frenz@dhi-online.de)

P.16 – Thu 24 Aug, 17:00-19:00

Speech-in-noise processing in elderly hearing-impaired listeners with or without hearing aid experience: Eye-tracking and fMRI measurements

Julia Habicht*, Oliver Behler, Birger Kollmeier, Tobias Neher *- Medizinische Physik and Cluster of Excellence "Hearing4all", Oldenburg University, Oldenburg, Germany*

While previous research has investigated speech-in-noise processing in normal and impaired hearing, the influence of hearing aid (HA) experience on these processes is underexplored. Using an eye-tracking paradigm that allows determining how fast a participant can grasp the meaning of a sentence presented against noise together with two pictures that correctly or incorrectly depict the sentence meaning, Wendt et al (Trends Hear., 2015) and Habicht et al (Trends Hear., 2016) found inexperienced HA (iHA) users to be slower than experienced HA (eHA) users, despite no differences in speech intelligibility. To further explore the influence of HA experience on speech-in-noise processing, we adapted the eye-tracking paradigm for functional magnetic resonance imaging (fMRI) measurements. We then performed eye-tracking and fMRI measurements with groups of eHA and iHA users (N = 15 each) matched in terms of age, hearing loss and working memory capacity. From the results, we expect that lack of HA experience is associated with the recruitment of brain areas outside the speech comprehension network, that is, reduced brain activity in auditory cortex and increased brain activity in regions of frontal and temporal gyrus.

Corresponding author: Julia Habicht (julia.habicht@uni-oldenburg.de)

P.17 – Wed 23 Aug, 17:00-19:00

Does interaural loudness and pitch matching improve bimodal benefit?

Wiebke Heeren**, Josef Chalupper, Gunnar Geissler *- Advanced Bionics, ERC, Hannover, Germany*
Silke Klawitter, Andreas Büchner *- Department of Otolaryngology, Medical University Hannover, Hannover, Germany*

Some bimodal listeners with significant spectral overlap between electric and acoustic hearing show improved localization and spatial release from masking compared to unilateral hearing. However, there are large inter-individual differences. This study aims at determining whether an optimized bimodal fitting comprising interaural loudness and pitch matching increases bimodal benefit and provides benefit in a larger group of patients. Subsequently, candidacy criteria most likely leading to a benefit are provided. For this purpose, the performance of 11 experienced bimodal patients with their clinical fitting compared to different experimental fittings is investigated. Experimental fittings are either (a) automatic-gain-control (AGC) matched, (b) AGC matched and loudness balanced or (c) AGC matched, loudness balanced and pitch matched. The results show a significant improvement of 1.3 dB for speech understanding in competing talker situations with interventions (a) and (b) compared to the clinical fitting. Intervention (c), however, reveals large individual variations with 2 patients showing a clear benefit and 3 degrading in performance. Localization improved by about 5° for all interventions compared to clinical fitting. Correlating the individual results with secondary measures such as loudness and pitch mismatch or performance in quiet does not lead to clear-cut criteria for gaining bimodal benefit.

Corresponding author: Wiebke Heeren (wiebke.heeren@advancedbionics.com)

P.18 – Thu 24 Aug, 17:00-19:00

Digit speech: A test material for measuring ongoing speech perception

Jens Hjortkjær* *- Hearing Systems, Department of Electrical Engineering, Technical University of Denmark, Kgs. Lyngby, Denmark; Danish Research Centre for Magnetic Resonance, Copenhagen University Hospital Hvidovre, Hvidovre, Denmark*
Jonatan Märcher-Rørsted *- Hearing Systems, Department of Electrical Engineering, Technical University of Denmark, Kgs. Lyngby, Denmark*

We present a new speech test material developed to monitor speech processing and perception during listening. Traditional speech tests typically measure speech recognition by collecting listener responses after single words or sentences. However, there is often a need to monitor behaviour during natural continuous speech, e.g., during concurrent physiological measurements. To address this need, we developed a 'digit speech' test material consisting of multiple recorded tokens of 3-digit numbers spoken with different natural intonation patterns. The tokens are concatenated to create continuous speech stimuli that match important properties of natural speech in terms of sentence intonation structure, syllabic rate, and phonetic content. Speech sequences can then be created to work with different behavioural tasks, e.g., simple repetition tasks to monitor attention or N-back tasks to control listeners' working memory load during listening. We used the material to derive speech reception thresholds in noise for normal hearing listeners, which enables the speech to be presented at pre-defined intelligibility levels. We demonstrate a potential use by showing EEG and pupilometry data obtained with the digit speech material. We show well-defined cortical response functions to the continuous speech stimulus obtained under varying degrees of working memory load.

Corresponding author: Jens Hjortkjær (jhjort@elektro.dtu.dk)

P.19 – Wed 23 Aug, 17:00-19:00

Clinical measures for investigating hidden hearing loss

Pernille Holtegaard* - *Hearing Systems, Department of Electrical Engineering, Technical University of Denmark, Kgs. Lyngby, Denmark*
Josefine Juul Jensen, Sara A. B. Al-Ward - *Department of Nordic Studies and Linguistics, Copenhagen University, Copenhagen, Denmark*
Bastian Epp - *Hearing Systems, Department of Electrical Engineering, Technical University of Denmark, Kgs. Lyngby, Denmark*

The present study compared clinical measures of auditory function in two listener groups prone to hidden hearing loss: a) listeners with tinnitus, and b) listeners with a history of noise-exposure relative to a control group. Auditory brainstem response (ABR) wave I, III and V were measured in response to a 4 kHz tone burst to quantify the level-growth of wave I and the amplitude difference between wave I-III and I-V. In addition, high-frequency sensitivity and speech-in-noise performance using the Danish hearing in noise test (HINT) were assessed in the control and the noise-exposed groups. The ABR results showed no difference between the tinnitus-, noise-exposed- and control groups regarding wave-I level-growth and amplitude differences of wave I. The listeners with tinnitus had, however, larger wave III and V amplitudes indicating a gain at brainstem level. High-frequency audiometry showed neither a significant difference between the noise-exposed and the control groups nor a correlation between high-frequency sensitivity and speech-in-noise performance. While the ABR results support that wave III can be used as a physiological indicator of tinnitus, none of the applied audiological methods show signs of a hidden hearing loss. The results suggest that a refinement of audiological methods might be required to investigate hidden hearing loss.

Corresponding author: Pernille Holtegaard (perholt@elektro.dtu.dk)

P.20 – Thu 24 Aug, 17:00-19:00

BEAR: A status on population characteristics of hearing-aid users obtained from the database

Sabina Storbjerg Houmoeller* - *Department of ENT/Audiology, Odense University Hospital, Odense, Denmark*

Anne Wolff*[S]*, Dan Dupont - *Department of Otolaryngology, Head & Neck Surgery and Audiology, Aalborg University Hospital, Aalborg, Denmark*

Dorte Hammershøi - *Department of Electronic Systems, Aalborg University, Aalborg, Denmark*

Christian Godballe, Jesper Hvass Schmidt - *Department of ENT/Audiology, Odense University Hospital, Odense, Denmark*

Recent studies estimate that around 20% of the hearing aid (HA) owners in Denmark do not regularly use their HA, which implies that they are not satisfied with the HA treatment and did not experience the benefits they expected. The overall vision of the BEAR project is to improve hearing rehabilitation in Denmark through an evidence-based renewal of clinical practice and optimization of the individual HA fitting, resulting in a successful hearing aid treatment. Initially the project aims to describe the current clinical practice, by building a clinical database containing the existing data on HA profiling and fitting strategies. Data from 2000 patients are collected from the departments of Audiology in Odense and Aalborg University Hospital. Some of the registered variables include gender, age, and health related questions, noise exposure and whether they are suffering from tinnitus. The database thereby contains descriptive data on the current population of HA users in Denmark and their health status. Descriptive data in different relevant subgroups of HA users obtained from the database will be presented. Based on the results, analyses of average age and the distribution of gender in different subgroups of HA users, including experienced versus new HA users will be made. Previous noise exposure and the relation to the occurrence of tinnitus will be investigated as well.

Corresponding author: Sabina Storbjerg Houmoeller (shoumoeller@health.sdu.dk)

P.21 – Wed 23 Aug, 17:00-19:00

A method to analyze and test the automatic selection of hearing aid programs

Hendrik Husstedt*, Simone Wollermann *- German Institute of Hearing Aids, Lübeck, Germany*
Jürgen Tchorz *- University of Applied Sciences Lübeck, Lübeck, Germany*

Digital hearing aids usually provide different hearing aid programs. This means different settings can be selected to adapt the signal processing to different hearing situations. For some devices, the user manually selects the desired hearing aid program. For other, more advanced devices, a classification algorithm continuously analysis the acoustic environment and selects a hearing aid program accordingly. However, there exist no method to analyze this adaptive feature. Therefore, we present a method that allows one to analyze and test what hearing aid program is really active in a specific hearing situation. To this end, different test signals are played and the hearing aid is programed with different settings. To proof the concept, hearing aids of different manufacturers are analyzed. These results depict the differences between the manufacturers and show what signals are difficult to distinguish for hearing aids. Furthermore, the results of one device are compared with the entries of the data logging feature, which shows good agreement. Using the data logging feature takes multiple hours, since entries are only stored after a long signal duration (typically after 30 min). Moreover, the data logging does not allow for analyzing any transient behavior so that this is no alternative to the method proposed.

Corresponding author: Hendrik Husstedt (h.husstedt@dhi-online.de)

P.22 – Thu 24 Aug, 17:00-19:00

A longitudinal fNIRS study of cross-modal plasticity and speech understanding outcomes in cochlear implant users: first update

Hamish Innes-Brown* *- Bionics Institute, Melbourne, Australia; Department of Medical Bionics, University of Melbourne, Melbourne, Australia*
Xin Zhou[S], Mehrnaz Shoushtarian *- Bionics Institute, Melbourne, Australia*
Colette McKay *- Bionics Institute, Melbourne, Australia; Department of Medical Bionics, University of Melbourne, Melbourne, Australia*

Although cochlear implants (CIs) improve speech understanding on average, there is wide and unexplained variability in outcomes. Predictive factors available pre-implant, including duration of deafness, age, and age at implant, collectively only account for ~12% of speech understanding outcome variance. Brain plasticity that occurs during deafness and rehabilitation may help explain this variance, but imaging methods like PET and fMRI are either too invasive for repeated use or cannot be used with CIs. We have therefore started a longitudinal study using a non-invasive brain imaging technique (functional near-infrared spectroscopy – fNIRS) which follows 30 CI candidates from 2 weeks pre-, to 2 years post-implant. We collect speech understanding and lip-reading scores pre-implant, and fNIRS data while speech is presented in audio-alone, visual-alone, and audio-visual modalities. Post-implant, we repeat the fNIRS and speech-understanding tasks. At the first post-implant session, we have found that CI users with poor speech understanding with the new CI had high levels of visually-evoked fNIRS activity in auditory brain areas, while those with good speech understanding did not, potentially highlighting the effect of cross-modal plasticity during deafness on speech understanding. This presentation will provide an update on the study, focusing on adaptive and maladaptive plasticity.

Corresponding author: Hamish Innes-Brown (hinnes-brown@bionicsinstitute.org)

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Objective assessment of binaural fusion in young normal-hearing listeners

Niclas A. Janßen* *- Hearing Systems, Department of Electrical Engineering, Technical University of Denmark, Kgs. Lyngby, Denmark*
Lars Bramsløw *- Eriksholm Research Centre, Snekkersten, Denmark*
Søren Riis *- Oticon Medical, Smørum, Denmark*
Jeremy Marozeau *- Hearing Systems, Department of Electrical Engineering, Technical University of Denmark, Kgs. Lyngby, Denmark*

The normal auditory system can fuse sounds from both ears into a single sound object (binaural fusion). This ability has only been measured subjectively so far by asking whether listeners perceive one or two sounds or by the scale illusion percept found by Diana Deutsch in 1975. The aim of this study was to develop an objective task to measure binaural fusion. Twelve young normal hearing participants had to detect one deviant note within a stream composed of a repeating melody while simultaneously being presented with another stream of randomized notes. The experiment included three conditions. First, in the monaural condition both streams were presented to the same ear. Then, in the binaural condition every second note from each stream was presented to the other ear. The results were expected to be higher if the listeners can fuse across ears. Finally, in the binaural control condition, the timbre of all the notes presented to one ear was altered severely, in order to prevent binaural fusion. Each condition had 24 repetitions. In the binaural and monaural conditions, average performance was about 80% correct, while the control condition showed a significantly lower performance of about 50%. Thus, this type of experiment can be used to test objectively if fusion takes place. It lays the foundation for further studies with bilateral and bimodal cochlear implant listeners.

Corresponding author: Niclas A. Janßen (najan@elektro.dtu.dk)

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Effects of slow-acting and very fast-acting compression on hearing-impaired listeners' CV identification in interrupted noise

Borys Kowalewski*, Johannes Zaar, Michal Fereczkowski, Ewen N. MacDonald *- Hearing Systems, Department of Electrical Engineering, Technical University of Denmark, Kgs. Lyngby, Denmark*

Olaf Strelcyk *- Sonova U.S. Corporate Services, Warrenville, IL, USA*

Tobias May, Torsten Dau *- Hearing Systems, Department of Electrical Engineering, Technical University of Denmark, Kgs. Lyngby, Denmark*

There is conflicting evidence about the relative benefit of fast- and slow acting compression for speech intelligibility. It can be hypothesized that fast-acting compression improves audibility at low signal-to-noise ratios (SNRs) but may distort the speech envelope at higher SNRs. The present study investigated the effects of compression with nearly instantaneous attack but either fast (10 ms) or slow (500 ms) release times on consonant identification in hearing-impaired listeners. Consonant-vowel (CV) speech tokens were presented at several sound pressure levels in two conditions: in the presence of interrupted noise, and in quiet (with the compressor "shadow"-controlled by the corresponding noisy mixture). By comparing these results the effects of consonant audibility and forward masking by the interrupted noise can be disentangled. A small but systematic benefit of fast-acting compression was found in both the quiet and the noisy conditions for the lower speech levels. Despite potentially detrimental speech envelope distortions, no negative effects of fast-acting compression were observed when the speech level exceeded the level of the noise. These findings suggest that fast-acting compression provides an audibility benefit in fluctuating interferers as compared to slow-acting compression, while not substantially affecting the perception of short CV tokens at higher SNRs.

Corresponding author: Borys Kowalewski (bokowal@elektro.dtu.dk)

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An improved privacy-aware system for objective and subjective ecological momentary assessment

Ulrik Kowalk*, Sven Kissner, Petra von Gablenz, Inga Holube, Joerg Bitzer - *Institute of Hearing Technology and Audiology, Jade University of Applied Sciences, Oldenburg, Germany*

In this contribution we present the technical components and software features of a new hearing-aid compatible smartphone-based ecological momentary assessment (EMA) system. EMA in hearing research provides deeper insight into the relation of acoustical characteristics and the respective individual perception of everyday listening situations. This work builds upon an already developed and deployed smartphone-based system. The microphones of the aforementioned EMA system, however, were incompatible with behind-the-ear hearing aids. Since linking objective acoustical measures to subjective assessments is particularly promising with regard to the hearing rehabilitation process, our system has been adapted for hearing aid users. The introduction of wireless data transfer has eliminated cable clutter, a customizable questionnaire allows for subjective assessment, and a streamlined user interface complements the design. The existing feature set is extended by enhanced estimators of signal-to-noise ratio and reverberation in order to refine the characterization of situations. Like the former version, the current revision ensures the privacy of participants and third parties. To facilitate cooperative research, source code and custom-built hardware are released under open-source licenses. All additional components are commercially available.

Corresponding author: Ulrik Kowalk (ulrik.kowalk@jade-hs.de)

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Investigating the effects of noise-estimation errors in simulated cochlear implant speech intelligibility

Abigail Anne Kressner, Tobias May, Rasmus Malik Thaarup Høegh*, Kristine Aavild Juhl, Thomas Bentsen, Torsten Dau *- Hearing Systems, Department of Electrical Engineering, Technical University of Denmark, Kgs. Lyngby, Denmark*

A recent study suggested that the most important factor for preserving high speech intelligibility in noise with cochlear implant recipients is to preserve the low-frequency modulations of speech across time and frequency by, for example, minimizing the amount of noise in speech gaps. In contrast, other studies have argued that the transients provide the most information. Thus, the present study investigates the relative impact of these two factors in the framework of noise reduction by systematically varying the gains applied within speech segments, speech gaps, and the transitions between them. Speech intelligibility in noise was measured using a basic cochlear implant simulation tested on normal-hearing listeners. Results suggest that minimizing noise in the speech gaps can substantially improve intelligibility, especially in modulated noise. However, significantly larger improvements were obtained when both the noise in the gaps was minimized and the speech transients were preserved. These results imply that being able to correctly identify the boundaries between speech segments and speech gaps is the most important factor in maintaining high intelligibility in cochlear implants, since knowing the boundaries will allow for both minimization of the noise in the gaps and enhancement of the low-frequency modulations.

Corresponding author: Abigail Kressner (aakress@elektro.dtu.dk)

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Semantic differential analysis of electric pulse trains in cochlear implant listeners

Wiebke Lamping* *- Hearing Systems, Department of Electrical Engineering, Technical University of Denmark, Kgs. Lyngby, Denmark*
Sébastien Santurette *- Hearing Systems, Department of Electrical Engineering, Technical University of Denmark, Kgs. Lyngby, Denmark; Department of Otorhinolaryngology, Head and Neck Surgery & Audiology, Rigshospitalet, Copenhagen, Denmark*
Jeremy Marozeau *- Hearing Systems, Department of Electrical Engineering, Technical University of Denmark, Kgs. Lyngby, Denmark*

For cochlear implant (CI) users, temporal cue and place cue are assumed to vary along two orthogonal perceptual dimensions linked to pitch height and timbre, respectively. The aim of the present project was to study the effect of electrode place, pulse rate, and amplitude modulation frequency on those perceptual dimensions and to assess whether they are independent from each other. Two different sets of loudness-balanced stimuli were presented to the listeners. The first set was created by all possible combinations of electrode numbers 22, 18, 14, and 10 with pulse rates of 80, 150, 300, 600, and 1200 pps. The second stimulus set was composed of amplitude-modulated pulse trains with modulation frequencies of 80, 150, 200, 300, and 400 Hz on a constant carrier of 1200 pps, presented via the same electrodes as in set 1. The subjects were asked to rate pitch and sound quality using multiple verbal attributes. For all attributes tested, the statistical analysis revealed no significant interaction effect between the electrode place with neither pulse rate nor modulation frequency. A comparison between the ratings for modulated and unmodulated pulse trains showed no significant difference in agreement with previous studies. Overall, these results indicate that place and temporal cues induce two independent perceptual dimensions that can be both linked to pitch and timbre.

Corresponding author: Wiebke Lamping (wila@elektro.dtu.dk)

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Speech intelligibility in dual task with hearing aids and adaptive digital wireless microphone technology

Matthias Latzel* *- Phonak AG, Global Marketing, Stäfa, Switzerland*
Kirsten C. Wagener, Matthias Vormann *- HZ Oldenburg, Oldenburg, Germany*
Hans Mülder *- Phonak Communications AG, Murten, Switzerland*

Wireless microphones have been developed in order to support hearing aid users to understand distant talkers. A drawback of these systems is the deteriorated speech intelligibility in the near field. A new system has been developed supporting the hearing aid user also in the near field when using an external microphone (EM), by enabling the directional microphone of the hearing aid. To verify the performance of this novel system, speech intelligibility tests have been conducted using a dual task paradigm. Primary task: Sentences of the female Oldenburg Matrix Test were presented continuously in a randomized order. The task of the subject was to mark the recognized name on a tablet after each trial. All 10 possible names where visible on the tablet throughout the whole test. Secondary task: At the same time a speech recognition test with meaningful sentences (Göttinger Sentence Test, male voice) was carried out at a fixed signal to noise ratio. The task of the subject was to repeat the sentence to an investigator. Firstly, the Primary task was presented from a far field loudspeaker and the secondary task from a near field loudspeaker and vice versa. Results of 19 hearing impaired subjects confirm a benefit of the EM both for the primary and the secondary task. The directional microphone showed superior performance in comparison to the omni microphone in the near field. For a higher ecological validity, the data were analyzed considering both tasks simultaneously. This analysis showed a positive effect of the directional microphone when the primary task is presented from the loudspeaker in near field. This is supported also subjectively, when assessing the listening effort.

Corresponding author: Matthias Latzel (matthias.latzel@phonak.com)

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On the cost of introducing speech-like properties to a stimulus for auditory steady-state response measurements

Søren Laugesen* *- Interacoustics Research Unit, Kgs. Lyngby, Denmark*
Julia E. Rieck[S] *- Faculty of Sciences, Free University Amsterdam, The Netherlands*
Claus Elberling *- Virum, Denmark*
Torsten Dau *- Hearing Systems, Department of Electrical Engineering, Technical University of Denmark, Kgs. Lyngby, Denmark*
James M. Harte *- Interacoustics Research Unit, Kgs. Lyngby, Denmark*

Validating hearing-aid fitting in pre-lingual infants poses a challenge as typical measures (such as aided audiometry, questionnaires, etc.) are not possible. One objective approach for validation in the clinic uses an aided auditory steady-state response (ASSR) measurement. In order to make a valid measurement, it is important that all of the hearing aid's adaptive signal processing features are activated or (de-activated) as if the ASSR stimulus were real speech. Rather than manipulating the settings of the hearing aid to achieve this, an ASSR stimulus with speech-like properties has been developed. The stimulus consists of the narrow-band (NB) CE-Chirps®, modified to mimic the International Speech Test Signal (ISTS). However, the modifications to the standard NB CE-Chirps® are accompanied by reduced ASSR amplitudes and increased detection times. This was investigated in an experiment with 10 normal-hearing listeners. The observed changes to the measured ASSR amplitudes and response times are compared to an objective characterisation of the ISTS-modified versus the standard NB CE-Chirps®.

Corresponding author: Søren Laugesen (slau@iru.interacoustics.com)

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The body as instrument: tissue conducted multimodal audio-tactile spatial music

Peter Lennox* *- College of Arts, University of Derby, Derby, UK*
Ian McKenzie *- School of Engineering, University of Derby, Derby, UK*
Michael Brown *- College of Arts, University of Derby, Derby, UK*

Whilst the putative human audible frequency range is 20 Hz to 20 kHz, and frequencies below 20 Hz do not reach auditory limen, they are supraliminal in the tactile domain. Their contribution to 'auditory perception' remains unclear; it may be that concomitant audio-plus-tactile stimuli have additive effects in multimodal perception. We describe early progress in exploring the compositional potential for multimodal music of a multi-transducer audio-plus-vibrotactile apparatus, originally envisaged as an assistive technology for those with some degree of conductive hearing loss. The equipment uses five transducers located at various sites on the cranium. Audio signals are controlled via ambisonic-plus-discrete signal transcoding to generate spatial impressions. The tactile component was an incidental by-product, carried by the same transducers. We conducted an elicitation exercise with one hundred uninstructed listeners who described the experience in their own terms. Responses were aggregated to identify emergent descriptive themes of available qualia. Apparently, the tactile component assumes greater importance in the perceptual experience than originally considered. In addressing quality-of-life and music-deprivation issues, addition of a tactile information-channel is potentially interesting. We review contemporary theories of multimodal perception in this context.

Corresponding author: Peter Lennox (p.lennox@derby.ac.uk)

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Contribution of low- and high-frequency bands to binaural unmasking in hearing-impaired listeners

Gusztáv Lőcsei *- Hearing Systems, Department of Electrical Engineering, Technical University of Denmark, Kgs. Lyngby, Denmark*
Sébastien Santurette *- Hearing Systems, Department of Electrical Engineering, Technical University of Denmark, Kgs. Lyngby, Denmark; Department of Otorhinolaryngology, Head and Neck Surgery & Audiology, Rigshospitalet, Copenhagen, Denmark*
Torsten Dau, Ewen N. MacDonald* *- Hearing Systems, Department of Electrical Engineering, Technical University of Denmark, Kgs. Lyngby, Denmark*

This study investigated the contribution of interaural timing differences (ITDs) in different frequency regions to binaural unmasking (BU) of speech. Speech reception thresholds (SRTs) and binaural intelligibility level differences (BILDs) were measured in two-talker babble in 6 young normal-hearing (NH) and 9 elderly hearing- impaired (HI) listeners with normal or close-to-normal hearing at and below 1.5 kHz. Target sentences were presented diotically, embedded in a stream of diotic or dichotic maskers. Both target and masker sentences were split into frequency regions above and below 1.25 kHz. In the dichotic listening conditions, the maskers were lateralized to the left side by introducing 0.68-ms ITDs in either the low-frequency band, the high-frequency band, or both bands simultaneously. BILDs were found to be similar in both listener groups when the ITDs were imposed on the low-frequency band only. ITDs in the high-frequency band alone did not produce any BILD in any of the groups. However, when ITDs were imposed in both frequency bands, the NH listeners yielded significantly greater BILDs than the HI listeners. The results suggest that, on a group level, HI listeners relied solely on ITDs in the low-frequency band while NH listeners were able to utilize envelope ITDs above 1.25 kHz to facilitate the BU of speech.

Corresponding author: Ewen MacDonald (emcd@elektro.dtu.dk)

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Lateralized speech perception with small interaural time differences in normal-hearing and hearing-impaired

Gusztáv Lőcsei *- Hearing Systems, Department of Electrical Engineering, Technical University of Denmark, Kgs. Lyngby, Denmark*

Sébastien Santurette *- Hearing Systems, Department of Electrical Engineering, Technical University of Denmark, Kgs. Lyngby, Denmark; Department of Otorhinolaryngology, Head and Neck Surgery & Audiology, Rigshospitalet, Copenhagen, Denmark*

Torsten Dau, Ewen N. MacDonald* *- Hearing Systems, Department of Electrical Engineering, Technical University of Denmark, Kgs. Lyngby, Denmark*

Spatial release from masking (SRM) elicited by interaural timing differences (ITDs) only can be almost normal for listeners with symmetrical hearing loss. This study investigated whether elderly hearing-impaired (HI) listeners still achieve similar SRMs as young normal-hearing (NH) listeners when SRMs are elicited by small ITDs. Speech reception thresholds (SRTs) and SRM due to ITDs were measured over headphones for 10 young normal-hearing (NH) and 10 older hearing-impaired (HI) listeners, who had normal or close-to-normal hearing below 1.5 kHz. Diotic target sentences were presented in diotic or dichotic speech-shaped noise (SSN) or two-talker babble (TT) maskers. In the dichotic conditions, maskers were lateralized by delaying the masker waveforms in the left headphone channel. Multiple magnitudes of masker ITDs were tested in both noise conditions. Although deficits were observed in speech perception abilities in SSN and two-talker babble in terms of SRTs, HI listeners could utilize ITDs to a similar degree as NH listeners to facilitate the binaural unmasking of speech. A slight difference was observed between the group means when target and maskers were separated from each other by large ITDs, but not when separated by small ITDs. Thus, HI listeners do not appear to require larger ITDs than NH listeners do in order to receive a benefit from binaural unmasking.

Corresponding author: Ewen MacDonald (emcd@elektro.dtu.dk)

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Influence of multi-microphone signal enhancement algorithms on auditory movement detection in acoustically complex situations

Micha Lundbeck* *- Medizinische Physik and Cluster of Excellence "Hearing4all", Oldenburg University, Oldenburg, Germany; HörTech gGmbH, Oldenburg, Germany*
Laura Hartog *- Medizinische Physik and Cluster of Excellence "Hearing4all", Oldenburg University, Oldenburg, Germany*
Giso Grimm, Volker Hohmann *- Medizinische Physik and Cluster of Excellence "Hearing4all", Oldenburg University, Oldenburg, Germany; HörTech gGmbH, Oldenburg, Germany*
Lars Bramsløw *- Eriksholm Research Centre, Snekkersten, Denmark*
Tobias Neher *- Medizinische Physik and Cluster of Excellence "Hearing4all", Oldenburg University, Oldenburg, Germany*

The influence of hearing aid (HA) signal processing on the perception of spatially dynamic sounds has not been systematically investigated so far. Previously, we observed that for elderly hearing-impaired (EHI) listeners concurrent distractor sounds impaired the detectability of left-right source movements, and reverberation that of near-far source movements (Lundbeck et al., Trends Hear., 2017). Here, we explored potential ways of improving these deficits with HAs. To that end, we carried out detailed acoustic analyses on the stimuli used previously to examine the impact of two beamforming algorithms and a binaural coherence-based noise reduction scheme on the cues underlying movement perception. While the binaural cues remained mostly unchanged, the applied processing led to greater monaural spectral changes, as well as increases in signal-to-noise ratio and direct-to-reverberant sound ratio. Based on these findings, we conducted a listening test with 20 EHI listeners. That is, we performed aided measurements of movement detectability with three different processing conditions in two acoustic scenarios. Our results indicate that, for both movement dimensions, the applied processing could partly restore source movement detection in the presence of concurrent distractor sounds.

Corresponding author: Micha Lundbeck (micha.lundbeck@uni-oldenburg.de)

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The relationship between frequency selectivity and stream segregation of complex tones

Sara M. K. Madsen*, Torsten Dau *- Hearing Systems, Department of Electrical Engineering, Technical University of Denmark, Kgs. Lyngby, Denmark*
Brian C. J. Moore *- Department of Psychology, University of Cambridge, Cambridge, UK*

Discrimination of changes in fundamental frequency (F0) is better for complex tones with low than with high harmonics, perhaps because the low harmonics are resolved. The reduced frequency selectivity of hearing-impaired (HI) listeners may lead to poorer resolution of low and medium harmonics. Also, the extent of perceptual segregation (streaming) of a rapid sequence of complex tones may depend on the discriminability of the difference in F0, F0, between successive tones. We assessed how the streaming of complex tones is affected by harmonic rank and whether tones with low and medium harmonics stream less for HI than for normal-hearing (NH) listeners. Subjective streaming was assessed for complex tones bandpass filtered between 2 and 4 kHz. Harmonic rank was varied by changing the baseline F0 (with F0 from 1 to 11 semitones). Auditory filter shapes were estimated from notched-noise masking using a 2-kHz signal. The auditory filters were wider for the HI than for the HI listeners. Streaming decreased with increasing harmonic rank. Streaming was similar for the two groups; however, auditory filter widths were correlated with the reduction in streaming for complex tones relative to 2-kHz pure tones. This, and the greater variance of streaming across listeners with narrow than with wide filters, suggests that resolvability affects streaming but is not the only factor involved.

Corresponding author: Sara M. K. Madsen (samkma@elektro.dtu.dk)

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The sound sensation of electric stimulation in single-sided deafened cochlear implant recipients

Jeremy Marozeau* - *Hearing Systems, Department of Electrical Engineering, Technical University of Denmark, Kgs. Lyngby, Denmark*
Marine Ardoint, Dan Gnansia - *Oticon Medical CI Scientific Research, Vallauris, France*
Diane Lazard - *Arthur Vernes Institute, Paris, France; Nottingham University, Nottingham, UK*

Ten cochlear implant users with single-sided deafness were asked to vary the parameters of an acoustic sound played to the normal-hearing ear, in order to match its perception with that of the electric sensation of two electrodes (e20 and e14). The experiment was divided in 3 consecutive conditions in which the nature of the acoustic sound and the controlled parameters varied. In the first condition the subject had to vary the frequency of a pure tone; in the second condition, the center frequency and the bandwidth of a filter applied to a harmonic complex sound; and in the last condition, the F0 and the harmonicity factor of a complex sound. Averaged frequencies of the pure tones matching e20's and e14's pitch were significantly different. In the second condition, only the average center frequencies of the band-pass filters were significantly different. In the third condition, the average F0s were not significantly different; the harmonicity factor was 1.7 for both electrodes. Overall, this study suggests that the sound sensation of different electrodes is more linked to a difference in timbre (brightness) than to a difference in pitch, and is more similar to an inharmonic complex sound than to a pure tone.

Corresponding author: Jeremy Marozeau (jemaroz@elektro.dtu.dk)

P.36 – Thu 24 Aug, 17:00-19:00

Overview of new outcome tools addressing auditory ecological validity: Analyses of behavior in real life listening environments

Markus Meis*, Melanie Krüger *- Hörzentrum Oldenburg GmbH, Germany; Cluster of Excellence Hearing4all, Oldenburg, Germany*

Maria Gebhard *- Hörzentrum Oldenburg GmbH, Germany; Cluster of Excellence Hearing4all, Oldenburg, Germany; Jade University of Applied Sciences, Oldenburg, Germany*

Petra von Gablenz, Inga Holube *- Institute of Hearing Technology and Audiology, Jade University of Applied Sciences, Oldenburg, Germany; Cluster of Excellence Hearing4all, Oldenburg, Germany*

Giso Grimm *- Hörtech gGmbH, Oldenburg, Germany; Cluster of Excellence Hearing4all, Oldenburg, Germany*

Richard Paluch [S] *- Hörzentrum Oldenburg GmbH, Germany; Cluster of Excellence Hearing4all, Oldenburg, Germany; Carl von Ossietzky Universität, Oldenburg, Germany*

Numerous studies showed that different hearing aid (HA) algorithms improve speech intelligibility in typical lab situations as measures of clinical efficacy. However, from the perspective of auditory ecology it remains obscure to what extent these results really allow for estimating the performance in complex listening situations in real life. In a series of lab studies, using virtual acoustics, it was shown, that different directional modes of HAs influenced real life behavior of the participants (Paluch et al., 2016; Meis et al., 2016). These behavioral changes can be linked to the International Classification of Functions (ICF) concepts activity and participation. Based on these findings, two further research threads are ongoing. First, within the realm of the Oldenburg Cluster of Excellence Hearing4all, the behavior of hearing-impaired adults in real life situations is described by means of ethnographical methods and compared to the behavior in the lab using complex audiovisual scenes to enhance ecological validity. Second, we are going to establish a new tool of 'momentary quality of life' in the Hearing Industry Research Consortium project 'Individual Hearing Aid Benefit in Real Life' (IHAB-RL). This tool provides a unique combination of individual self-assessment and related external observation in the field following the ICF. The underlying theoretical approaches and first results in the direction of behavior driven new outcome tools will be discussed.

Corresponding author: Markus Meis (m.meis@hoerzentrum-oldenburg.de)

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Characterizing speech-in-noise perception under aided conditions based on mobile EEG recordings

Bojana Mirkovic*, Stefan Debener *- Department of Psychology, University of Oldenburg, Germany; Cluster of Excellence "Hearing4all", Oldenburg, Germany*
Tobias Neher *- Medizinische Physik, University of Oldenburg, Germany; Cluster of Excellence "Hearing4all", Oldenburg, Germany*

Features derived from electroencephalography (EEG) recordings may help to infer individual listening demands. If combined with hearing aids (HA), such an approach could be used for online fitting purposes, potentially leading to better speech perception in daily-life situations. To investigate the potential of listening demand characterization in daily-life, groups of normal-hearing and hearing-impaired participants were instructed to listen to an audio book in speech-babble noise while EEG was recorded with a wireless, mobile EEG device. Two HA conditions were tested: unprocessed and noise reduction processing. The participants' task was to answer questions about the story they heard. In addition, participants' motivation was manipulated by offering a monetary reward in half of the trials. Consistent with previous findings, preliminary event-related potential analysis at the onset of acoustic conditions suggested a condition difference in the P300 response. Furthermore, initial results suggest that a linear regression algorithm estimating the auditory evoked potential from continuous EEG during listening can be employed to extract additional features contributing to the characterization of individual listening demands. The possibility of online classification of extracted EEG features will be discussed.

Corresponding author: Bojana Mirkovic (bojana.mirkovic@uni-oldenburg.de)

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Assessing listening effort with pupillometry, one word at a time

Annie Moulin *- Inserm U1028, CNRS UMR 5292, Brain Dynamics and Cognition Team, Lyon Neuroscience Research Center, University of Lyon, Lyon, France*
Fanny Cholvy *- Starkey, Créteil, France*
Stéphane Gallego *- Rehabilitation Science and Technology Institute, University of Lyon, Lyon, France; Audition Conseil, Lyon, France*
Mathieu Fernschneider *- Audition Conseil, Lyon, France*
Christophe Micheyl* *- Starkey, Créteil, France; Starkey Hearing Research Center, Starkey Hearing Technologies, Berkeley, CA, USA; Inserm U1028, CNRS UMR 5292, Cognition and Auditory Perception Team, Lyon, France; Neuroscience Research Center, University of Lyon, Lyon, France*

An important aspect of hearing-aid benefits relates to reduced listening effort. Previous research indicates that listening effort may be assessed objectively through pupil dilation. However, additional work is needed before pupil dilation can be used to quickly and reliably measure listening effort at an individual and/or single-stimulus level. This study sought to measure changes in pupil-dilation during speech-in-noise listening at the single-subject, single-stimulus level. Pupil dilation was measured before, during, and after the presentation of isolated bisyllabic words in speech-shaped or cocktail-party noise in both normal-hearing and hearing-impaired individuals. Listeners repeated each word after a time lag. For normal-hearing participants, pupil dilation peaked about 1.3 s after word onset, on average. With scores at ceiling (100% correct), words presented in cocktail party noise induced significantly larger dilation than words presented in speech-shaped noise. Partially correctly repeated words induced larger pupil dilations than fully correctly repeated ones. For some participants, such changes in pupil dilation could be detected at the single-stimulus level. These findings further indicate that pupil-dilation measures offer promise as a non-invasive technique to objectively assess listening effort even under conditions where speech intelligibility is at ceiling.

Corresponding author: Christophe Micheyl (christophe_micheyl@starkey.com)

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From the auditory nerve to the auditory cortex in a cochlear implanted animal model: how are variables of the electric waveform represented?

Charlotte Amalie Navntoft,S* - *Brain Sound Lab, Department of Biomedicine, University of Basel, Basel, Switzerland*
Jeremy Marozeau - *Hearing Systems, Department of Electric Engineering, Technical University of Denmark, Kgs. Lyngby, Denmark*
Tania Rinaldi Barkat - *Brain Sound Lab, Department of Biomedicine, University of Basel, Basel, Switzerland*

Cochlear implants (CI) can restore the sense of hearing in deaf people by electrically stimulating the auditory nerve (AN). The standard electric stimulation is a series of symmetric, biphasic pulses but such a pulse shape makes it hard to determine which aspects of the stimuli most efficiently excite the AN. This study aims to better understand the transmission chain of the electric waveform from the AN to the auditory cortex using a CI animal model. To date, we have implemented an anaesthetized CI mouse model (4-channel electrode array: 0.2 mm diameter, 0.2 mm contact width, inserted into the 1st basal turn) and have recorded electrically-evoked auditory brainstem response with biphasic and monophasic stimuli of different intensities and durations (biphasic: 80 µs/phase; monophasic: 100-1000 µs cathodic/anodic phase duration). In parallel, we have performed population imaging (voltage sensitive dye imaging) of acoustic-evoked signals in the auditory cortex in normal-hearing mice to establish a tonotopic map. Following experiments will combine the CI-model and cortical imaging to investigate how cortical network representation is affected by different pulse shape, polarities, and amplitude modulations. Our findings will help us to better understand the electro-neural interface. The outcome has the potential to guide future pulse shape design that can lead to more efficient CI.

Corresponding author: Charlotte Amalie Navntoft (charlotte.navntoft@unibas.ch)

P.40 – Thu 24 Aug, 17:00-19:00

Hearing aid noise suppression and working memory function

Tobias Neher *- Medizinische Physik and Cluster of Excellence "Hearing4all", University of Oldenburg, Germany*
Kirsten C. Wagener *- Hörzentrum Oldenburg GmbH, Oldenburg, Germany*
Rosa-Linde Fischer* *- R&D Audiology, Sivantos GmbH, Germany*

Research findings concerning the relation between outcome from hearing aid (HA) noise suppression and working memory function are unclear. The current study thus investigated the effects of three noise suppression algorithms on auditory working memory as well as the relation with reading span. Using a computer simulation of bilaterally fitted HAs, four settings were tested: (1) unprocessed, (2) directional microphones, (3) single-channel noise reduction, and (4) binaural coherence-based noise reduction. Settings 2-4 were matched in terms of the speech-weighted signal-to-noise ratio (SNR) improvement. Auditory working memory was assessed at +6 dB SNR using a listening span and an N-back paradigm. Twenty experienced HA users aged 55-80 yr with large differences in reading span took part. For the listening span measurements, there was an influence of HA setting on final word recognition and recall, with the directional microphone setting leading to approx. 6% better performance than the single-channel noise reduction setting. For the N-back measurements, there was substantial test-retest variability and no influence of HA setting. No interactions with reading span were found. These results imply that HA noise suppression can affect the recognition and recall of speech at positive SNRs, irrespective of individual reading span.

Corresponding author: Rosa-Linde Fischer (rosa-linde.fischer@sivantos.com)

P.41 – Wed 23 Aug, 17:00-19:00

Adapting bilateral directional processing to individual and situational influences

Tobias Neher* *- Oldenburg University, Oldenburg, Germany*
Kirsten C. Wagener *- Hörzentrum Oldenburg GmbH, Oldenburg, Germany*
Matthias Latzel *- Phonak AG, Stäfa, Switzerland*

This study examined differences in benefit from bilateral directional processing. Groups of listeners with symmetric or asymmetric audiograms <2 kHz, a large spread in the binaural contribution to speech-in-noise reception (BILD), and no difference in age or overall degree of hearing loss took part. Aided speech reception was measured using virtual acoustics together with a simulation of a linked pair of hearing aids. Five processing schemes and three acoustic scenarios were used. The processing schemes differed in the tradeoff between signal-to-noise ratio (SNR) improvement and binaural cue preservation. The acoustic scenarios consisted of a frontal target talker and two lateral speech maskers or spatially diffuse noise. For both groups, a significant interaction between the BILD, processing scheme, and acoustic scenario was found. This interaction implied that, for lateral speech maskers, users with BILDs >2 dB profit more from low-frequency binaural cues than from greater SNR improvement, whereas for smaller BILDs the opposite is true. Audiometric asymmetry reduced the BILD influence. In spatially diffuse noise, maximal SNR improvement was beneficial. Moreover, binaural tone-in-noise detection ($N0S\pi$ threshold) at 500 Hz predicted benefit from low-frequency binaural cues. These results provide a basis for adapting bilateral directional processing to the user and the scenario.

Corresponding author: Tobias Neher (tobias.neher@uol.de)

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Ethnographic research: The interrelation of auditory spatial awareness, everyday life / laboratory environments and effects of hearing aids

Richard Paluch* [*,S] *- Hörzentrum Oldenburg GmbH, Oldenburg, Germany; Social Sciences, University of Oldenburg, Oldenburg, Germany; Cluster of Excellence "Hearing4all", Oldenburg, Germany*

Melanie Krüger *- Hörzentrum Oldenburg GmbH, Oldenburg, Germany; Cluster of Excellence "Hearing4all", Oldenburg, Germany*

Maartje Hendrikse *- Medical Physics, University of Oldenburg, Oldenburg, Germany; Cluster of Excellence "Hearing4all", Oldenburg, Germany*

Giso Grimm, Volker Hohmann *- HörTech gGmbH, Oldenburg, Germany; Medical Physics, University of Oldenburg, Oldenburg, Germany; Cluster of Excellence "Hearing4all", Oldenburg, Germany*

Markus Meis *- Hörzentrum Oldenburg GmbH, Oldenburg, Germany; Cluster of Excellence "Hearing4all", Oldenburg, Germany*

Hearing is multidimensional. It affects the whole body (e.g., balance and motility). However, it is still an open question whether and how these general factors of everyday life are affected by the use of modern hearing aids (HA) with complex signal processing options. This study, therefore, addressed the question to what extent hearing aids may shape the HA users' everyday life. Thus, the behavior of HA users was observed experimentally using a theory-based ethnographic research design that comprises written reports and several steps of theorizing and reasoning. Data were collected in two specific situations (road traffic and restaurant), by three modes (unaided, omnidirectional and directional microphone mode) and by two settings (everyday life and laboratory with realistic virtual audio-visual reality). The analytical results of the ethnographical studies were summarized and used for testing hypotheses in an advanced laboratory with realistic virtual audio-visual environments. Furthermore, the analytical results led to suggestions for the (re-)creation of realistic virtual audio-visual scenes. The research outcomes (e.g., different behavior patterns) are shown in this contribution.

Corresponding author: Richard Paluch (richard.paluch@uni-oldenburg.de)

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A bilateral hearing-aid algorithm that provides directional benefit while preserving situational awareness

Tobias Piechowiak*, Rob de Vries, Andrew Dittberner, Chang Ma *- GN ReSound A/S, Ballerup, Denmark*

A directional filter or beamformer is a classical approach to ensure speech intelligibility by suppressing distracting sounds from certain directions. However, they have some challenges on their own: (a) white-noise gain, (b) diminished benefit in reverberation, and (c) off-axis audibility problems. In this study a new algorithm with the name Binaural Auditory Steering Strategy (BASS) is introduced. It is designed to provide good situational awareness (SA) while mainting directional benefit (DB). SA is maximized by combining the sensitivity of both left and right hearing aids to create a true binaural omnidirectional sensitivity pattern. DB is ensured by promoting the better-ear effect and thus allowing for better separation of sounds.

Corresponding author: Tobias Piechowiak (tpiechowiak@gnresound.com)

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Resolving front-back ambiguity via narrow-band binaural dynamics

Henri Pöntynen*, Nelli Salminen *- Department of Signal Processing and Acoustics, Aalto University, Espoo, Finland*

Behavioral studies assessing the role of head movements in free-field localization often report excellent performance with stimuli providing low-frequency interaural time differences (ITD), but markedly poorer performance with narrow-band high-frequency stimuli. In the current study, we investigated the effectiveness of dynamic narrow-band interaural level differences (ILD) in resolving front-back ambiguities in a set of experiments, where subjects used head rotations to localize sinusoidal targets to the front or rear hemiplanes. Head tracking was used to set the effective movement window with head-orientation-coupled gating of the target signals. Experiments were conducted both under free-field conditions and as headphone studies for which simplified ITD and ILD contours were derived from the diffraction equations of a sphere. Results from free-field experiments show that low-frequency tones were consistently localized to the correct hemiplane, even when the movement range was severely limited. Performance was poorer with high-frequency tones but increased at wider movement windows. Results from headphone experiments display similar results for both ITD and ILD stimuli regardless of frequency. This suggests that the poor free-field localization performance observed with high-frequency tones could be accounted for by morphology-related acoustic effects that give rise to a complex relationship between narrow-band ILD and source azimuth.

Corresponding author: Henri Pöntynen (henri.pontynen@aalto.fi)

P.45 – Wed 23 Aug, 17:00-19:00

Preferences for digital noise reduction and directional microphone settings

Karrie Recker*, Adriana Goyette, Jason Galster *- Starkey Hearing Technologies, Eden Prairie, MN, USA*

Can we predict hearing aid wearers' preferences for strength of noise reduction and for microphone mode using tests of noise tolerance and preferred speech levels (PSLs)? Our hypotheses were (1) people who are intolerant of background noise will prefer more aggressive noise reduction than people who are tolerant of background noise; (2) people who receive more benefit from noise reduction algorithms and directional microphones will prefer more aggressive versions of these features than people who receive little benefit from them; (3) people who prefer to listen to speech at higher signal-to-noise ratios (SNRs) will be more likely to prefer directional microphones than those who prefer to listen at lower SNRs. Twenty individuals with hearing loss participated in this laboratory study: 10 had a low noise tolerance and 10 had a high noise tolerance, as measured with the Acceptable Noise Level (ANL) test. All participants were fitted with bilateral hearing aids and completed laboratory testing (ANLs, PSLs and preference testing) with four levels of noise reduction (0, 6, 10 and 20 dB), three microphone modes (omnidirectional, high-frequency directionality, and broadband directionality) and four combinations of noise reduction and directional microphones. Participants completed multiple iterations of each test (randomized) at two SNRs. Detailed results and clinical implications will be discussed.

Corresponding author: Karrie Recker (karrie_recker@starkey.com)

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Extending a computational model of auditory signal processing and perception towards predicting speech intelligibility

Helia Relaño-Iborra,S*, ***Johannes Zaar, Torsten Dau*** *- Hearing Systems, Department of Electric Engineering, Technical University of Denmark, Kgs. Lyngby, Denmark*

A speech intelligibility model is presented, based on the computational auditory signal processing and perception model (CASP; Jepsen et al., 2008). CASP employs outer- and middle-ear filtering, a non-linear auditory filterbank (DRNL, López-Poveda & Meddis, 2001), adaptation loops, and a modulation filterbank, and has previously been shown to successfully account for psychoacoustic data of normal-hearing (NH) listeners in conditions of, e.g., spectral masking, amplitude-modulation detection, and forward masking (Jepsen et al., 2008). Furthermore, it has been demonstrated that CASP can be tuned to account for data measured in individual hearing-impaired listeners in various behavioral experiments (Jepsen & Dau, 2011). In this study, the potential of the CASP model for predicting speech intelligibility in NH listeners was investigated. The cross-correlation between CASP's internal representations of the degraded speech and the clean speech was used as the decision metric. Predictions of speech intelligibility obtained with the speech-based CASP are presented and compared to predictions obtained with other intelligibility models as well as data in conditions of additive noise, phase jitter, spectral subtraction, ideal binary mask processing and reverberation. The proposed model framework sets the foundations to investigate consequences of hearing loss on speech intelligibility.

Corresponding author: Helia Relaño-Iborra (heliaib@elektro.dtu.dk)

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Pediatric hearing aid use: A study based on data logging information

Mina Salamatmanesh*,S, Elizabeth Fitzpatrick *- School of Rehabilitation Science, Faculty of Health Science, University of Ottawa, Ottawa, ON, Canada*

Tim Ramsay *- Ottawa Methods Centre (OMC) & School of Epidemiology, Faculty of Health Science, University of Ottawa, Ottawa, ON, Canada*

Josée Lagacé *- School of Audiology and Speech-Language Pathology, Faculty of Health Science, University of Ottawa, Ottawa, ON, Canada*

Lindsey Sikora *- Health Science Library, Faculty of Health Science, University of Ottawa, Ottawa, ON, Canada*

JoAnne Whittingham *- Child Hearing Lab Children Hospital of Eastern Ontario, Ottawa, ON, Canada*

A particular challenge for parents in the early years is achieving consistent hearing aid (HA) use which is critical to the child's development and constitutes the first step in the rehabilitation process. This study examined the consistency of hearing aid use in young children based on data logging information in the first three years after hearing aid fitting. The first 100 children who were diagnosed with bilateral hearing loss (HL) before 72, with least two HA evaluation sessions included in the study. Data from each audiology session (age, average hours of use, etc.) and complementary information were collected. Preliminary statistical analysis showed the median hours of use in the first follow-up session is about 4h per day and 47% of the children use the HA ≤5h a day. However, by the third follow-up session the median of use increased to 9h per day and 38% of children used the HAs ≥10h. Factors like age and level of HL significantly impact the hours of use. The use of data logging information to assess the actual hours of HA can be a powerful tool to identify problems and to encourage and assist families in maximizing their child's hearing potential.

Corresponding author: Mina Salamatmanesh (ssala098@uottawa.ca)

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Data-driven approach for auditory profiling

Raul H. Sanchez*, Federica Bianchi, Michal Fereczkowski *- Hearing Systems, Department of Electrical Engineering, Technical University of Denmark, Kgs. Lyngby, Denmark*
Sébastien Santurette *- Hearing Systems, Department of Electrical Engineering, Technical University of Denmark, Kgs. Lyngby, Denmark; Department of Otorhinolaryngology, Head and Neck Surgery & Audiology, Rigshospitalet, Copenhagen, Denmark*
Torsten Dau *- Hearing Systems, Department of Electrical Engineering, Technical University of Denmark, Kgs. Lyngby, Denmark*

Nowadays, the pure-tone audiogram is the main tool used to characterize hearing loss and to fit hearing aids. However, the perceptual consequences of hearing loss are typically associated not only with a loss of sensitivity, but also clarity loss that is not captured by the audiogram. Detailed characterization of hearing loss has to be simplified to efficiently investigate the specific compensation needs of individual listeners. We hypothesized that any listeners' hearing can be characterized along two dimensions of distortion: type I and type II. While type I can be linked to factors affecting audibility, type II reflects non-audibility-related distortions. To test our hypothesis, the individual performance data from two previous studies was re-analyzed using archetypal analysis. Unsupervised learning was used to identify extreme patterns in the data which form the basis for different auditory profiles. Next, a decision tree was determined to classify the listeners into one of the profiles. The new analysis provides evidence for the existence of four profiles in the data. The most significant predictors for profile identification were related to temporal processing, loudness growth, and speech perception. The current approach is promising for analyzing other existing data sets in order to select the most relevant tests for auditory profiling.

Corresponding author: Raul Sanchez (rsalo@elektro.dtu.dk)

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Influence of a remote microphone on localization with hearing aids

Johan G. Selby *- Hearing Systems, Department of Electrical Engineering, Technical University of Denmark; GN Resound A/S, Ballerup, Denmark*
Adam Weisser *- GN Resound A/S, Ballerup, Denmark; Department of Linguistics, Maquarie University, North Ryde, Australia*
Ewen N. MacDonald* *- Hearing Systems, Department of Electrical Engineering, Technical University of Denmark*

When used with hearing aids (HA), the addition of a remote microphone (RM) may alter the spatial perception of the listener. First, the RM signal provides no interaural level or time differences. Second, the processing in the HA often delays the RM signal relative to the HA microphone signals. Finally, the level of the RM signal is independent of the distance from the RM to HA. The present study investigated localization performance of 15 normal and 9 hearing-impaired listeners under conditions simulating the use of an RM with a behind the ear (BTE) HA. Minimum audible angle discrimination at an average angle of 45° was measured for three sets of relative gains and seven sets of relative delays for a total 21 conditions. In addition, a condition with just the simulated BTE HA signals was tested. Overall, for both groups, the minimum audible angle discrimination was best when the relative RM gain was small (-3 and -6 dB) and the delay was approximately 10-20 ms. Under these conditions, localization performance approached the level obtained in the BTE HA only condition.

Corresponding author: Ewen MacDonald (emcd@elektro.dtu.dk)

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Preliminary investigation of the categorization of gaps and overlaps in turn-taking interactions: effects of noise and hearing loss

Anna Josefine Sørensen* *- Hearing Systems, Department of Electrical Engineering, Technical University of Denmark, Kgs. Lyngby, Denmark*

Adam Weisser *- GN Resound A/S, Ballerup, Denmark; Macquarie University, Sydney, Australia*

Ewen N. MacDonald *- Hearing Systems, Department of Electrical Engineering, Technical University of Denmark, Kgs. Lyngby, Denmark*

Conversational turn taking requires interlocutors to monitor the ongoing acoustic signal to judge when it is appropriate to start talking. However, the perception of cues that listeners use to judge turn-ends is likely to be affected by noise and/or hearing loss. In the present study, categorical thresholds for gaps and overlaps in turn-taking interactions were measured for normal-hearing and hearing-impaired listeners in both quiet and multitalker babble (+6 dB signal-to-noise ratio). For both groups of listeners, the categorical thresholds for both gaps and overlaps increased by approximately 30 ms in the presence of noise. The thresholds for overlaps and gaps were similar between both groups. However, the categorization functions of the hearing impaired listeners were shallower.

Corresponding author: Anna Josefine Sørensen (ajs@elektro.dtu.dk)

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Methods for non-intrusive objective intelligibility prediction

Charlotte Sørensen* *- Research, GN Hearing A/S, Ballerup, Denmark; Audio Analysis Lab, Department of Architecture, Design and Media Technology, Aalborg University, Aalborg, Denmark*
Jesper Bünsow Boldt *- Research, GN Hearing A/S, Ballerup, Denmark*
Mads Græsbøll Christensen *- Audio Analysis Lab, Department of Architecture, Design and Media Technology, Aalborg University, Aalborg, Denmark*

Automatic adjustment of the signal processing according to the speech intelligibility in the specific listening scenario could be beneficial for hearing aid users. Most speech intelligibility metrics are intrusive, i.e., they require a clean reference signal, which is rarely available in real-life applications. This work proposes new non-intrusive intelligibility prediction methods, which allow using an intrusive short-time objective intelligibility (STOI) metric without requiring access to a clean signal. The principle of all the proposed methods is to replace the clean speech reference signal with an estimation of the relevant features representing the clean speech signal. The difference between the proposed methods lies in how the reference signal of the clean signal is estimated using, respectively, the spatial content, the fundamental frequency content or the envelope content to reconstruct the clean signal. The simulations show a high correlation between the proposed non-intrusive methods and the intrusive STOI indicating that proposed non-intrusive STOI methods are suitable for automatic classification of speech signals.

Corresponding author: Charlotte Sørensen (csoerensen@gnresound.com)

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Preferred listening levels - a silent disco study

***Rikke Sørensen**[*,S]* - *University of Southern Denmark, Odense, Denmark*
Elizabeth Beach, Megan Gilliver - *National Acoustic Laboratories, Sydney, Australia*
Carsten Daugaard - *Delta, Odense, Denmark*

This poster reports from two experiments on preferred listening levels (PLL). The first experiment was a laboratory setup evaluating PLL for 13 different pieces of music from both headphones (HP) and loudspeakers (LS). A reference song was repeated to evaluate the participants' consistency in choosing PLL. For 4 iterations of the reference song, participants varied 0.8-19.1 dB in expressed PLL. HP in general displayed higher PLL than LS. The second experiment was a "Silent disco" event where the disco music was played through personal headphones that each participant could adjust individually. 22 of the 59 participants (37%) chose levels of 72-86 dB, 37 (63%) chose the loudest volume setting (89-93 dB). 19 of the 37 were pleased with that level, only 18 indicating preference for louder volume settings. The experiments indicate that 1) one person's PLL may vary up to 19 dB even within minutes. It is therefore recommended making repeated measures when testing PLL; 2) PLL (and likely loudness perception) changes with presentation mode. Therefore, care should be taken when transferring loudness assumptions from a free field listening situation to a HP listening situation; 3) many people show PLL lower than the 90-105 dB typically offered at regular discos, even when correcting for differences in presentation mode.

Corresponding author: Rikke Sørensen (tegnsprog@gmail.com)

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Estimating auditory filter bandwidth using distortion product otoacoustic emissions

Sigurd van Hauen*[S]*, Andreas Harbo Rukjær*, Rodrigo Ordoñez, Dorte Hammershøi *- Section for Signal and Information Processing, Department of Electronic Systems, Aalborg University, Aalborg, Denmark*

The basic frequency selectivity in the listener's hearing is often characterized by auditory filters. These filters are determined through listening tests, which determine the masking threshold as a function of frequency of the tone and the bandwidth of the masking sound. The auditory filters have been shown (Glasberg and Moore, 1986) to be wider for listeners with sensorineural impairment. In a recent study (Christensen et al., 2017) it was demonstrated on group basis that the distortion product stimulus ratio that provided the strongest 2f1-f2 component at low frequencies, had a strong correlation to the theoretical relation between frequency and auditory filter bandwidth, described by the equivalent rectangular bandwidth (ERB, Glasberg and Moore, 1990). The purpose of the present study is to test if a similar correlation exists on an individual basis at normal audiometric frequencies. The optimal 2f1-f2 (L1/L2=65/45 dB SPL). The distortion-product-otoacoustic-emission ratio is determined using a custom-made measurement system programmed in MATLAB. The auditory filters are determined using notched-noise method in a two alternative forced choice experiment with noise levels at 40 dB SPL/Hz. Optimal ratios and auditory filters are determined at 1, 2, and 4 kHz for 10 young normal-hearing subjects.

Corresponding author: Sigurd van Hauen (smalle11@student.aau.dk)

P.54 – Thu 24 Aug, 17:00-19:00

Adjusting expectations: Hearing abilities in a population-based sample using a SSQ short-form

Petra von Gablenz* *- Institute of Hearing Technology and Audiology, Jade University of Applied Sciences and Cluster of Excellence "Hearing4All", Oldenburg, Germany*

Fabian Sobotka *- Department of Health Services Research, School for Medicine and Health Sciences, Carl von Ossietzky University Oldenburg, Oldenburg, Germany*

Inga Holube *- Institute of Hearing Technology and Audiology, Jade University of Applied Sciences and Cluster of Excellence "Hearing4All", Oldenburg, Germany*

Self-reports of hearing (dis)abilities play an important role in hearing rehabilitation. Among the large variety of questionnaires, the Speech, Spatial, and Qualities of Hearing Scale (SSQ) has become an internationally used measure to assess hearing abilities in specified everyday listening situations using a visualized scale ranging from 0 to 10. Research mainly focused on adults with impaired hearing and adults with "normal" hearing were hardly considered. However, the ratings of adults out of the general population could be of particular interest when it comes to the question of score benchmarks based on different definitions of "normal" hearing. In the cross-sectional, population-based study HÖRSTAT (n=1903) the German SSQ17 short-form was used along with a standardized interview and comprehensive hearing examinations. As the SSQ score distributions are extremely positively skewed, semiparametric quantile and expectile regression analysis was performed to examine the conditional score distribution and the effects of age, gender, globally reported hearing problems, hearing loss, and social status. Though no normative cut-off values can be established from empirical findings only, this contribution addresses how the knowledge of "normal" hearing abilities might improve the management of expectations during the process of hearing rehabilitation.

Corresponding author: Petra von Gablenz (petra.vongablenz@jade-hs.de)

P.55 – Wed 23 Aug, 17:00-19:00

National Better hEAring Rehabilitation (BEAR) project: A status on the database with special focus on patients' motivation on hearing aid treatment

Anne Wolff,S* - *Department of Otolaryngology, Head & Neck Surgery and Audiology, Aalborg University Hospital, Aalborg, Denmark*
Sabina Storbjerg Houmoeller - *Department of ENT/Audiology, Odense University Hospital, Odense, Denmark*
Michael Gaihede, Dan Hougaard - *Department of Otolaryngology, Head & Neck Surgery and Audiology, Aalborg University Hospital, Aalborg, Denmark; Department of Clinical Medicine, Aalborg University Hospital, Aalborg, Denmark*
Jesper Hvass Schmidt - *Department of ENT/Audiology, Odense University Hospital, Odense, Denmark*
Dorte Hammershøi - *Department of Electronic Systems, Aalborg University, Aalborg, Denmark*

It is estimated that around 250,000 people (DK) own a hearing aid (HA) and that about 20% do not use their hearing aids regularly. The reasons for this are not understood, but it is clear that these 20% do not perceive the benefit of their devices to be sufficient. The goal of the BEAR is to improve hearing rehabilitation through an update of clinical practice. As a part of the BEAR project a clinical database has been created to document the current hearing aid treatment of hearing impaired people in Denmark. The database will collect knowledge of different aspects that may prove to be important for user satisfaction. Data are collected from consecutive clinical data, from the Dept. of Audiology Odense and Aalborg University Hospital, and from several additional questionnaires. As a part of the survey the patients are asked to score their motivation to acquire a HA. With the use of the database it is possible to analyze whether one or more of patient stand out compared to others. A status on demographic data compared with motivational scores will be presented. Whether HA-motivation of the individual patients has an essential role in user satisfaction is not well investigated. With this project a broad variety of variables will be investigated and the aim is to identify clinically relevant subgroups which previous studies have not been able to succeed in.

Corresponding author: Anne Wolff (a.wolff@rn.dk)

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Effects of non-stationary noise on consonant identification

Johannes Zaar*, Borys Kowalewski, Torsten Dau *- Hearing Systems, Department of Electrical Engineering, Technical University of Denmark, Kgs. Lyngby, Denmark*

Consonant perception has typically been measured using consonant-vowel (CV) syllables presented in a stationary noise masker at various signal-to-noise ratios (SNRs). Recently, a consonant perception model was proposed (Zaar and Dau, 2017) and shown to account well for consonant perception data obtained in stationary noise. However, unlike stationary noise, real-life interfering sounds typically exhibit strong fluctuations. The present study therefore investigated the effects of highly non-stationary noise on consonant perception and assessed the predictive power of the consonant perception model in such conditions. 15 Danish CVs were presented in 5-Hz interrupted noise at SNRs of -20, -10, 0, and 10 dB. Five different CV starting times with respect to the noise bursts were considered, differing in the amount of simultaneous and forward masking induced. As expected, the consonant recognition scores were inversely related to the amount of simultaneous masking. However, even with minimum simultaneous masking, a substantial loss of consonant recognition was observed at low SNRs, suggesting a forward masking effect. The model, which employs adaptive processes in the front end, could account for these experimental data to a large extent and may thus be useful for assessing temporal effects of hearing-aid algorithms on consonant perception, e.g., slow- vs. fast-acting compression.

Corresponding author: Johannes Zaar (jzaar@elektro.dtu.dk)

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N.B. In the above programme, * denotes a presenting author and ^s a scholarship recipient.

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