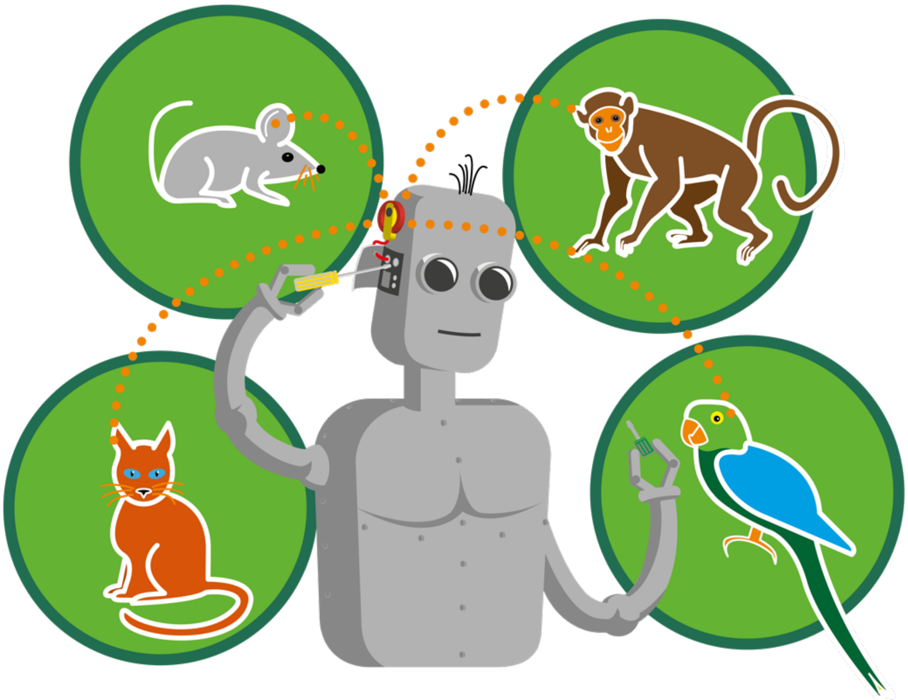
7th

International Symposium on Auditory and Audiological Research

ISAAR 2019

"Auditory Learning in Biological and Artificial Systems"



August 21-23, 2019
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 2019 edition corresponds to the 28th symposium in the series and the 7th symposium under the ISAAR name, which was 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.

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with thanks to Eva Helena Andersen

Welcome to ISAAR 2019

The general topic of the ISAAR 2019 symposium is "Auditory Learning in Biological and Artificial Systems". 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 four sessions, to which the following speakers have been invited:

1. ***Auditory precision medicine***
   Jeffrey Holt, Thomas Lenarz, Hartwig Siebner, Josef Schlittenlacher, Dorte Hammershøi

2. ***Learning from natural sounds***
   Alex Kell, Wiktor Młynarski, Frederic Theunissen, Jennifer Linden

3. ***Machine listening and intelligent auditory signal processing***
   Björn Schuller, Tuomas Virtanen, Malcolm Slaney, Bernd Meyer

4. ***Novel directions in hearing-instrument technology***
   Patrick Naylor, Thomas Lunner, Maarten De Vos, Tobias May

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 an online version in the ISAAR 2019 proceedings.

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

Wednesday 21 August

08:30–09:30	**Registration and hanging of posters**
09:30-10:00	**Welcome** Introduction by TORSTEN DAU
10:00-11:00	**Session 1a: Auditory precision medicine**
10:00-10:30	JEFFREY HOLT Function, dysfunction and restoration of hair cell transduction channels
10:30-11:00	THOMAS LENARZ Auditory precision medicine using novel diagnostic methods and prediction models
11:00-11:20	**Coffee break**
11:20-12:00	**Session 1b: Auditory precision medicine**
11:20-11:40	BIRGER KOLLMEIER Towards precision medicine in audiology: Modelling unaided and aided speech recognition with clinical data
11:40-12:00	SARINEH KESHISHZADEH From derived-band envelope-following responses to individualized models of near- and supra-threshold hearing deficits
12:00-13:30	**Lunch**
13:30-15:00	**Session 1c: Auditory precision medicine**
13:30-14:00	HARTWIG SIEBNER Precision stimulation medicine: A conceptional framework
14:00-14:30	JOSEF SCHLITTENLACHER The implementation of efficient hearing tests using machine learning
14:30-15:00	DORTE HAMMERSHØI Highlights from the better hearing rehabilitation (BEAR) project in Denmark
15:00-17:00	**Poster Session I (with coffee)** Odd-numbered posters

17:00-17:50 **Session 2a: Learning from natural sounds**

17:00-17:20 ANDREW OXENHAM
Auditory learning and plasticity through musical training: How far does it generalize?

17:20-17:50 ALEX KELL
Task-optimized deep neural networks as models of auditory cortex and behavior

17:50-18:10 **Break**

18:10-19:00 **Session 2b: Learning from natural sounds**

18:10-18:30 FRANCOIS DELOCHE
The fine-grained statistical structure of speech may be congruent with nonlinear cochlear filtering

18:30-19:00 WIKTOR MŁYNARSKI
Ecological origins of perceptual grouping principles in the auditory system

19:00-20:30 **Dinner**
Presentation during dinner by PÄR THURESSON

20:30-21:00 **Anniversary celebration**
Presentation by KURT JACOBSEN
GN's 150-year journey from telegraph lines to intelligent auditory solutions

21:00-23:00 **Social and posters**

Thursday 22 August

08:30–08:40 **Coffee**

08:40-10:00 **Session 2c: Learning from natural sounds**

08:40-09:10 JENNIFER LINDEN
Fundamental mechanisms of change detection in the auditory system: A theory of gap detection

09:10-09:40 FREDERIC THEUNISSEN
Neural computations underlying categorical responses for communication signals in the auditory cortex

09:40-10:00 KAREN BANAI
The incredible robustness and specificity of perceptual learning for speech: Implications for auditory rehabilitation

10:00-10:30 **Coffee break**

10:30-12:00 **Session 3a: Machine listening and intelligent auditory signal processing**

10:30-10:50 SARAH VERHULST
Normal and hearing-impaired cochlear models for real-time hearing-aid and machine-hearing applications

10:50-11:10 BRIAN C. J. MOORE
Using a deep neural network to speed up a model of loudness for time-varying sounds

11:10-11:30 MARLIES GILLIS
Measuring speech understanding from the EEG using semantic features

11:30-12:00 BERND T. MEYER
Deep machine listening for modeling speech intelligibility

12:00-13:30 **Lunch**

13:30-15:20 **Session 3b: Machine listening and intelligent auditory signal processing**

13:30-14:00 BJÖRN W. SCHULLER
Call for attention: Audio Intelligence++

14:00-14:20 TIMO OESS
Computational investigation of visually guided learning of spatially aligned auditory maps in the colliculus

	14:20-14:50	TUOMAS VIRTANEN Computational recognition of everyday sound scenes and events
	14:50-15:20	MALCOLM SLANEY Intelligent personal hearing for everybody
15:20-17:20		**Poster Session II (with coffee)** Even-numbered posters
17:20-18:30		**Session 4a: Novel directions in hearing-instrument technology**
	17:20-17:50	TOBIAS MAY Scene-aware dynamic range compression in hearing aids
	17:50-18:10	JENS BREHM BAGGER NIELSEN Machine-learning based hearing-aid personalization: Preference optimizer, preference sensor or both?
	18:10-18:30	SIMON LANSBERGEN Classification of hearing aids into feature profiles using hierarchical latent class analysis applied to a large dataset of hearing aids
18:30-19:00		**Anniversary celebration** Presentation by TORBEN POULSEN 50 years with The Danavox Jubilee Foundation: The foundation behind the Danavox Symposia and ISAAR
19:00-20:30		**Dinner**
20:30-23:00		**Social and posters**

Friday 23 August

08:30–08:40 **Coffee**

08:40-10:00 **Session 4b: Novel directions in hearing-instrument technology**

08:40-09:00 WILLIAM M. WHITMER
Adaptation to hearing-aid microphone modes in a dynamic localisation task

09:00-09:30 THOMAS LUNNER
Towards intention-controlled hearing aids: Experiences from eye-controlled hearing aids

09:30-10:00 PATRICK A. NAYLOR
Improving Robustness of Adaptive Beamforming for Hearing Devices

10:00-10:30 **Coffee break**

10:30-12:00 **Session 4c: Novel directions in hearing-instrument technology**

10:30-11:00 MAARTEN DE VOS
Auditory attention decoding in real life: Challenges and opportunities

11:00-11:20 ANDREW DITTBERNER
Machine learning for humans

11:20-11:40 TOBIAS NEHER
Effects of directional hearing aid processing and motivation on EEG responses to continuous noisy speech

11:40-12:00 SEPP CHALUPPER
Optimizing spatial perception for bimodal users

12:00-12:20 **Conclusion**
Closing remarks by TORSTEN DAU

12:20-13:50 **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"). It is about 134 km from Copenhagen Airport (CPH), and it takes approximately 1½ hours to travel from the airport to the hotel 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. Please plan your journey accordingly.

About the weather

The weather in Denmark is unpredictable. Day temperatures can vary between 15 and 25 degrees centigrade, there are frequent showers, and it is often windy. See www.dmi.dk for the current forecast.

Practical Information

Posters

Hanging of posters: Wed 21 Aug 08:30-09:30

Presenters of odd-numbered posters are encouraged to be present at their poster during Poster Session I, and presenters of even-numbered posters are encouraged to be present at their poster during Poster Session II. Posters will remain on display throughout the symposium to allow further interaction outside these dedicated sessions.

Talks

Dedicated time with assistance for uploading slides and running technical tests in the auditorium:

Wed 21 Aug 08:30-09:30 and 15:00-15:15
Thu 22 Aug 15:20-15:35

A PC with PowerPoint software will be available in the auditorium. It will also be possible to connect your own laptop.

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. The proceedings will 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 *proceedings.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 2019.

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 the open-access journal Trends in Hearing (see *http://tia.sagepub.com/*).

Trends in Hearing remains the only fully open-access journal that specializes 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 2019. 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 2020. Articles will be published and added to the special issue shortly after acceptance.

Session 1

Auditory precision medicine

Chairs: Bastian Epp & Lisbeth Tranebjærg

Session 1: Auditory precision medicine

S1.01 – Wed 21 Aug, 10:00-10:30

Function, dysfunction and restoration of hair cell transduction channels

Jeffrey R Holt* *– Departments of Otolaryngology & Neurology, Harvard Medical School, Boston, MA, USA*

Transmembrane Channel-Like One (TMC1) forms the pore of hair cell transduction channels (Pan et al., Neuron, 2018). TMC1 is also one of the more common hair cells genes that causes hearing loss when mutated. Recently, our lab has focused on the basic biology of TMC1 function with particular attention devoted to TMC1 point mutations that cause dysfunction. These studies have yielded insight into structure-function relationships for TMC1. Additional insight has led to development of novel gene therapy approaches for preservation and restoration of auditory function in mice that carry TMC1 mutations. For this presentation, I will discuss our most recent data and insights into the function, dysfunction and restoration of TMC1 channels in auditory and vestibular hair cells. Our goal is to develop auditory precision medicine approaches for patients with genetic inner ear disorders, including those with mutations in TMC1.

Corresponding author: Jeffrey R Holt (jeffrey.holt@childrens.harvard.edu)

S1.02 – Wed 21 Aug, 10:30-11:00

Auditory precision medicine using novel diagnostic methods and prediction models

Thomas Lenarz*, Max Eike Timm, Rolf Salcher, Waldo Nogueira, Andreas Büchner *– Clinic for Laryngology, Rhinology and Otology, Hannover Medical School, Germany; Cluster of Excellence "Hearing4all", Hanover, Germany*

Every hearing impaired patient has a unique hearing loss in terms of type, grade, etiology, pathophysiology and time cause. Current methods of hearing rehabilitation provide a wide spectrum of treatment options. The goal of precision medicine is to determine and propose the most appropriate method for optimized actual and future treatment. Indications for different rehabilitation strategies will be based on advanced diagnostic methods for pretreatment evaluation, including audiometric tests, advanced imaging and molecular diagnostics. This will be further supported by artificial intelligence including machine learning and big data. In terms of individualized cochlear implantation, most candidates for a cochlear implant today have some degree of residual hearing. Prediction can be used for electric stimulation only or for electro-acoustic stimulation. Based on the frequency range of residual hearing and the individual cochlear anatomy, the appropriate electrode and insertion depth can be determined. Intraoperative cochlear monitoring using electrocochleography supports hearing preservation cochlear implantation. The concept of partial insertion of long electrodes allows to adept the electrical stimulation to progressive hearing loss by after loading of the electrode. Post operative cochlear functional analysis with electrical and acoustic stimulation allows to determine the best stimulation strategy for optimized hearing rehabilitation. Precision medicine in audiology will be further refined by advances in functional as well as morphological analysis of the auditory system. Atraumatic robot-assisted and model based implantation will improve hearing preservation. Cochlear analyzer will allow to determine the functional properties along the cochlear partition for best possible stimulation of residual cochlear function for automated multi parametric fitting.

Corresponding author: Thomas Lenarz (lenarz.thomas@mh-hannover.de)

S1.03 – Wed 21 Aug, 11:20-11:40

Towards precision medicine in audiology: Modelling unaided and aided speech recognition with clinical data

Birger Kollmeier*, Anna Warzybok, David Hülsmeier, Marc-René Schädler *– Medizinische Physik & Cluster of Excellence Hearing4All, Universität Oldenburg*

To assess speech recognition performance in a precise and multilingual way, the matrix sentence recognition test has been made available in more than 20 languages so far (see Kollmeier et al., Int. J. Audiol. 2015). For its modelling and individual performance prediction, the Framework for Auditory Discrimination Experiments (FADE, Schädler et al., JASA 2016) is employed using the audiogram and one suprathreshold performance parameter that reflects Plomp´s D-factor. 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. In a set of 19 hearing-impaired listeners the average prediction error is reduced to 4,6 dB if an individual estimate of suprathreshold distortion is employed. The current contribution investigates the prediction performance for a large clinical data set (315 ears from Wardenga et al., 2015) with or without accounting for Plomp´s D factor in comparison to the SII: FADE clearly outperforms the SII by reducing the prediction error from 7 to 4 dB. The application of this approach to predict the individual performance without and with a hearing device will be discussed for a variety of noise conditions, different degrees of hearing impairment, and various hearing instruments.

Corresponding author: Birger Kollmeier (birger.kollmeier@uol.de)

S1.04 – Wed 21 Aug, 11:40-12:00

From derived-band envelope-following responses to individualized models of near- and supra-threshold hearing deficits

Sarineh Keshishzadeh*, Sarah Verhulst *– Hearing Technology @ WAVES, Information Technology Department, Ghent University, Ghent, Belgium*

Auditory models which include frequency-dependent profiles of near and supra-threshold hearing deficits can aid the design of effective and individualized hearing-aid algorithms. However, determining individual auditory-nerve (AN) fiber loss parameters is controversial as diagnostic metrics are based on auditory brainstem responses (ABRs) or envelope following responses (EFRs) which are not necessarily frequency-specific and are affected by both outer-hair-cell (OHC) and AN damage. To render EFRs more frequency specific, we developed a derived-band EFR technique which adopts modulated broadband noise stimuli with different low cut-off frequencies. We recorded OAEs, ABRs, and constructed derived-band EFRs from normal-hearing listeners. We used these data to derive OHC and AN damage parameters which were inserted in individual auditory models. We set the cochlear parameters using the CEOAE and audiometric data, after which different AN deficit profiles were introduced and EFRs simulated. We adopted a clustering technique to determine the best match between the experimental and simulated EFRs to yield the frequency-specific AN fiber loss pattern which best explained the pattern of results. The technique was validated using the ABR data collected from the same listeners. The combined experimental and model approach shows promise in offering more sensitive hearing loss profiles.

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Session 1: Auditory precision medicine

S1.05 – Wed 21 Aug, 13:30-14:00

Precision stimulation medicine: A conceptional framework

Hartwig Roman Siebner *– Danish Research Centre for Magnetic Resonance (DRCMR), Copenhagen University Hospital Hvidovre, Hvidovre, Denmark*

Brain diseases are the core health challenge of the 21st century. This challenge will grow steadily with population aging. Most, if not all, brain diseases are "circuit" or "network" disorders in which a failure of specific brain circuits determines the expression and evolution of functional impairments in individual patients. Circuit alterations often differ from patient to patient and they are non-stationary, creating substantial within- and between-patient variability. No circuit-specific therapies currently exist that can individually target abnormal brain circuit function, leading to a therapeutic road-block in the treatment of brain disorders in recent years. Several non-invasive brain stimulation technologies are available which can shape neural integration within and between brain circuits at a level of specificity that surpasses any other existing non-invasive therapy. Yet current stimulation-based therapies use a one-size-fits-all approach, ignoring the individual variation in circuit dysfunction and lacking circuit specificity. In my talk, I will argue for a brain-circuit centered approach to advance precision medicine for brain diseases. Merging preclinical research with state-of-the art mapping, modelling, and modulation of brain circuits, one can leverage the full potential of stimulation-based therapies through precise and individually optimized stimulation.

Corresponding author: Hartwig Roman Siebner (h.siebner@drcmr.dk)

S1.06 – Wed 21 Aug, 14:00-14:30

The implementation of efficient hearing tests using machine learning

Josef Schlittenlacher*, Richard E. Turner, Brian C. J. Moore –
Department of Psychology, University of Cambridge, Cambridge, UK

Time-efficient hearing tests are important in both clinical practice and research studies. Historical approaches include Békésy tracking. Machine-learning (ML) methods were first proposed in the 1990s. We developed ML methods for measuring the audiogram, conducting notched-noise tests, determination of the edge frequency of a dead region (fe), and estimating equal-loudness contours. The methods all use a probabilistic model of the outcome, which can be classification (audible/inaudible), regression (loudness) or model parameters (fe, outer hair cell loss at fe). The stimulus parameters for the next trial (e.g. frequency, level) are chosen automatically to yield maximum reduction in the uncertainty of the parameters of the probabilistic model. The approach reduced testing time by a factor of about 5 and, for some tests, yielded results on a continuous frequency scale. For example, auditory filter shapes can be estimated from 500 to 4000 Hz in about 15 minutes, providing information that may allow more personalized fitting of a hearing aid. The probabilistic modelling allows comparison of different methods quantitatively. For audiogram determination, counting the number of audible tones in a sequence with decreasing level was slightly better than responding Yes/No. Counting tones yields higher variance for a single response, but this is offset by the higher information per trial.

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S1.07 – Wed 21 Aug, 14:30-15:00

Highlights from the better hearing rehabilitation (BEAR) project in Denmark

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

The BEAR project runs from 2016-2021, and includes several scientific efforts, incl. the collection of data for almost 2.000 patients fitted according to current practice, development and assessment of new diagnostics for profiling and fitting strategies, as well as development and assessment of methods for measurement of the aided performance. The on-going work includes a proposal for a differentiated fitting based on extended auditory profiles and is accompanied by both in- and out-of-clinic options for testing and/or reporting on the aided performance experience. Future work will include an experimental validation of the proposed differentiated fitting, as well as a separate effort to investigate common denominators for patients with poor compensation benefits and options for out-of-clinic application of the proposed methods.

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Collaboration and support by Innovation Fund Denmark (Grand Solutions 5164-00011B), Oticon, GN Resound, Widex and other partners (University of Southern Denmark, Aalborg University, the Technical University of Denmark, Force, and Aalborg, Odense and Copenhagen University Hospitals) is sincerely acknowledged.

Session 2

Learning from natural sounds

Chairs: Jeremy Marozeau & Sarah Verhulst

Session 2: Learning from natural sounds

S2.01 – Wed 21 Aug, 17:00-17:20

Auditory learning and plasticity through musical training: How far does it generalize?

Andrew J. Oxenham* – *Department of Psychology, University of Minnesota, Minneapolis, MN, USA*

Studies over the past decade have suggested that musical training confers benefits that extend beyond the realm of music, that it may enhance the neural coding and perception of speech in noise, and that it may therefore protect against some of the effects of aging on understanding speech in noisy backgrounds. Here we explore the possible roots of these benefits in terms of enhanced pitch coding and perception, and present data that question some of the earlier conclusions. The new results, along with a critical review of the existing literature, suggest that a large-scale effort is required to determine the reproducibility of the original neurophysiological and perceptual studies.

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

S2.02 – Wed 21 Aug, 17:20-17:50

Task-optimized deep neural networks as models of auditory cortex and behavior

Alexander J.E. Kell* – *Zuckerman Institute, Columbia University, New York, NY, USA; Department of Brain and Cognitive Sciences, MIT, Cambridge, MA, USA; Center for Brains, Minds, and Machines, Cambridge, MA, USA*
Josh H. McDermott – *Department of Brain and Cognitive Sciences, MIT, Cambridge, MA, USA; Center for Brains, Minds, and Machines, Cambridge, MA, USA; Program in Speech and Hearing Biosciences and Technology, Harvard University, Cambridge, MA, USA; McGovern Institute for Brain Research, MIT, Cambridge, MA, USA*

A complete model of the human auditory system would perform auditory tasks as well as humans do. Motivated by this idea, we sought improved models of auditory cortex by optimizing deep neural networks for multiple real-world recognition tasks. Despite not being trained to fit behavioral or neural data, the resulting model replicated human auditory behavior, predicted neural responses, and revealed a cortical processing hierarchy, with distinct network layers mimicking response in distinct parts of auditory cortex. Next, to test the generality of such networks as models of the human auditory system, we simulated an extensive set of psychophysical and fMRI experiment on the network. The observed similarities between the network and human suggest that deep neural networks can replicate a variety of aspects of human audition and the apparent discrepancies suggest targets for future modeling efforts.

Corresponding author: Alexander J.E. Kell (alex.kell@columbia.edu)

S2.03 – Wed 21 Aug, 18:10-18:30

The fine-grained statistical structure of speech may be congruent with nonlinear cochlear filtering

Francois Deloche* *– Centre d'analyse et de mathematique sociales, PSL University, Paris, France*

The speech signal presents regularities that can be exploited by biological or artificial systems for efficient coding. Independent Component Analysis (ICA) revealed that cochlear frequency selectivity reflects the overall statistical structure of speech, in line with the hypothesis that low-level sensory processing provides efficient codes for information contained in natural stimuli. Recently, this result has been refined with the demonstration that variations in frequency selectivity would provide additional adaptation to phonetic categories. In this work, I introduce a parametric model to further describe the relationship between the acoustic properties of speech and an optimal signal decomposition. The parameter of this model is the exponent of the power law satisfied by the quality factor in the high frequency range 1-8kHz. This parameter, which controls frequency selectivity, is estimated by evaluating the sparseness of decompositions in Gabor dictionaries, whose atoms are Gaussian-modulated sinusoids. The distribution of the power-law exponent at the level of phonemes provides a rich interpretation of the statistical structure of speech. The results show that an efficient strategy is to reduce frequency selectivity with sound intensity level in high frequencies, reflecting the nonlinear behavior of the cochlea.

Corresponding author: Francois Deloche (francois.deloche@ehess.fr)

S2.04 – Wed 21 Aug, 18:30-19:00

Ecological origins of perceptual grouping principles in the auditory system

Wiktor Młynarski* – *Institute of Science and Technology Austria*

The brain infers events and objects in the environment from sensory signals. Because sensory measurements are temporally and spatially local, the estimation of an object or event can be viewed as the grouping of these measurements into representations of their common causes. Perceptual grouping is believed to reflect internalized regularities of the natural world, yet grouping cues have traditionally been identified using informal observation. Here we derive auditory grouping cues by measuring and summarizing statistics of natural sound features. Feature co-occurrence statistics reproduced established cues but also revealed previously unappreciated grouping principles. The results suggest that auditory grouping is adapted to natural stimulus statistics, show how these statistics can reveal novel grouping phenomena, and provide a framework for studying grouping in natural signals.

Corresponding author: Wiktor Młynarski (wmlynars@ist.ac.at)

S2.05 – Thu 22 Aug, 08:40-09:10

Fundamental mechanisms of change detection in the auditory system: A theory of gap detection

Jennifer F. Linden* *– Ear Institute, University College London, London, UK*

Humans are remarkably sensitive to brief interruptions of ongoing sound. Duration thresholds for detection of a brief gap in noise are only 3-6 ms in normal young adults. Gap-detection thresholds are typically longer in older adults, patients with developmental disorders, or people with auditory processing difficulties, and are therefore used in the clinic as a measure of auditory temporal acuity. However, the neural mechanisms of gap detection remain poorly understood. In this talk, I will argue that at near-threshold gap durations, gap-in-noise detection is both perceptually and neurophysiologically best described as a "not-noise" detection – a basic form of change detection. Neurophysiological recordings in an unusual mouse model with elevated gap-detection thresholds reveal that (i) auditory brainstem responses to sound offsets (disappearances) play a key role in defining the limits of gap-in-noise acuity, and (ii) adaptive gain control in the auditory thalamus and cortex augments gap-in-noise sensitivity. The results illustrate how diverse mechanisms of temporal processing throughout the central auditory system combine to enable detection of changes in stimulus statistics. Moreover, the findings suggest that elevated gap-detection thresholds in patients with auditory perceptual difficulties could arise from abnormalities in several different auditory brain areas.

Corresponding author: Jennifer F. Linden (j.linden@ucl.ac.uk)

S2.06 – Thu 22 Aug, 09:10-09:40

Neural computations underlying categorical responses for communication signals in the auditory cortex

Frederic Theunissen* – *Department of Psychology and Neuroscience, UC Berkeley, USA*

Our studies in both birds and humans use natural sounds and communication signals produced in natural contexts in combination with quantitative modeling approaches to decipher the neural computations needed for the sound to meaning transformation. We have found neurons in avian secondary auditory pallial areas that respond in a categorical fashion to calls belonging to a particular call type of the zebra finch vocal repertoire. These categorical responses required selectivity for a particular call type and invariance to the natural variations within these categories. To understand how these categorical responses can emerge in the ascending auditory pathway, we used unsupervised agglomerative clustering on the responses of ensembles of auditory neurons and compared those results to an unsupervised agglomerative clustering applied to an acoustical representation of the calls. We used the results of this analysis to quantify the extent with which non-linear processing enhances the separation of acoustical clusters that correspond to different natural call-types. In addition to performing this single neuron analysis in the avian auditory system, we have examined how speech signals are represented in the human cortex. Using nested encoding models, we show that the sound to meaning computations for human speech involve invariant representations to combination of phonemes.

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S2.07 – Thu 22 Aug, 09:40-10:00

The incredible robustness and specificity of perceptual learning for speech: Implications for auditory rehabilitation

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

The use of auditory training for hearing rehabilitation is limited by the specificity of the learning it yields. For the past decade, we have used the perceptual learning of time-compressed speech to study the conditions that may broaden the transfer of learning. This work yielded three major findings. First, brief exposure (< 30 sentences) induces substantial improvements in recognition, which are highly specific to the acoustics of the trained stimuli. Second, more extensive practice may broaden the scope of transfer to untrained stimuli, but only in normal hearing listeners, and only to stimuli that share some acoustic or semantic properties with the trained stimuli. Third, the transfer of learning is negatively correlated with individual differences in language and hearing acuity. Together, these findings suggest that the high-level requirements of the training tasks elicit learning that fails to abstract over the acoustic characteristics of the trained stimuli, unless stimuli are repeated, in line with the predictions of the Reverse Hierarchy Theory. That the transfer of learning is further penalized in individuals with poorer hearing acuity implies that in order to support hearing rehabilitation, training may need to include more varied acoustics and more stimulus repetition during training. Whether such training results in greater transfer remains to be seen.

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Session 3

Machine listening and intelligent auditory signal processing

Chairs: Petteri Hyvärinen & Andrew Oxenham

S3.01 – Thu 22 Aug, 10:30-10:50

Normal and hearing-impaired cochlear models for real-time hearing-aid and machine-hearing applications

Arthur Van Den Broucke, Deepak Baby, Sarah Verhulst* – *Hearing Technology @ WAVES, Dept of Information Technology, Ghent University, BE*

Biophysically realistic models of the cochlea are based on cascaded transmission-line (TL) models which capture longitudinal coupling, cochlear nonlinearities, as well as the human frequency selectivity. However, these models are slow to compute (on the order of seconds/minutes) while machine-hearing and hearing-aid applications require a real-time solution (in ms). Consequently, real-time applications often adopt more basic and less time-consuming descriptions of cochlear processing (e.g., gammatone, DRNL, CARFAC and MFCC models) even though there are clear advantages in using more biophysically correct models. To overcome this, our study combines nonlinear Deep-Neural Nets (DNN) with nonlinear TL cochlear models to build a real-time model of the cochlea which captures the biophysical properties associated with the TL model. The DNN model was trained using a speech dataset at a fixed sound level, but performed well on a set of basic auditory stimuli of various stimulus levels and frequencies to assess the coupling, tuning, nonlinearity and distortion properties of the new model. The normal-hearing DNN model was adjusted to simulate frequency-specific patterns of cochlear gain loss, yielding a set of normal and hearing-impaired models which can be computed in real-time, are differentiable, and can serve the next generation of hearing-aid and machine hearing applications.

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S3.02 – Thu 22 Aug, 10:50-11:10

Using a deep neural network to speed up a model of loudness for time-varying sounds

Brian C. J. Moore* – *Department of Experimental Psychology, University of Cambridge, Downing Street, Cambridge, UK*
Richard E. Turner – *Department of Engineering, University of Cambridge, Cambridge, UK*
Josef Schlittenlacher – *Department of Experimental Psychology, University of Cambridge, Downing Street, Cambridge, UK*

A model for the loudness of time-varying sounds, the TVL model, has been proposed as a new ISO standard, ISO 532-3. The model calculates "instantaneous loudness" every 1 ms, and this is used to generate predictions of short-term loudness, the loudness of a short segment of sound such as a word in a sentence, and of long-term loudness, the loudness of a longer segment of sound, such as a whole sentence. The calculation of instantaneous loudness is computationally intensive and real-time implementation of the TVL model is difficult. To speed up the computation, a deep neural network (DNN) has been trained to predict instantaneous loudness using a large database of speech sounds and artificial sounds (tones alone and tones in white or pink noise), with the predictions of the TVL model as a reference (providing the "correct" answer, specifically the loudness level in phons). A simple multilayer perceptron with three hidden layers was found to be sufficient, with more complex DNN architecture not yielding higher accuracy. After training, the deviations between the predictions of the TVL model and the predictions of the DNN were typically less than 0.5 phons, even for types of sounds that were not used for training (music, rain, animal sounds, washing machine). The DNN calculates instantaneous loudness over 100 times more quickly than the TVL model.

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S3.03 – Thu 22 Aug, 11:10-11:30

Measuring speech understanding from the EEG using semantic features

Marlies Gillis**, Jonas Vanthornhout, Damien Lesenfants, Tom Francart
– ExpORL, Department of Neurosciences, KU Leuven, Leuven, Belgium

In previous work we have shown that a person's speech understanding can be measured from the EEG. We record EEG while the person listens to speech and build a model to predict the EEG signal based on features of the stimulus. The extent to which the model can predict the EEG signal is an indication of neural coding of the speech signal and relates with speech understanding. In this study, we aim to further improve the estimation of a person's speech understanding by integrating stimulus features that relate to semantic information.
Via EEG-data of 19 subjects listening to speech in noise at different signal-to-noise ratios (SNRs) and a forward, grand average model, the EEG response is predicted based on features of the stimulus. These features represent low-level acoustical features (e.g. envelope) or higher-level speech representations (e.g. phoneme or semantic contextual information). We investigated the model's prediction accuracy as a function of included features and stimulus SNR. Preliminary results indicate that EEG responses predicted from semantic contextual information correlate with the actual EEG responses. Additionally, we evaluate the effect on the estimated speech understanding with respect to the behaviorally measured speech understanding. We hypothesize that combining low- and high-level features increases the accuracy of the estimated speech understanding.

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S3.04 – Thu 22 Aug, 11:30-12:00

Deep machine listening for modeling speech intelligibility

Angel Mario Castro Martínez, Constantin Spille, Birger Kollmeier, Bernd T. Meyer* – *Carl von Ossietzky Universität Oldenburg*

In this work, we explore a model for predicting the perceived speech intelligibility, which is based on automatic speech recognition (ASR) using a deep neural network (DNN). The model has previously been shown to accurately predict the speech reception threshold (SRT) of normal-hearing listeners: Although it does not require the clean speech reference, it outperforms four baseline models when considering a wide range of maskers for subjects who listened to matrix sentence tests. However, the model requires the true word labels which are compared to the ASR transcript to produce the word error rate (WER). This form of a priori knowledge prevents this ASR-based model from being used in real-world applications, e.g., as an online monitor for SI. We therefore investigate if the WER can be estimated directly from the phoneme probabilities obtained from the DNN, omitting the actual recognition step. The probabilities should result in sparse, clear patterns in clean condition, and be noisier or generally degraded in acoustically difficult situations. We test the entropy-based mean temporal distance as a measure to quantify this degradation, and find a very similar prediction performance based on this estimate, thereby dropping the requirement for word labels.

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S3.05 – Thu 22 Aug, 13:30-14:00

Call for attention: Audio Intelligence++

Björn W. Schuller* *– GLAM - Group on Language, Audio, & Music, Imperial College London, London, UK; ZD.B Chair of Embedded Intelligence for Health Care and Wellbeing, University of Augsburg, Augsburg, Germany; audEERING GmbH, Gilching, Germany*

Computer Audition has made major progress over the last decades, but still seems to be far from human hearing abilities. Imagine, for example, the sound of a water glass put onto a table. As humans, we would be able to roughly "hear" the material of the glass, the table, and perhaps even how full the glass is. Current machine listening, on the other hand, would mainly recognise the event of "glass put onto a table". In this context, this contribution aims to provide on the one hand insight into the already made remarkable advances in computer audition, in particular in speech analysis, where machines can already hear better than humans our heart rate, or how tall we are to name but two examples. On the other hand, it highlights deficits in reaching human-alike hearing abilities such as in the given example. Moreover, after summarising the state-of-the-art in traditional signal-processing-based audio pre-processing and feature representation as well as automated learning such as by deep neural networks, avenues are given towards reaching the ambitious goal of "holistic human-parity" machine listening abilities – the "Audio Intelligence++". This includes improved attention models throughout modelling, automated machine learning for the shaping of optimised learning architectures, and reinforced learning in real-life usage context.

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S3.06 – Thu 22 Aug, 14:00-14:20

Computational investigation of visually guided learning of spatially aligned auditory maps in the colliculus

Timo Oess*, Marc O. Ernst *– Applied Cognitive Psychology, University of Ulm, Ulm, Germany*
Heiko Neumann *– Institute of Neural Information Processing, University of Ulm, Ulm, Germany*

Visually guided development of spatially registered auditory maps and their maintenance in adult animals is a dynamic process for which the underlying neural mechanisms are yet unresolved. To investigate these mechanisms, we developed a mechanism of stabilized Hebbian correlative learning which is augmented by an eligibility signal and a temporal trace of activations. This 3-component learning algorithm facilitates stable yet flexible formation of spatially registered auditory space maps composed of conductance-based topographically organized neural units. Spatially aligned maps are learned for visual and auditory input stimuli that arrive in temporal and spatial registration. We show that by shifting visual sensory inputs the topography of auditory space maps is shifted accordingly but can be realigned after restoring normal visual input. Simulation results explain why a shift of the auditory maps in matured animals is still possible only if corrections are induced in small steps. The reliability of visual sensory inputs (given by the receptive field width) can be used to regulate learning rate in the form of an eligibility trace. We conclude that learning aligned auditory maps is flexibly controlled by reliable visual sensory neurons and can be formalized by a 3-component learning mechanism. The maintenance of registered auditory maps even in a matured state is demonstrated.

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S3.07 – Thu 22 Aug, 14:20-14:50

Computational recognition of everyday sound scenes and events

Tuomas Virtanen* – *Tampere University, Tampere, Finland*

This presentation will give an overview of recent machine learning approaches that have been automatically used to recognize everyday sound scenes and events. We will first describe how computational analysis of everyday soundscapes can be formulated as acoustic scene classification and sound event detection tasks. Then we will present a generic machine learning methodology based on convolutional recurrent neural networks that has been successfully applied to address these tasks in many different contexts. We will discuss how data required for training machine learning models can be obtained. We will present results and demonstrations of the methods, and also compare their performance to results from human listening tests. We will also discuss advanced methods, such as the use of weak labels without temporal annotation, and transfer learning, which can be used to partly deal with the difficulty of obtaining large annotated datasets for training acoustic models.

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S3.08 – Thu 22 Aug, 14:50-15:20

Intelligent personal hearing for everybody

Malcolm Slaney*, Richard F. Lyon – *Google Research, Machine Hearing*

Deep neural networks (DNNs) understand and react to speech in novel and amazing ways. Current speech recognizers fit into our pocket and perform well, all due to deep networks trained on thousands of hours of speech. This technology enables real-time transcriptions, in dozens of languages for the deaf and hard of hearing. Speech-enhancement networks know what speech sounds like and filter out the non-speech sounds. When combined with visual control signals, they "solve" the cocktail party in a way that none of us would have predicted. New EEG decoding methods, perhaps with sensors just near the ear, allow us to think about new ways to control our auditory environment. Finally, today's neural-network accelerator chips consume very little power. Today they can be powered by a phone battery, but maybe someday an even smaller battery. All of us have limited hearing, and new personal technology can make it easier to understand the auditory world around us. What would you like a personal auditory device to do for you?

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Session 4

Novel directions in hearing-instrument technology

Chairs: Ewen MacDonald & Brian C. J. Moore

Session 4: Novel directions in hearing-instrument technology

S4.01 – Thu 22 Aug, 17:20-17:50

Scene-aware dynamic range compression in hearing aids

Tobias May* *– Hearing Systems Section, Department of Health Technology, Technical University of Denmark, DK-2800 Kgs. Lyngby, Denmark*

Wide dynamic range compression aims at improving audibility while maintaining acceptable loudness at high sound pressure levels for hearing-impaired listeners. While fast-acting compression with a short release time allows to amplify low-intensity speech sounds on short time scales corresponding to phonemes, such processing also typically amplifies noise components in speech gaps and disrupts the acoustic properties of the background noise. Some of these shortcomings can be avoided by using a longer release time, but such a slow-acting compression system compromises on the ability to restore loudness perception. This contribution describes a scene-aware dynamic range compression strategy, which attempts to combine the advantages of both fast- and slow-acting compression. Specifically, the release time of the compressor is adapted in individual time-frequency units to provide fast- or slow-acting compression depending on whether the target is present or absent. The benefit of this scene-aware compression strategy was evaluated in two acoustic scenarios, where the target signal was either processed in the presence of interfering noise or room reverberation. The evaluation included both objective measures, such as the relative change in the signal-to-noise ratio, as well as subjective measures, reflecting speech intelligibility, speech quality and the perceived spatial location.

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S4.02 – Thu 22 Aug, 17:50-18:10

Machine-learning based hearing-aid personalization: Preference optimizer, preference sensor or both?

Jens Brehm Bagger Nielsen*, Lasse Lohilahti Mølgaard, Niels Søgaard Jensen, Laura Winther Balling *– Widex A/S, Lynge, Denmark*

Personalization is an increasingly important aspect of modern hearing-aid (HA) solutions, which may be tailored to the individual end user via parameter adjustments. However, traditional HA fitting and fine-tuning cannot fully exploit these capabilities and therefore cannot utilize the HA's full potential in special real-life situations relevant to the particular end user. We present a recently launched machine-learning based approach to HA personalization. The approach involves an iterative series of paired comparisons of different HA parameter settings made by the end user, and it treats the user-preference assessments using probabilistic non-parametric models that learn under uncertainty. The approach provides full probabilistic prediction of the end user's preference, which enables use of active learning to only probe user-preference assessments that provide maximal information. Thereby, the approach discovers the end user's optimal HA setting as fast as possible. The approach can be viewed both as a preference optimizer and a preference sensor in the hands of end users in real life. We will present both views, explaining the technology of the preference optimizer and present results based on data gathered during one year's real-life use of the preference sensor. Finally, we will discuss where this might take HA personalization going forward.

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S4.03 – Thu 22 Aug, 18:10-18:30

Classification of hearing aids into feature profiles using hierarchical latent class analysis applied to a large dataset of hearing aids

Simon Lansbergen*, Wouter Dreschler *– Clinical and Experimental Audiology, Amsterdam UMC: Univ of Amsterdam, Amsterdam, The Netherlands*

Today's hearing care professionals and audiologists can choose from an overwhelming number of different types and brands of hearing aids (HA's). HA manufacturers try to distinguish themselves from each other by using their own terminology describing comparable HA characteristics; making HA comparison a complicated task. Currently, there is not a well-defined method to compare common HA features between different types and brands. Technical data from 3911 HA's were used to develop a framework for an objective comparison between HA's, independent of brand, type, or product family. Finally, 10 HA features were selected to be relevant for compensation of audiological deficits. A novel exploratory cluster analysis approach called Latent Class Tree (LCT) analysis was used to discover sub-populations of HA's. The LCT analysis yielded a set of 11 mutually exclusive HA populations, called modalities. These modalities were characterized by particular profiles of the included features. The resulting HA modalities could be thought of as a generic alternative to the manufacturer dependent concepts and could potentially support the selection of an appropriate HA for technical rehabilitation. This study fits well in a growing need for justification of HA selection and the increasing demand for evidence-based practice.

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S4.04 – Fri 23 Aug, 08:40-09:00

Adaptation to hearing-aid microphone modes in a dynamic localisation task

William M Whitmer* *– Hearing Sciences – Scottish Section, University of Nottingham, Glasgow, UK*
Nadja Schinkel-Bielefeld *– WS Audiology Group GmbH, Erlangen, DE*
David McShefferty *– Hearing Sciences – Scottish Section, University of Nottingham, Glasgow, UK*
Cecil Wilson *– WS Audiology Group GmbH, Erlangen, DE*
Graham Naylor *– Hearing Sciences – Scottish Section, University of Nottingham, Glasgow, UK*

New technology can require new ways of listening. A new hearing aid programme can alter how we hear not only sources of sound but their locations. While previous research has established how different hearing aid types and microphone modes affect static localisation ability, the current study explored the effects of introducing unfamiliar devices and microphone modes – on dynamic localisation ability. Twelve experienced bilateral behind-the-ear (BTE) hearing-aid users oriented themselves as quickly and accurately as comfortable to 5-s segments of a target talker in a continuous background of far-field babble of the same overall level. Targets were presented from angles centred at ±30, ±75 and ±120°. Head-orientation trajectories were measured with infra-red cameras. Participants first wore their own hearing aids for one block of 60 trials, then wore a new hearing aid and completed five more blocks in three directional-microphone modes. In general, results showed trajectory differences between modes, but only a modest influence of the preceding mode (i.e., adaptation). Two additional participants with in-the-ear hearing aids oriented poorly with the new BTE device for the first two blocks, then returned to their baseline performance. This suggests that such a form-factor change requires additional time for spatial adaptation.

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Session 4: Novel directions in hearing-instrument technology

S4.05 – Fri 23 Aug, 09:00-09:30

Towards intention-controlled hearing aids: Experiences from eye-controlled hearing aids

Thomas Lunner* *– Eriksholm Research Centre, Oticon A/S, Denmark; Linköping University, Department of Electrical Engineering, Department of Behavioural Sciences; Technical University of Denmark, Department of Health Technology*

A hearing impairment causes a reduced ability to segregate acoustic sources. This gives problems to switch between and to follow speech streams in complex scenes with multiple talkers. Current hearing aid beamforming technologies rely on a listener's ability to point with the head towards a source of interest. However, this is very difficult in a conversation situation with spatially separated talkers where rapid switches between talkers takes place. In this talk I will show that eye-gaze position signals can be picked up electrically in the ear canal through electrooculography, and that these signals can be used for fast intentional eye-gaze control towards the source of interest in a complex listening scene like a restaurant. Experiments where eye-gaze signals are combined with motion sensors and beamformers show that high benefits in form of improved speech intelligibility is possible for the hearing-impaired listeners. Results also indicate that eye-control combined with head movements is faster and more precise than head movements alone. The presentation will include several videos to show the use cases.

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S4.06 – Fri 23 Aug, 09:30-10:00

Improving robustness of adaptive beamforming for hearing devices

Patrick A. Naylor* – *Imperial College London, UK*
Alastair Moore – *Square Set Sound, London, UK*
Mike Brookes – *Imperial College London, UK*

Beamforming is an almost essential component in hearing devices that enables a selected region of the acoustic sound field to be presented to the user while sounds from outside the selected region are attenuated. Whereas fixed beamforming is suboptimal but robust, hearing devices that implement adaptive beamforming are less robust in general but can advantageously adapt the beam-pattern dynamically as the sound field around the user changes. There is a significant risk in practice however that the beamformer may degrade the signal if the acoustic properties used in the adaptive beamformer design are incorrectly estimated. This problem is prevalent at low SNRs. In this talk, the problem of adaptive beamforming will be discussed in terms of noise covariance matrix estimation and a technique for constraining the estimation of this matrix will be presented that leads to robust adaptive beamforming. Simulation results will be presented for behind-the-ear hearing aids taking into account some of the practical challenges of real-world scenarios including reverberation and noise. Numerical results and listening examples will be given to show robustness to simultaneous interfering speakers, babble and spatially white noise.

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S4.07 – Fri 23 Aug, 10:30-11:00

Auditory attention decoding in real life: Challenges and opportunities

Maarten De Vos* *– Department of Engineering, University of Oxford, Oxford, UK*
Rob Zink *– Department of Electrical Engineering, KU Leuven, Leuven, Belgium*

Auditory attention is widely recognized as a very important concept that plays a vital role in the way humans process auditory information. In recent years there has been overwhelming evidence that it is possible to decode auditory attention from invasive and non-invasive brain recording modalities. Those breakthroughs allowed dreaming that decoding auditory attention in real life would become possible. However, most decoding approaches ignore challenges that might impact its feasibility in a real-life scenario: convenient and inobtrusive recording equipment usable in natural environments, impact of noise and reverberation, real-time processing limitations, just to name a few. In this talk, we will go deeper into those challenges and highlight some promising directions to translate auditory attention decoding into real-life applications and reveal critical factors that might influence neurophysiological brain responses as described in traditional confined EEG experiments and thus impact decoding performance.

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S4.08 – Fri 23 Aug, 11:00-11:20

Machine learning for humans

Andrew Dittberner*, Jon Boley, Erik Van Der Werf, Bert de Vries *– GN Hearing*

As machine learning techniques have evolved over the past few decades, we have often used them to optimize hearing aid algorithms, parameters, and configurations. Adaptive and online learning algorithms are already used for tasks such as acoustic scene analysis, feedback control, source separation, and steering. As new opportunities arise, we have to consider how people will interact with intelligent systems and how they might learn from each other. In this presentation, we will discuss some of the challenges of learning from users – like gathering sufficient relevant data, not overloading the user, and providing meaningful opportunities for enhanced experiences. We will also discuss some of the challenges that can arise when users learn to use machines – like adopting non-natural behaviors to achieve specific goals. New technologies for hearing-instruments are being designed with these interactions in mind to maximize the benefits and minimize the user experience costs.

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S4.09 – Fri 23 Aug, 11:20-11:40

Effects of directional hearing aid processing and motivation on EEG responses to continuous noisy speech

Bojana Mirkovic, Stefan Debener *– Department of Psychology, University of Oldenburg, Oldenburg, Germany*
Tobias Neher* *– Institute of Clinical Research, University of Southern Denmark, Odense, Denmark*

Arguably, devices that can infer (or learn) the user's needs via non-invasive physiological measurements such as electroencephalography (EEG) and adjust themselves accordingly are the next frontier in hearing aid (HA) development. A promising approach to translating EEG signals into HA control signals is the analysis of EEG impulse responses to running speech, as obtained by cross-correlating the audio stimulus with the concurrently recorded EEG signal. Here, we used this method for examining neural correlates of the effects of directional HA processing and listener motivation on speech comprehension in noise. Groups of older participants with normal or impaired hearing listened to an audiobook embedded in realistic cafeteria noise while their EEG was recorded using mobile hardware. A HA simulator was used for providing individual amplification and for (dis)engaging a directional microphone setting. Motivation was manipulated by offering a monetary reward for good speech comprehension in half of the trials. Motivation influenced the behavioral performance of the hearing-impaired listeners, but not the EEG responses. Directional HA benefit was reflected in the behavioral performance and EEG responses of both groups, thereby illustrating the potential of the tested approach for enabling automatic HA adjustments.

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S4.10 – Fri 23 Aug, 11:40-12:00

Optimizing spatial perception for bimodal users

Sepp Chalupper*, Iris Arweiler *– European Research Center, Advanced Bionics, Hannover, Germany*
Sven Kliesch, Andreas Büchner *– Department of Otolaryngology, Medical School Hannover, Hannover, Germany*

CI recipients using a contralateral hearing aids usually perform much poorer in experiments regarding localization and spatial release from masking than bilateral HA users, bilateral CI users and normal listeners. This finding can be at least partially explained by the fact that these patients oftentimes do not have enough aided high frequency audibility to efficiently use ILDs and cannot use ITDs because of envelope-based coding strategies in the CI. Introducing artificial low-frequency ILDs by employing interaural beamforming should help to improve localization and increase the head shadow effect. In an acute experiment, localization performance increased significantly with the interaural beamformer activated. In order to assess the head shadow effect, speech reception threshold was measured for speech from the HA-side and noise from the CI-side and vice versa. All subjects had a benefit in at least one condition. Benefit in localization and speech perception was further increased by aligning loudness growth functions across ears.

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Poster Sessions I and II

Posters will remain on display throughout the symposium.

SP.01 – Wed 21 Aug, 15:00-17:00

Assessing daily-life benefit from hearing aid noise management: SSQ12 vs. ecological momentary assessment

Line Storm Andersen*, Klaudia Edinger Andersson, Mengfan Wu – *Institute of Clinical Research, University of Southern Denmark, Odense, Denmark*
Niels Pontoppidan, Lars Bramsløw – *Eriksholm Research Centre, Snekkersten, Denmark*
Tobias Neher – *Institute of Clinical Research, University of Southern Denmark, Odense, Denmark*

Assessing daily-life benefit of noise management (NM) in hearing aids is a challenge in audiological research. Ecological momentary assessment (EMA) using smartphone-connected hearing aids has recently emerged as a promising tool for real-life data acquisition. However, there is a lack of research linking this method to commonly used questionnaires, such as the SSQ12. In the current study, 12 hearing-impaired participants were asked to assess two hearing aid fittings using the SSQ12 questionnaire and a smartphone-based EMA-method combining soundscape logging with momentary self-reports. The two fittings differed in terms of their NM settings (omnidirectional with noise reduction off vs. directional with noise reduction on). The participants were aged 23-75 years and had different occupations and lifestyles. The testing period for each fitting was 2 weeks. Overall, the EMA and SSQ scores were higher for the setting with active NM, but the differences were not significant. Analysis of the soundscape data showed that only some participants experienced noisy surroundings frequently. Future work on EMA-based hearing aid assessment should therefore address the interplay between the tested hearing aid features and the auditory ecology of the user.

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SP.02 – ~~Thu 22 Aug, 15:20-17:20~~ (cancelled)

Neurophysiologic plasticity in geriatric brain after hearing aid use: An electrophysiologic investigation

Mohammad Shamim Ansari* *– Department of Audiology, AYJNISHD (D), Mumbai, India*

The literature reveals that hearing aids (HA) can improve speech understanding and suggests that auditory system reorganizes the auditory structures with the use of HA. However, it is unknown how the neurophysiologic auditory mechanisms change with amplified sound exposure. Therefore, the study aims to examine physiological changes in the auditory brainstem in geriatric population after HA usage. 16 geriatric with average age of 62.5 years, having bilateral moderate SNHL who had no experience of HA use were examined with click-evoked auditory brainstem response (ABR) to determine absolute latency and amplitude and interpeak latencies at threshold and 20dB above threshold level. The subjects were recommended monaural (right ear) HA and divided into study group (SG) and control group (CG). The SG used HAs for a period of 4 months whereas the CG did not use HAs during this period. Both groups were reexamined with ABR after minimum 4 months of HA use and electrophysiological measures were calculated and compared. The study did not find significant change in the absolute latency and amplitude of ABR wave across the groups. However, the mean amplitude of wave V at threshold level and at 20 dBSL was approximately 100μV larger in the ear that used hearing aid for 4 months. The study suggests that HA use can encourage changes and prevent deprivation at brainstem in geriatric population.

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Poster Session I

SP.03 – Wed 21 Aug, 15:00-17:00

Subjective loudness ratings of vehicles noise with the hearing aid fitting methods NAL-NL2 and trueLOUDNESS

Dirk Oetting *– HörTech, Oldenburg, Germany; Cluster of Excellence Hearing4all, Oldenburg, Germany*
Jörg-Hendrik Bach*, Melanie Krueger *– HörTech, Oldenburg, Germany; Hörzentrum Oldenburg, Oldenburg, Germany; Cluster of Excellence Hearing4all, Oldenburg, Germany*
Matthias Vormann, Michael Schulte *– Hörzentrum Oldenburg, Oldenburg, Germany; Cluster of Excellence Hearing4all, Oldenburg, Germany*
Markus Meis*– HörTech, Oldenburg, Germany; Hörzentrum Oldenburg, Oldenburg, Germany; Cluster of Excellence Hearing4all, Oldenburg, Germany*

The individual loudness perception plays a decisive role for fitting hearing aids and is one of the most important criterions for the overall satisfaction with hearing aids. Listeners with similar hearing thresholds showed large differences in loudness summation of binaural broadband signals after narrow-band loudness compensation. The effect has been well described in Oetting et al. (Hearing Res. 2016, Ear Hearing 2018). Based on these findings, the fitting method trueLOUDNESS was developed to restore the individual binaural broadband loudness perception. The aim of this study was to show that the lab measurements of loudness scaling for the trueLOUDNESS fitting corresponds to the real-world loudness perception with hearing aids. Loudness ratings of 19 listeners with hearing loss and hearing aids fitted according to NAL-NL2 and trueLOUDNESS were compared. The study was conducted at a closed road and the subjects rated the perceived loudness of four different vehicles in various driving conditions. For eight listeners the gain predictions with trueLOUDNESS were lower compared to NAL-NL2 and for 11 listeners the gains were higher with trueLOUDNESS compared to NAL-NL2. Results of loudness ratings of trueLOUDNESS and NAL-NL2 fitting compared to normal-hearing listeners will be presented.

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SP.04 – Thu 22 Aug, 15:20-17:20

Exploring simultaneous assessment of cortical and subcortical entrainment to ongoing speech

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Speech processing relies on fine temporal coding at the level of the auditory brain stem as well as at later cortical stages. Comparing subcortical responses with cortical measures provides a deeper understanding of the processing along the auditory pathway. Generally, this interplay has been investigated using different stimuli. Cortical activity is often assessed with ongoing speech, while brainstem responses are traditionally investigated with repetitions of simple stimuli such as clicks. However, recent research has shown that subcortical responses can also be measured using ongoing speech stimuli, with potential applications both in hearing-loss research and clinical diagnostics. These new approaches further suggest that it should be possible to simultaneously observe neural entrainment at the cortical and subcortical level to the same ongoing speech stimuli, which is explored in the present study. At the subcortical level, two previously suggested, complementary approaches are tested. One directly computes a linear transfer function between the filtered auditory input and the measured EEG via regularized regression (Maddox & Lee, 2018). The other identifies an empirical mode that oscillates at the fundamental frequency in the auditory signal, and cross-correlates the EEG with this fundamental waveform (Forte, Etard, & Reichenbach, 2017).

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SP.05 – Wed 21 Aug, 15:00-17:00

Hearing examinations in Southern Denmark (HESD): Database description and preprocessing

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Disabling hearing loss is one of the most common sensory disorder worldwide, affecting more than 5% of the global population. Despite the high incidence of hearing loss, there is a general lack of large sample sized data necessary for a better understanding of the risk factors and mechanisms underlying this condition and whether hearing impairment has a direct effect on other outcomes (e.g. cognitive functioning). From 1996 beyond, all hearing examinations from the public system of the region of Southern Denmark have been electronically recorded in the AuditBase system. All gathered information have been recently merged into a single database, entitled the HESD (Hearing Examinations in Southern Denmark) database, which contains hearing information of over than 141,000 adults and therefore consists in a promising source of audiology-related epidemiological data. Besides information on pure-tone thresholds, this database also contains data on speech detection, acoustic reflexes and tympanometry. The use of this dataset, however, needs to be preceded by an intensive preprocessing procedure. This study is aimed at describing the HESD database, as well as all preprocessing steps and rules used to classify different types of hearing loss. It is expected that a better knowledge on this database can be of great interest for researchers within the fields of audiology and epidemiology.

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SP.06 – Thu 22 Aug, 15:20-17:20

The vent effect in instant eartips and its impact on the fitting of modern hearing aids

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Today, 70-80% of hearing aid fittings are made with instant ear tips. This may be due to ease of fit, improved physical comfort and the reduction in occlusion compared to custom earmolds. Hearing aid manufacturers provide different types of eartips for a range of hearing losses. They can be completely open, vented, closed, or have a form that provides natural venting/leakage depending on its coupling to the ear canal. The acoustical properties of the eartip will depend on its type, size, and the way it fits into the ear canal of the individual. Depending on the resulting direct transmitted sound (DTS) and vent effect (VE), the sound quality and aided benefit provided by the hearing aid may vary among individuals fitted with the same eartip type. This study explored five eartips from one manufacturer in relation to DTS and VE by using Real Ear Measurements and subjective occlusion ratings. Thirty normal-hearing subjects (sixty ears), 20 males and 10 females with a mean age of 45 years (±11) participated in the study. The results showed large variation in DTS and VE both between eartips and across subjects with the same eartip types. This poster will present the study findings and discuss some important implications in relation to fitting strategies and the importance of assessing the acoustics of instant eartips as part of the hearing aid fitting process.

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SP.07 – Wed 21 Aug, 15:00-17:00

Objective correlates of tinnitus and cochlear synaptopathy

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Tinnitus is often connected to a hearing loss and a damage of the auditory system with a maladaptation leading to the perception of a phantom sound. There exist, however, cases where audiologically normal hearing listeners suffer from tinnitus. Studies in animal models showed that extensive noise exposure with a subsequent temporal threshold shift leads to a loss of synaptic connections to the auditory nerve, referred to as hidden hearing loss (HHL). HHL in connection with a gain regulation can lead to overrepresentation of specific frequencies in a computational model. In this study, elements of a screening and characterization battery for tinnitus based on subjective and objective measures are optimized in terms of sensitivity and test time. In particular, the estimation of psychoacoustical tuning curves, tinnitus masking curves and high-frequency audiometry is optimized using Bayesian statistics and compared to methods from the literature. In addition, a clinical method for wide band tympanometry is optimized to improve sensitivity using ipsilateral noise stimulation in combination with ear canal pressurization. The applicability in a clinical environment will provide a basis for the electrophysiological evaluation of listeners prone to HHL and to the identification of a subset of tinnitus characterized by a certain set of acoustical and electrophysiological biomarkers.

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SP.08 – Thu 22 Aug, 15:20-17:20

Limitations of sound localization with linear hearing devices

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Limited abilities to localize sound sources and other reduced spatial hearing capabilities remain a largely unsolved issue in hearing aids and comparable devices. Hence, the impact of several linear factors that potentially disturb localization was assessed in a subjective listening experiment with normal-hearing listeners, including the microphone location, the frequency response as well as processing delays in superposition with direct sound leaking through a vent. Using binaural synthesis and individually measured transfer functions for 6 device styles and microphone positions, it was possible to assess these aspects separately. Both horizontal- and median plane localization was tested with short white noise bursts. In our data, the microphone location is the governing factor for localization abilities with linear hearing devices. Non-optimal microphone locations have a disruptive influence on localization in the vertical domain, and a detrimental effect on lateral sound localization. Processing delays cause additional detrimental effects for lateral sound localization; and diffuse-field equalization to the open-ear response should be preferred over free-field equalization. The median-plane localization results are in line with predictions from computational models of vertical-plane localization and allow a comparison and refinement of concurrent model versions.

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SP.09 – Wed 21 Aug, 15:00-17:00

The comparison of the Finnish Digit Triplet Test with the Finnish Matrix Sentence Test in cochlear implant recipients

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The number of cochlear implant (CI) recipients is rapidly increasing due to expanding candidacy criteria, technical improvements and demographics. Tele-audiology with self-testing and CI remote programming could reduce costs on the health-care system and patients. The Digit Triplet test (DTT) is designed as a self-administered hearing screening test and is, therefore, well suited for tele-audiology applications. Currently there are still little data on the correlation of DTT results to the results of clinicaly used tests, such as the Finnish Matrix Sentence Test (FMST). The aim of this study was to compare the characteristics of the recently developed FDTT with the FMST, which represents the golden standard for measuring CI performance in Finland. 83 CI recipients were measured in the best-aided condition with the FMST and the FDTT. Similar to normal-hearing listeners, the speech reception thresholds (SRT) for the FDTT were significantly better (i.e. more negative) than for the FMST. The mean SRTs were -3.9 dB (SNR) (-7.7- +3.8 dB; SD 2.4 dB) and -4.5 dB (SNR) (-8.2- +1.1 dB, SD 2.0 dB) for the FMST and the FDTT, respectively. There was a strong correlation between the FMST and FDTT results (R2=0.61). More detailed analysis of the effect of age and performance will be presented. The preliminary data suggest that the FDTT is reliable for monitoring performance in CI recipients.

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SP.10 – Thu 22 Aug, 15:20-17:20

Comparison of clinical feasibility of behavioural and physiological estimates of peripheral compression

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Previous research has shown that the rate of peripheral compression (estimated from the slope of the basilar-membrane input-output function) is not correlated with the pure-tone sensitivity (the audiogram). However, efficient estimation of peripheral compression has proven challenging and the methods are based on several assumptions. The aim of this study was to investigate and compare results from three methods of estimating peripheral compression in terms of their accuracy and clinical feasibility. Two psychoacoustic behavioural measures, based on forward (temporal masking curves) and simultaneous (notched-noise) masking were investigated, together with a physiological, otoacoustic-emissions based measure. 45 hearing-impaired listeners with mild-to-moderate hearing loss and 5 normal-hearing listeners were tested. Correlation analysis of the data was performed, including partial-correlations, in order to factor out the potential influence of the pure tone-thresholds on the compression estimates. The results demonstrated limitations of each of the considered methods; however, the experiment involving estimates of auditory filters showed good stability and small training requirements across the listeners.

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SP.11 – Wed 21 Aug, 15:00-17:00

Individual hearing aid benefit: Ecological momentary assessment of hearing abilities

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Laboratory hearing tests aim at creating reproducible results in controlled acoustic environments. Though considerable effort is made to approximate natural communication scenarios, real-life listening demands are still imperfectly established. More importantly, testing rarely captures the individual's point of view. In the course of hearing aid (HA) fitting, therefore, questionnaires are used to address the subjective perspective on hearing abilities. Weaknesses of this approach, e.g., memory bias and possible mismatch of the pre-defined and individually experienced listening situations, are tackled by Ecological Momentary Assessment (EMA), an ad-hoc query including prompt and repeated assessments in real-life. We conducted an EMA study to examine how HA uptake changes the perception of everyday hearing abilities. First-time and follow-up HA wearers used a smartphone-based EMA system for 3-4 full days before HA fitting and after HA acclimatization. This EMA system allows for specifying situations and sound sources as well as for assessing various aspects like speech understanding, loudness, and listening effort. Respecting data privacy, acoustical parameters are simultaneously stored. This contribution analyzes the impact of HA uptake on self-assessed hearing abilities at an individual level and discusses the potential of EMA in the HA fitting process.

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SP.12 – Thu 22 Aug, 15:20-17:20

Relationships between objective and behavioural markers of suprathreshold hearing and speech-in-noise intelligibility

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Suprathreshold hearing is important for communication in challenging listening environments. Even people with normal audiometric thresholds can have degraded suprathreshold hearing as a consequence of aging/noise-exposure. We investigate how behavioural and objective electrophysiological markers of suprathreshold hearing relate to speech-in-noise recognition in young/elderly normal-hearing (yNH/oNH) and elderly listeners with sloping high-frequency audiograms (oHI). We report audiogram, distortion-product otoacoustic emission and brainstem EEG measures aiming to quantify synaptopathy. We also assessed temporal fine-structure and envelope coding performance and compared these metrics to speech reception thresholds. Speech-in-noise material was filtered to only contain frequencies present in temporal fine-structure/envelope tasks or brainstem EEG measures. While low-frequency relationships between speech intelligibility, temporal coding and brainstem measures were complex, high-frequency relationships showed an emerging pattern. An EEG metric predicted high-pass filtered speech intelligibility across listeners, suggesting that oNH and oHI suffer from synaptopathy, in line with post-mortem temporal bone studies. Findings are important for the design of hearing-aid algorithms which can only be effective when understanding how suprathreshold hearing impacts speech encoding in noise.

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SP.13 – Wed 21 Aug, 15:00-17:00

Feature-based audiovisual speech integration of multiple streams

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Speech perception often involves the integration of information from the auditory and visual streams. This is shown in the McGurk effect, in which a visual utterance, e.g., /ipi/, dubbed onto an acoustic utterance, e.g., /iki/, produces a combination percept, e.g., /ipki/. Here, we studied feature-based audiovisual speech integration by decomposing the auditory component into two streams. One consisted of /i_i/ articulated with an intersyllabic silence, and the other contained the release burst extracted from a naturally produced consonant /k/. An auditory continuum was created with nine stepwise temporal alignments of the two streams. Also, two audiovisual continuums were made by pairing the auditory continuum with either visual /ip_i/ or /i_pi/. We found that the auditory continuum elicited /iki/ percepts despite the lack of formant transitions. In contrast, for both audiovisual continuums, the percept was mostly a visually driven response /ipi/ when the burst overlapped either acoustic vowel. Other temporal alignments frequently produced combination responses. Mostly /ikpi/ combinations were obtained when the burst was closer to the initial vowel, and reverse /ipki/ responses when it was closer to the final vowel. This shows how processing various streams of speech information can result in multiple percepts according to the temporal organization of the speech features.

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SP.14 – Thu 22 Aug, 15:20-17:20

Evaluation of a gaze-controlled directional hearing device in an audiovisual virtual environment

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Directional benefit of hearing devices interacts with head motion behavior of the user: for fixed beamformers it can only be achieved if the user is facing the target. Steerable beamformers rely on DOA estimation that can be less accurate with head movements. In multi-talker situations it remains unclear which of the sources belongs to desired sources. Taken together, current beamformers may not improve speech reception in everyday dynamic listening conditions involving movement of listener as well as in standard lab conditions. To overcome this problem, we proposed an algorithm that estimates auditory attention based on the combination of gaze behavior and DOA estimation (Grimm et al., 2018). Its aim is to support communication and natural movement behavior in dynamic listening conditions better than fixed beamformers. Attention is modeled in two steps: First, the gaze-to-object probability is estimated by classification of gaze direction based on a acoustic scene model. Next, the temporal statistics of the gaze-to-object probability is analyzed. Finally, non-attended sources are attenuated by a spatial filter. In this study, extensions of the model are introduced and evaluated with normal hearing subjects. Results show that a similar benefit was achieved as with conventional directional microphones. As a tendency, a positive effect towards natural head movement was observed.

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SP.15 – Wed 21 Aug, 15:00-17:00

Pitch perception of concurrent high-frequency complex tones

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Accurate pitch perception is possible for harmonic complex tones with fundamental frequencies (F0s) in the musical range (e.g., 1.4 kHz) but with all harmonics beyond the putative limits of phase locking. However, it is unknown whether pitch perception in more complex scenarios, such as with concurrent complex tones, is possible using these stimuli. To address this question, we measured (1) F0 difference limens (F0DLs) and (2) target-to-masker ratios (TMRs) required to detect a fixed F0 difference in mixtures of complex tones with low F0s (~280 Hz) or high F0s (~1400 Hz). The target tones were filtered to ensure that in the high-F0 case only harmonics beyond the limits of phase locking were present. Pitch perception was poorer for isolated high-F0 tones than for isolated low-F0 tones, and adding a concurrent masker complex tone (masker F0 centered geometrically between the target and reference F0) impaired performance for both high-F0 and low-F0 tones. The TMRs required to achieve good performance in the presence of two concurrent complex tone maskers (one masker F0 higher than and one masker F0 lower than the target F0) were higher for high-F0 tones than for low-F0 tones. The results should help determine whether different mechanisms underlie the perception of combinations of complex tones at low and high frequencies.

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SP.16 – Thu 22 Aug, 15:20-17:20

Towards unblinding the surgeons: Complex electrical impedance for electrode array insertion guidance in cochlear implantation

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The complications during electrode array insertion in scala tympani for cochlear implantation may cause trauma, residual hearing loss and affect speech outcomes. The inner ear is like a black box for surgeons during insertion process with no real time feedback and radiation based extraoperative imaging. Impedance measurement of electrodes during insertion is a simple yet effective method to assess array position. For this, an impedance meter has been designed which can measure magnitude $|Z|$, phase (θ), real (R) and imaginary (X_c) parts of impedance. A switching circuit can sequentially scan all electrode pairs at regular intervals during insertion. An Evo® straight electrode array is inserted in a transparent 2:1 scaled up 2D cochlea model (11 trials) filled with 0.9% saline using 3-DoF actuation system. Bipolar impedance measurements of 8 pairs (40 samples each) are taken at regular intervals during 25 mm insertion at speed of 0.05mm/sec. A notable increase in $|Z|$ and R is observed in the apex 3 electrode pairs when they first get in to contact with lateral wall. At the same time, phase gets less negative (more resistive impedance) and Xc increases (less capacitance). These results show that impedance can be used for electrode array localization in cochlea and impedance change due to electrode proximity to different material can have application in other electrode implants.

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Poster Session I

SP.17 – Wed 21 Aug, 15:00-17:00

A word elicitation study including the development of scales characterizing aided listening experience

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The purpose of the present study was to identify the terms hearing aid professionals and their patients use in the communication about the aided listening experience and develop scales that would help characterize this experience in the domain of corrective actions that a hearing care professional may apply. The study comprised a word elicitation study based on observations and interviews from 18 examinations, involving 7 audiologists, 2 medical audiologists, and 15 adult patients from the audiological department at Aalborg University Hospital. The words patients and professionals used for describing the aided listening experience were itemized, noted on cards, and analyzed by developing an affinity diagram. The resulting 80 words were then sorted by three hearing professionals in a supervised card sorting session, leaving 65 attributes (grouped in 13 main categories) that was considered suitable for suggesting corrective actions. These 65 attributes were included in a 63-point scale design, which (in a usability test including 8 hearing aid users) were considered easy to survey and use, but also including some redundancy and ambiguities. The results suggest that it is possible to develop scales based on the voluntary statements expressed during actual consultations, but that the expressions may not be interpreted the same way by other patients and professionals.

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SP.18 – Thu 22 Aug, 15:20-17:20

A new road to decode the direction of auditory attention in humans

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In contrast to animals like cats, it seems to be common knowledge that we humans do not re-orient our pinna when focusing our attention on external sounds. However, it has recently been suggested that there is a vestigial pinna–orienting system in humans as well. In two consecutive studies, we investigated whether the direction of auditory attention could be reflected in the electrical response of muscles within that vestigial auriculomotor system. In the first study surface electromyograms (EMGs) were taken from participants' auricular muscles during stimulus-driven, reflexive attention towards novel sounds presented from external speakers at different spatial positions. In the second study we assessed goal-directed, voluntary attention while participants listened to an audiobook coming from one of these speakers, while ignoring a competing audiobook from the opposite one. In both experiments, EMG recordings showed increased activity at the ear on the side of the attended stimulus, but with slightly different patterns. Taken together, our data show that we direct our ears towards sounds to which we pay attention and that auriculomotor EMG could be used to study auditory attention.

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SP.19 – Wed 21 Aug, 15:00-17:00

Applicability and outcomes of a test for binaural phase sensitivity in elderly listeners

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Interaural phase difference (IPD) discrimination in the binaural auditory system has been shown to be related to localization abilities and speech intelligibility in background noise. One of the tests for binaural phase sensitivity determines the highest frequency for which an interaural phase difference of 180° is detectable (named as "IPD-FR"). This test was included in an extensive test battery together with examination of visual and hearing abilities, balance, tactile- and motor-skills, and cognitive abilities. 223 participants aged 55 to 81 years completed the test battery. The IPD-FR test was conducted as an adaptive 3-AFC experiment starting with a frequency of 250 Hz. Participants were instructed orally about the signals to ensure that the task was understood and responded to ten trials minimum prior to the test. Of the 223 participants, 65 participants could not discriminate between the three intervals and 3 participants did not participate in the test due to various reasons (e.g., single sided deafness, technical problems). Statistical analyses of the differences between those who could and could not successfully perform the task and the test results for those who completed the task revealed hearing loss, gender, age, tactile- and motor skills, vision, and cognitive abilities as significant factors.

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SP.20 – Thu 22 Aug, 15:20-17:20

Hearing aid satisfaction and differences in self-reported and data logged hearing aid usage time for experienced and first time users

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Background: Hearing aid (HA) satisfaction is assessed by the self-administered International Outcome Inventory of Hearing Aids (IOI-HA) questionnaire. Objectives: The aims of the current study were to investigate the level of HA satisfaction for experienced and first time HA users, and to evaluate any difference between self-reported and objectively measured HA usage time (through data logging). Design: Self-reported questionnaire survey. Patients enrolled in the national BEAR project, from January 2017 to January 2018, answered the seven-item IOI-HA questionnaire targeting different hearing outcome domains; each scored from 1-5. Data logged HA usage time was obtained at two months follow-up visits and compared to the self-reported usage time obtained from the initial IOI-HA questionnaire item. Results: The study population (n=1649) comprised of both experienced (n=458) and first time HA users (n=1191). Total mean IOI-HA scores for experienced HA users increased by Δ0,36 (SD=0,92). Differences in levels of satisfaction between the two groups were further analyzed. Moreover, data logged usage time for experienced users was 10.4 hours (SD=5,10) and 8.35 hours (SD=1,02) for first time users. 15,9% of experienced users (n=422) reported in average a usage time from 4 to 8 hours whereas 21,5% of first time users (n=1152) reported in average 4 to 8 hours usage time.

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SP.21 – Wed 21 Aug, 15:00-17:00

Audiovisual sound localization in virtual reality

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Virtual reality can be a strong research tool in audiovisual experiments. It allows us to research audiovisual integration in more complex and realistic settings. Here, using a virtual reality setup up in combination with a loudspeaker array, 16 normal-hearing participants were tested on their sound localization abilities. The virtual environment consisted of a 1:1 model of the experimental environment except with the loudspeaker array replaced by a ring. This ring indicated the height, but not the horizontal position of the loudspeakers. The visual component of the stimuli consisted of a ball appearing at a location, falling and then bouncing once on the ring after which it disappeared again. On visual impact with the ring, an impact sound was played from a loudspeaker. Participants were asked to indicate the apparent sound origin, for both congruent and incongruent visual and audio spatial positions ranging from -45 to 45 degrees. The dynamic virtual reality visual stimuli in combination with real auditory stimuli were capable of inducing audiovisual integration, as indicated by a visual bias in the sound localization. The range of this integration extended, in several of the participants, over far larger ranges of audiovisual disparity compared to earlier studies on audiovisual integration.

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SP.22 – Thu 22 Aug, 15:20-17:20

Duration threshold for identifying speech samples for different phonemes

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In daily life, the identification or classification of sounds enables us to build an acoustic scenery with certain objects in our brain. Moreover, it allows us to decide in what way a sound needs to be interpreted, e.g. the sound of a car has other features than a speech signal. In recent studies, it has been shown that the minimal duration of a sound, which is required for a correct identification, could be a useful audiological parameter, e.g. providing information about the hearing ability of a person. These studies report a duration threshold for speech in the range of 20-40 ms, which is remarkable short compared to technical algorithms e.g. used in modern hearing aids. In this work, we want to investigate what cues are used by humans to classify a sound correctly as speech. To this end, the duration thresholds for identifying speech samples starting with different phonemes are analyzed for normal hearing and hearing impaired listeners. On the one hand, the analysis of phonemes, which can be easier detected as speech, helps to understand what cues humans usually exploit. On the other hand, since hearing impaired people are also involved, it should be investigated whether or not a hearing impairment affects the duration thresholds for all phonemes in the same way. This could provide information about the impact of hearing impairment on the ability to classify short sounds.

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SP.23 – Wed 21 Aug, 15:00-17:00

Evaluation of a mobile phone implementation of a notched noise masking paradigm

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The ability to conduct hearing tests and estimate auditory function at home or in the workplace can be useful for screening or longitudinal studies and allow the collection of diagnostic data from tests that are too time consuming to be feasible in the clinic. However, moving away from an acoustically controlled environment may influence the results of a test (e.g., through masking due to higher levels of background noise), increasing the uncertainty in the test measurements. In this study, 9 normal-hearing participants completed a notched noise masking experiment with three different experimental setups: in a psychoacoustic test booth with a standard laboratory PC; in a psychoacoustic test booth with a mobile device; and in a quiet office room with a mobile device. The accuracy and reliability of the mobile implementation was compared to results obtained with the laboratory setup. The effect of the test environment was investigated by comparing the mobile platform results between booth and office. The mobile device implementation corresponded well with the laboratory results for a notch width of zero but showed a systematic bias when the width of the notch was increased. The reliability of the mobile implementation was comparable to the laboratory. Moving outside the sound-insulated booth did not affect the mobile platform results.

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SP.24 – Thu 22 Aug, 15:20-17:20

Latent representation linear speaker recognition using deep transfer learning

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Transfer learning is the application of an already trained neural network for a new task, and it enables utilization of insights gathered from very large datasets in the analysis of much smaller datasets. We investigate transferring learnt latent representations of audio from two very different state-of-the-art audio-processing neural networks by analysing their embeddings of the TIMIT speech corpus. The networks used are the VGGish (trained on classifying the AudioSet, over 2 million human-labelled 10-second YouTube videos), as well as the NSynth WaveNet (a generative model trained on 300k musical notes from 1k instruments). We demonstrate the utility of the transferred latent representations by showing how they can form the basis for a speaker identification system trained on very few utterances from each speaker. Using a simple linear classifier (regularized multinomial regression) on the sentence-averaged embeddings, we compare these data-driven embedding systems to a domain-knowledge driven baseline system (based on sentence-averaged log Mel-filterbank energy features). The study shows how transfer learning reduces the otherwise ubiquitous need for prohibitively large data sets in deep learning, and it highlights how it is a promising avenue for compounding existing approaches with deep learning approaches - even for problems characterized by small datasets.

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Poster Session I

SP.25 – Wed 21 Aug, 15:00-17:00

Neural health in cochlear implant users with ipsilateral residual hearing

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Progressive auditory nerve degeneration is known in patients suffering from severe cochlear hair cell loss and is partially assumed to be responsible for the variability in speech reception abilities among cochlear implant (CI) users. Measurements using electrically evoked compound action potentials (ECAP) recorded with varying inter-phase gaps (IPG) have been shown to correlate with duration of deafness. In the present study we investigated the ECAP characteristics in CI users with residual hearing in the low frequencies. We assumed better neural health in apical regions of the cochlea in these subjects, allowing a validation of the measurement. So far 17 MED-EL Flex users participated in ECAP recordings, electric-acoustic masking and speech reception thresholds measurements. Amplitude growth functions (AGFs) were recorded for IPGs of 2.1 and 10 µs and the slope determined. N1 latency and AGF slope were compared. IPG showed an increasing effect on ECAP amplitude in seven subjects. IPG showed an inconsistent effect on N1 latency. The difference (Δ) in slope and latency did not depend on the electrode position. Currently CI users without residual hearing are being measured to compare the pattern. The duration of deafness could be a confounding factor in the pattern, but no correlation was observed. Δ N1 latency and Δ slope did not correlate with speech reception thresholds.

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SP.26 – Thu 22 Aug, 15:20-17:20

Physiological correlates of masking release in conditions of streaming

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Our auditory system enables communication in noisy environments by making use of specific signal properties (cues). Comodulation and interaural phase differences (IPD) were found to be beneficial cues to segregate a tone from noise, leading to a masking release. In addition, streaming has been shown to affect the masking release induced by comodulation. There is, however, a lack of understanding of how these cues are involved in signal processing along the auditory pathway. In this study, we investigated an "internal signal-to-noise ratio" of a sound at the level of the auditory cortex with electroencephalography (EEG). We conducted psychoacoustic and electrophysiological experiments in parallel to link behavioral measures and neuronal representations. We hypothesized that the temporal contexts will affect the effects of two cues on the masking release. For each task, a tone was presented at supra-threshold levels with, or without IPD, and four different maskers were applied: uncorrelated, comodulated and two different temporal contexts as used in previous streaming experiments. Psychoacoustical data were in line with our hypothesis and its correlation with electrophysiological data will provide us information about whether cortical auditory evoked potentials can be a biomarker of individual audibility of masked tones in complex acoustic conditions.

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SP.27 – Wed 21 Aug, 15:00-17:00

Auditory adaptation in real and virtual rooms

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Walking from room to room in real listening conditions is a natural process in our everyday life, and there is no obvious challenge for our auditory system to cope with. However, in experiments with virtual acoustic environments switching the virtual room or switching from real to virtual rooms can result in severe auditory confusions which can lead to in-head localization. This effect is known as room divergence effect. A series of listening tests were conducted to verify this effect under different conditions as well as experiments which studied the effect of prior sound exposure and the time variant behavior of it. In this paper two of these experiments are described and discussed. The first experiment shows, that the extent of the room divergence effect depends on the room acoustics we have just learned. That indicates, that the room divergence effect is diminished during ongoing exposure to a specific room acoustic condition. The second listening test shows, that it is necessary to suppress this time-variant effects in order to study other quality elements such as the effect of specific system components (in this case head-tracking) on the perception of externalization. These tests raise the question why switching virtual rooms leads to temporary confusions but doing so with real rooms is unproblematic. Different theories are discussed in this publication.

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SP.28 – Thu 22 Aug, 15:20-17:20

Development of a Danish sentence material for assessing speech-in-noise reception in school-age children

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For the audiological assessment of the speech-in-noise abilities of children with normal or impaired hearing, an appropriate test material is required. The purpose of the current study was to develop a Danish sentence material suitable for school-age children. Based on the 600 test sentences from the Danish DAT corpus (Nielsen, Dau & Neher, JASA 2014), 11 test lists comprising 20 sentences each were carefully constructed. These lists were evaluated in terms of their perceptual similarity and reliability with a group of 20 typically developing, normal-hearing children aged 6-12 yrs. Using stationary speech-shaped noise and diotic stimulus presentation, speech recognition thresholds (SRTs) were measured twice per list and participant at two separate visits. The analyses showed an overall SRT of -2.0 dB SNR, an average test-retest improvement of 0.5 dB, and a within-subject standard deviation of 1.1 dB. Furthermore, eight test lists produced mean SRTs that were within 1 dB of each other. Altogether, it is concluded that the developed sentence material is suited for assessing speech-in-noise recognition of Danish school-age children.

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SP.29 – Wed 21 Aug, 15:00-17:00

Characterizing the speech-in-noise hearing abilities of school-age children with a history of middle-ear diseases

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Recently, a number of studies have indicated that recurrent or chronic middle-ear disease during early childhood may lead to long-term supra-threshold hearing deficits. The current study follows up by investigating differences in monaural and binaural hearing abilities in noise among school-age children with or without a history of middle-ear diseases. Participants were children aged 6-12 years with a history of recurrent otitis media with infection or effusion and without any previous ear diseases. All children had normal middle-ear function and normal audiometric hearing thresholds at the time of testing. Measurements included monaural and binaural speech recognition thresholds in the presence of stationary noise or competing speech. Sensitivity to monaural and binaural phase information in the presence of background noise was also assessed. Preliminary analyses based on the data from the first 20 participants indicate group differences in terms of speech recognition with competing speech and binaural phase sensitivity. Follow-up analyses based on a larger dataset will shed further light on how recurrent early-childhood middle-ear disease affects hearing abilities in noise and how any resulting deficits can be identified in clinical practice.

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SP.30 – Thu 22 Aug, 15:20-17:20

Pupillary response to auditory memory recall task in normal hearing young adults

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Although noise reduction in hearing aid listeners has a significant impact on better recall performance (Lunner, Rudner, Rosenbom, Agren, & Ng, 2016), comparatively little is known about the association between cognitive demands of a free recall task and listening effort in response to auditory information. The current pupillometric study examined it during the encoding interval masked by 4-talker babble using a revised free recall paradigm consisting of 14 seven-sentence lists extracted from the Korean Hearing in Noise Test (KHINT) (Moon et al., 2005). In addition to the KHINT and reading span task, ten normal hearing adults were instructed to recall the sentence-first words after each set of seven sentences. Raw pupil data was processed in the guided steps (Kret & Sjak-Shie, 2018) and 25% of blink inclusion criterion. A Linear Mixed Models ANOVA revealed that the baseline associated with working memory load steadily and significantly increased ($p <.05$), in parallel with the increasing number of words to be remembered. A significantly marginal group difference between high and low RS appeared ($p= .052$). Conversely, the PPDs or listening effort significantly decreased as opposite to the baseline pattern ($p <.01$). The latency of PPD for the middle or asymptote items implying retrieval difficulty was the longest and significantly different from the first items ($p <.01$).

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SP.31 – Wed 21 Aug, 15:00-17:00

The influence of phoneme-based auditory training on speech intelligibility and listening effort in hearing-aid users

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Despite enormous progress in hearing-aid technology over the past years, many users still experience difficulties in understanding speech and report high listening-related effort, especially in noisy environments and large-group conversations. Auditory training can serve as a complementary strategy that can foster the development of perceptual and cognitive skills via properly designed task and thus help alleviate the speech-comprehension problems that hearing-impaired face in everyday life. In the presented study, a training program based on stimuli from Danish Nonsense Word Corpus (DANOK) was designed. The participants with similar hearing profiles and at least six months of experience with using hearing aids were recruited to undergo a two-week training program or to be a part of an active control group involved in a non-relevant task throughout the same period. Then the effects of training on speech intelligibility, listening effort, self-perceived hearing abilities, and cognitive functions were assessed. The findings of the study will help to evaluate the efficacy of the proposed training paradigm as a potential strategy to improve speech intelligibility in Danish-speaking hearing-aid users.

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SP.32 – Thu 22 Aug, 15:20-17:20

"Psychophysical" modulation transfer functions in a deep neural network trained for natural sound recognition

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Representation of amplitude modulation (AM) has been characterized by neurophysiological and psychophysical modulation transfer functions (MTFs). Our recent computational study has demonstrated that a deep neural network (DNN) trained for natural sound recognition can be a good model for explaining the functional significance of neuronal MTFs derived physiologically. The present study asks whether the DNN provides insights into AM-related human behaviours such as AM detectability. Specifically, we measured "psychophysical" MTFs in the DNN model we have developed previously. We presented to the DNN sinusoidally amplitude-modulated white noise with various AM rates, and quantified AM detectability as d' derived from the model's internal representation of modulated and non-modulated stimuli. The overall d' increased along the layer cascade, with human-level detectability observed in the higher layers. In a given layer, the d' tended to decrease with increasing AM rates and with decreasing AM depth, which is reminiscent of a psychophysical MTF. The results suggest that a DNN trained for natural sound recognition can be a model for understanding psychophysical AM detectability. Since our approach is not specific to amplitude modulation, the present paradigm opens the possibility of exploring a broad range of auditory functions that can be evaluated by psychophysical experiments.

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SP.33 – Wed 21 Aug, 15:00-17:00

Effect of binaural loudness summation on listening effort

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Listeners with hearing loss often report of increased listening effort in acoustically challenging situations. The analysis of subjective listening effort ratings showed that listeners with comparable hearing thresholds perceive different degrees of listening effort. A possible explanation could be differences in the binaural broadband loudness summation. Oetting et al. (2016, 2018) revealed large individual differences of binaural broadband loudness summation in listeners with similar hearing thresholds. The aim of this study was to investigate the relation between the perceived listening effort and the binaural broadband loudness summation. 20 listeners with hearing loss participated. Perceived listening effort was measured unaided with the subjective scaling method ACALES (Krueger et al., 2017) using two different background noise levels (50 and 65 dB SPL). All listeners conducted the monaural and binaural categorical loudness scaling with narrow- and broadband stimuli to assess the individual binaural broadband loudness summation. Pairs of listeners with matched thresholds were analyzed and showed a correlation between binaural broadband summation and listening effort for 50 dB SPL. No relation was found for 65 dB SPL. The results and conclusions of these findings will be discussed.

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SP.34 – Thu 22 Aug, 15:20-17:20

Studying how reverberation affects speech intelligibility in noise with multichannel auralizations

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Test stimuli in many speech-in-noise-tests contain little to no reverberation whereas people often report on difficulties specifically in understanding speech in reverberant conditions. The aim of this project is to investigate how advanced spatial sound technologies can be used in hearing diagnostics and to develop novel tools with more complex and authentic sound scenes. A recently developed auralization method (Spatial Decomposition Method) may be well suited for bringing advanced spatial sound technology into clinically feasible environments. Here, we present results of our study, where we have used the Finnish Matrix Sentence Test with multichannel auralizations of two real spaces with approx. one and two second reverberation times to measure speech reception thresholds in noise between anechoic/dry and reverberant conditions. Perhaps surprisingly, the results indicate a small but significant improvement in SRTs with one second reverberation time compared to the reference case without reverberation.

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SP.35 – Wed 21 Aug, 15:00-17:00

Using neural health correlates to predict the utilization of electric simultaneous stimulation

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The individualization of Cochlear Implant (CI) sound processing is a vital factor to improve the performance of the user and the power consumption of the device. Neural health correlates such as focused thresholds and electrically evoked compound action potentials (eCAP) can be used as such individualization factors to characterize a CI user. The study's goal is to investigate if CI users that gain a benefit from simultaneous stimulation with respect to sequential can be characterized with neural health correlates. Speech intelligibility was measured at individual speech-to-noise ratios with three different sound coding strategies (sequential, simultaneous with two and three virtual channels using current steering). Focused thresholds were measured with a sweep procedure and quadrupolar stimulation across the electrode array, eCAPs were measured with a forward masking scheme for two different interphase intervals at individual amplitudes to result in amplitude growth functions. A computational model based on a simulation Framework for Auditory Discrimination Experiments is used to predict speech intelligibility from individualized data sets generated from the results (neural health data and sound coding strategy settings). The results will be presented at the conference, answering the question if neural health correlates can predict the utilization of simultaneous stimulation.

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SP.36 – Thu 22 Aug, 15:20-17:20

Hearing aid feature profiles and the success of rehabilitation

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The selection of a hearing aid (HA) that covers the rehabilitation needs of a patient is one of the key elements of a successful rehabilitation. Yet, it is complex to compare different models, whereas each manufacturer prefers to describe comparable features in its own terminology. This complicates the match between personal and diagnostically determined rehabilitation needs and the relevant characteristics of HA's. We recently developed a method to objectively classify HA's based on audiologically relevant features (e.g. compression, noise reduction, etc.) using technical information from over 3900 different HAs. This yielded HA populations characterized as distinct feature profiles. Our present research aims to combine audiological quantified rehabilitation needs with well-defined HA characteristics, independent of manufacturer or type. Our dataset includes audiological diagnostic tests, personal rehabilitation needs with post evaluation scores (COSI), and other pre-/post-rehabilitation PROMs. We investigated: which HA features/feature profiles contributed (positively) to a successful rehabilitation, and to which extent these feature profiles were related to specific rehabilitation needs; the impact of age and HA experience on these relations; and if the selection of a HA and the degree of compensation power of the HAs has any relation to diagnostic hearing tests.

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SP.37 – Wed 21 Aug, 15:00-17:00

Benefit from different beamforming schemes in bilateral hearing aid users: Do binaural hearing abilities matter?

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Neher et al (Hear Res 2017) recently showed that binaural hearing abilities influence speech-in-noise reception with different bilateral directional processing schemes using a hearing aid simulator and virtual acoustics. The current study aimed to extend this finding to real acoustic environments and ear-level devices. Three beamforming schemes that differed in signal-to-noise ratio (SNR) improvement and binaural cue preservation were tested. The participants were 38 older experienced hearing aid users. Speech understanding and localisation performance were measured. Binaural hearing abilities were assessed using the binaural intelligibility level difference (BILD). The analyses revealed a clear effect of the BILD on speech understanding in noise, but no interaction with the beamformer conditions. Greater SNR improvement was generally beneficial. In contrast, localization of static and dynamic stimuli was more accurate when low-frequency binaural cues were preserved. Furthermore, the interaction with the BILD was marginally significant for dynamic stimuli ($p = 0.054$). Altogether, these results suggest that when selecting directional processing schemes in bilateral hearing aid fittings both speech understanding and aspects of spatial awareness perception should be considered.

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SP.38 – Thu 22 Aug, 15:20-17:20

How to compare hearing-aid processing of real speech and a speech-modified stimulus for measuring the aided Auditory Steady-State Response?

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For validating hearing-aid fittings in prelingual infants, an objective assessment based on the Auditory Steady-State Response (ASSR) is considered. To ensure that the hearing aid(s) under test supply speech-relevant gain and signal-processing features during the measurement, a speech-modified ASSR stimulus was devised from three bandlimited CE-Chirps® modified by frequency-band specific envelopes derived from the International Speech Test Signal (ISTS) and scaled in level to match the ISTS. Prior to testing, it needs to be verified that this stimulus in fact drives the hearing aid into speech mode. This work proposes a 'black-box' measurement method which compares the gain applied to the actual ISTS and the ISTS-modified 3B CE-Chirp in 1-octave bands. Because the detailed waveforms of the two signals are markedly different, a direct comparison is difficult. Instead, short probe snippets of steady-state noise spectrally shaped to the ISTS are inserted into the two signals. Thus, the ISTS and the ISTS-modified 3B CE-Chirp are used as conditioning signals, whereas the actual gain comparison is based on the probe snippets. Since the method does not require any specialist hearing-aid brand knowledge it should be broadly useful also in clinics. Example results will be presented to open a discussion about the viability of this approach across a wide range of pediatric hearing aids.

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SP.39 – Wed 21 Aug, 15:00-17:00

Analysis of a forward masking paradigm proposed to estimate cochlear compression using an auditory nerve model and signal detection theory

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The healthy human auditory system is sensitive to a large dynamic range of incoming sounds. An "active mechanism", presumably due to the electromotility of the outer hair cells (OHC) in the cochlea, leads to a level-dependent amplification of basilar membrane (BM) vibration and a compressive BM input/output (I/O) function. Different methods to estimate this compressive function based on a psychoacoustical forward making paradigm have been suggested. These methods assume that cochlear processing can be isolated from the response of the overall system and that sensitivity is dominated by the tonotopic place of the probe. In the present study, a computational model of the auditory nerve (AN) in combination with methods from signal detection theory (SDT) was used to test these assumptions. The simulated AN response was quantified in terms of rate and synchrony for different types of nerve fibers. The analysis allowed to localize the tonotopic contributions to the overall sensitivity to the stimuli. Results show that the AN model can account for the behavioral data and that local activity at the probe frequency is the dominant contributor to the sensitivity when simulating a normal hearing cochlea. The results also show in addition, that the spontaneous activity of the different nerve fibers is an important factor to consider when evaluating sensitivity.

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SP.40 – Thu 22 Aug, 15:20-17:20

The Danish version of the Speech, Spatial and Qualities of hearing scale 12, the SSQ12 – A study of validation and correlation

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Objective: To test the validity and reliability of the Danish version of the Speech, Spatial and Qualities in Hearing Scale 12 (DK-SSQ12). To this day, a test-retest reliability study of any translation of the SSQ12 has not been conducted. To study the correlation of quality of hearing and Quality of Life (QoL) we wish to investigate the correlation between the DK-SSQ12 and the hearing domain of the Quality of Life questionnaire, 15D, domain 3. Study sample: 1961 participants in the Better hEAring Rehabilitation project was included in population group A. Forty-one subjects without hearing aids and 52 subjects with hearing aids from group A were recruited to population group B. Design: Population group A tested the internal validity of the DK-SSQ12, using Cronbach's alpha (CA). Population group B tested the reliability of the DK-SSQ12 assessing the Inter Correlation Coefficient (ICC). The DK-SSQ12 was distributed twice with a two-week interwall. Spearman's rho was applied to test the correlation of DK-SSQ12 and 15D, 3. Results: The CA, divided into three domains of the DK-SSQ12, ranged from 0.89-0.91 in the Speech domain, 0.81-0.91 in the Spatial domain and 0.75-0.81 in the Qualities domain. Group B showed an ICC of 0.66-0.89 (95% CI 0.44-0.94). All domains of the SSQ12 is significantly correlated to the 15D question 3.

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SP.41 – Wed 21 Aug, 15:00-17:00

"Yes, I have experienced that!" - How daily life experiences may be harvested from new hearing aid users

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The aim of the present pilot study was to assess daily life experiences of new hearing aid users and explore ways to utilize these assessments in a follow-up situation with the hearing care provider. The method was designed as an online pass-time activity, where the patients swipe through the randomly presented experiences, and select the ones they have had recently. The sentences were expected to evoke the memory of recent experiences and provide a language for the patients to describe these. Thirty new hearing aid users were included in the study. Data were collected over a period of two months and consisted of 453 pre-fabricated sentences representing experiences related to HA use. Each sentence correlated with one of 13 categories covering both auditory and non-auditory aspects. Data for each patient was visualized to elucidate both short- and long-term challenges and successes experienced, as well as irrelevant and not experienced situations. Presently the first three patients included in the study have completed a two-month follow-up. The overall response rate is 63% taking into account that some patients may not have started the log activity yet as the work is ongoing.

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SP.42 – Thu 22 Aug, 15:20-17:20

A method for evaluating audio-visual scene analysis in multi-talker environments

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In cocktail-party environments, normal-hearing listeners are able to comprehend and localize multiple simultaneous talkers. Previous investigations mainly focused on audio-only scenarios, not considering the effect of visual information. With current virtual reality hardware, it has become possible to present visual content via a head-mounted display in a controlled laboratory setting. By utilizing such a head-mounted display and a spherical loudspeaker array, it is possible to reproduce realistic audio-visual multi-talker scenarios. However, current sentence-based speech corpora most often used for speech intelligibility testing are not well-suited for presentation in realistic virtual environments. Therefore, a new continuous speech corpus was designed, recorded and applied in this work. The corpus contains ten monologues from five female and five male talkers. Using a proposed egocentric interaction method, listeners can label perceived talkers according to source position and content of speech. With an increasing amount of simultaneous acoustic talkers, the listeners' accuracy in performing this task decreases. With this proposed method, a new approach is explored to gauge listeners' ability to analyze complex audio-visual scenes.

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SP.43 – Wed 21 Aug, 15:00-17:00

Investigating the effect of extended-frequency amplification as tinnitus treatment

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More than 80% of people suffering from chronic tinnitus also have hearing loss, and earlier studies have found that the tinnitus pitch is often located within the range of the hearing loss. These findings led to the hypothesis that it may be possible to decrease tinnitus loudness by increasing the auditory nerve activity via amplification provided by hearing aids. In this longitudinal crossover study, the effect of amplification is investigated in a randomized double-blinded trial. 20 participants suffering from tinnitus with clinically normal hearing (≤ 25 dB) in the frequency range from 125 Hz to 4 kHz and hearing loss in the high frequencies are included in the study. To assess the possible effect of amplification, tinnitus questionnaires, visual analogue scales for loudness and annoyance, and psychoacoustic measures of the tinnitus likeness and loudness are measured before and after treatment. The study consists of an experimental condition where participants receive amplification in an extended range of frequencies (125 Hz to 10 kHz) and a control condition where participants receive amplification in a normal frequency range (125 Hz to 4 kHz). We hypothesize that the experimental condition will provide a larger decrease in the tinnitus perception compared to the control condition. Preliminary results at the three-month crossover time point will be presented and discussed.

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SP.44 – Thu 22 Aug, 15:20-17:20

The effect of conversational task on turn taking and timing in dialogue

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In previous studies, several methods have been used to elicit conversation between talkers. Some involved participants solving a shared task (e.g., describing a map or finding differences between two near-identical pictures), while others have recorded more spontaneous dialogue (e.g., telephone calls). Since the goals of the talkers, and thus the definition of successful conversation, varies across these methods, it is thought likely that turn taking behaviour will vary depending on how conversations are elicited. The present study investigated this by eliciting English conversations from 7 pairs of native-Danish talkers using two methods: solving a Diapix task and engaging in unguided "small talk". For each method, in both quiet and 70 dBA babble, two conversations were recorded for each pair. Overall, several differences in conversational behaviour were observed. When engaged in "small talk", participants spoke more rapidly, produced longer turns, and replied more quickly than compared to when they were solving the Diapix task. These within-pair differences indicate that comparisons of behaviour across studies should also consider the method by which conversations were elicited.

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SP.45 – Wed 21 Aug, 15:00-17:00

Timing of turn taking between normal-hearing and hearing-impaired interlocutors

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Having a conversation requires more resources than just understanding speech. Previous studies of the timing of turn taking in conversations suggest that in order to sustain normal, fluid turn-taking, interlocutors have to predict the end of each other's turns. Thus, while noise and hearing loss should make understanding speech more difficult, it should also reduce the resources available for speech planning and possibly reduce the saliency of cues used to predict turn ends, resulting in delayed and more variable turn taking. We recorded conversations between 12 pairs of native-Danish young normal-hearing (NH) and older hearing-impaired (HI) listeners with mild presbyacusis in quiet and multitalker babble at three levels. The interlocutors conducted a Diapix task, finding differences in two near-identical pictures. The time it took the pairs to complete the task increased with increasing noise level, suggesting that communication was impaired by the noise. Both HI and NH talkers responded more slowly and with more variability with increasing noise level, and the effects were larger for HI. In addition, talkers held their turns significantly longer in increasing noise levels, which allows more time for speech planning and understanding. While the HI talkers kept their speech rates constant, the NH decreased their speech rates with increasing noise level.

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SP.46 – Thu 22 Aug, 15:20-17:20

The effects of noise and second language on the timing of turn taking in conversation

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Previous studies of floor transfer offsets (FTO), the interval between one talker stopping and the other starting, suggest that normal conversation requires interlocutors to predict when each other will finish their turn. We hypothesized that increasing the difficulty of holding a conversation by adding noise and/or speaking in a second language (L2) would result in longer FTOs. Conversations from 20 pairs of normal hearing, native-Danish talkers were elicited using the Diapix task in four conditions consisting of combinations of language (Danish vs. English) and noise background (quiet vs. ICRA 7 noise presented at 70 dBA). Overall, participants took longer to complete the task in both noise and in L2 indicating that both factors reduced communication efficiency. However, L2 had very little effect beyond completion time, likely because the participants were very good in English. In contrast to our predictions, in the presence of noise, the median of the FTO distribution decreased by approximately 30ms and the standard deviation decreased by approximately 10%. However, the average duration of interpausal units (i.e., utterances of continuous speech) increased by 40% in noise. These findings are consistent with talkers holding their turn for longer, allowing more time for speech planning.

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SP.47 – Wed 21 Aug, 15:00-17:00

The effect of harmonic number and pitch salience on the ability to understand speech-on-speech based on differences in fundamental frequency

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Differences in fundamental frequency (F0; an attribute of sound that elicit a sensation of pitch) between competing voices facilitates the ability to understand speech. Although lower-numbered harmonics produce greater pitch salience than higher-numbered harmonics, it remains unclear whether differences in harmonic ranks present, and therefore pitch salience, will affect such benefit of pitch differences. A study by Oxenham and Simonson (2009) tested conditions with either only high or low harmonic ranks present and did not find an effect of harmonic rank. However, their study only tested conditions where the difference in long-term average F0 (ΔF0) between the two competing voices was fairly large (4 and 8 semitones, ST) and it is possible that the effect of pitch salience would be greater in more challenging conditions, i.e. in conditions with a smaller ΔF0. This study tested speech intelligibility in conditions with one speech masker for ΔF0s of 0,2, and 4 ST. The speech was presented in a broadband condition or was high or lowpass filtered. Preliminary results show similar performance in the high- and low-pass filtered conditions for all ΔF0s, suggesting little or no effect of harmonic rank in the ability to use F0 to segregate voices, even with smaller ΔF0s between competing voices.

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SP.48 – Thu 22 Aug, 15:20-17:20

Age-dependent changes in frequency-following responses as a potential marker of cochlear synaptopathy in humans

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Cochlear synaptopathy occurs as a consequence of noise exposure and aging but diagnostic measures in humans are missing. With synaptopathy, a reduction of the number of auditory nerve fibers may degrade the processing of fine temporal cues relying on synchronous activity of many nerve fibers. The frequency following response (FFR) is considered to reflect synchronous neural activity phase locked to the temporal fine structure of the stimulus. At higher stimulus levels, due to the spread of neural excitation across frequency, the FFR represents a fairly broadband response dominated by the synchronized neural activity stemming from more basal ("off-frequency") nerve fibers. A degraded neuronal synchrony due to loss of nerve fibers may lead to a reduced FFR, even for stimulus frequencies where no sensitivity loss is found. Here, we investigated age-related changes in FFRs to tones at 703 Hz and 319 Hz, presented at a sound pressure level of 85 dB nHL. Furthermore, FFRs to frequency sweeps from 0.2 to 1.2 kHz were measured to explore the upper limit of the FFR existence region, both in quiet and in 5dB SNR. Additional potential measures of synaptopathy such as the middle-ear muscle reflex, high-frequency audiometry and early components of auditory brainstem responses were evaluated. The results demonstrate a reduction in the upper limit of the FFR response in aging listeners.

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SP.49 – Wed 21 Aug, 15:00-17:00

The role of fundamental frequency in competing-talker scenarios

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In competing-talker scenarios it is essential to perceptually segregate the target speech from the interfering speech. Even when only monaural cues are available, normal-hearing (NH) listeners exhibit exceptional abilities in identifying and understanding the target while hearing-impaired listeners often experience difficulties. The investigation of monaural cues is therefore essential for developing hearing-aid compensation strategies. Previous studies with NH listeners showed that differences in fundamental frequency (F0) between target talker and one interfering talker can help segregate the speech signals. However, most of these studies used speech materials that are far from everyday speech. Furthermore, the F0 was either defined by talker sex or measured as a talker-specific average, thus ignoring the significant F0 variability across sentences. The present study used everyday-speech type sentences and employed a more accurate method for assessing the impact of F0 on intelligibility. Pairs of sentences from the Danish Hearing in Noise Test (HINT) spoken by the same talker were processed to obtain a desired F0-contour difference and presented to NH listeners in a target-masker paradigm with keyword cueing. The results will be analyzed and compared to previous studies in order to contribute to a detailed characterization of the role of F0 in competing-talker scenarios.

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SP.50 – Thu 22 Aug, 15:20-17:20

Speech related hearing aid benefit index derived from standardized self-reported questionnaire data

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Speech understanding in noisy environments has been the most desired hearing aid (HA) benefit sought by hearing aid users. This paper examines the possibility of developing a speech related HA benefit index from correlated speech related questions from three different self-reported questionnaire's (SSQ12, IOI-HA, and 15D). The 4 questions in SSQ12 (question number 1,4,11 and 12), 3 questions in IOI-HA(question number 3,5 and 6) and 3rd question from health-related quality of life questionnaire 15D relating to speech were found correlated and are chosen for further analysis. After the normalization of the relevant questions, a principal component analysis (PCA) is used to reduce the dimensionality and determine the coefficients. The resultant coefficients are used to create a common speech related HA benefit index.

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SP.51 – Wed 21 Aug, 15:00-17:00

Exploring ramped pulse shapes for cochlear implants in an animal model

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A limiting factor in cochlear implants (CI) is the broad excitation patterns caused by the current spread. In current CI devices, the electrical pulse has a biphasic rectangular (Rec) shape. However, we hypothesize that a pulse shape with a ramped slope might better match the biophysics of spiral ganglion cells. In this study, we explore if those pulses are more efficient than Rec pulses in vivo. Deafened mice were implanted with a 4-ch array and eABR were recorded in response to Rec and three ramped pulses: rampUP (increasing slope), rampDOWN (decreasing slope), or rampLONG (increasing slope over both phases). rampUP and rampLONG had a significantly lower threshold of eABR wave II compared to Rec in terms of charge injected (n = 10), and all ramped pulses had a significant steeper wave II growth function compared to Rec. Interestingly, rampUP had a significantly lower threshold and steeper growth function than rampDOWN. eCAP was recorded in response to Rec and rampUP. Pilot data showed a lower threshold with rampUP (n = 3). Future eCAP experiments will look into the spread of excitation and temporal jitter with non-Rec. In summary, the study shows the first evidence of neural responses to ramped CI pulse shapes in vivo. So far, the data support the hypothesis that those pulses require less charge to trigger a neural response than Rec, implying lower power consumption.

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SP.52 – Thu 22 Aug, 15:20-17:20

Reliability of pupillometry method on different features of the pupil

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Pupillometry as a tool indicating listening effort has been extensively analyzed on a group level lately, but less is known about how reliable the pupil dilation is as an indicator of an individual person's listening effort. The aim of this study was to investigate the reliability of the pupil dilation measured during a speech-in-noise task as an indicator of listening effort. The pupil dilation of 27 normal- hearing and 24 hearing- impaired participants were recorded while performed a speech in noise test at two different days. Since there can be systematic variabilities among participants, measures of intraclass correlation coefficient (ICC) consistency were considered in the analysis. The ICC was applied to the individual peak pupil dilation and the mean pupil dilation. Moreover, ICC was tested on the different terms given by the third-order orthogonal polynomial within Growth Curve Analysis (intercept, 1st term, 2nd term, 3rd term), which are assumed to provide further information about temporal changes of the pupil dilation. High values of consistency were found on some measures of the pupil response. Additionally, a BlandAltman analysis was applied as a graphical representation of the reliability of the pupillometry method. The results show different levels of reliability for the different features of the pupil response.

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Poster Session I

SP.53 – Wed 21 Aug, 15:00-17:00

Correlates of electrical field interactions with electrical auditory brainstem responses in pediatric cochlear implantees

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Although electrical auditory brainstem responses (eABR) via the cochlear implant (CI) have been used to assess the neural response to CI stimulation, their prognostic value is still unclear. Most CIs use biphasic current pulses. However, biphasic pulses (BP) can generate significant artifacts that disrupt brainstem responses, which leads to inappropriate information regarding spiral ganglion nerve (SGN) integrity. We hypothesized that wider biphasic current pulses between electrodes may capture a larger SGN population. In this study, we compared interactive- and classic BP eABR findings of pediatric cochlear implantees as a point of reference to the latency and presence of wave V. Furthermore, we conducted 3D cochlear modeling complemented by a multimodal imaging-based detailed anatomical model of the human head and neck to identify differential electrical field interactions depending on biphasic pulse mode. Implants from Cochlear (Lane Cove, Australia; CI522 or CI532) were used in 19 pediatric implantees. All recordings were made during surgery directly after implant insertion. Two separate systems were used, one for recording (Eclipse, Interacoutics, Denmark) and one for stimulation (Cochlear company stimulation programming). The latency and presence of wave V from pediatric cochlear implantees were analyzed. Shorter latency or higher amplitude of wave V was found with interactive BP compared to classic BP, especially in the low-frequency region. Also, there was a significantly higher occurrence of apparent wave V with interactive BP. From 3D cochlear modeling, interactive BP results in more intensive electromagnetic energy and wider activation compared with classic BP. Thus, eABR using interactive BP stimulation leads to a more robust response. Our results provide novel perspectives for the understanding of electrical interaction in the cochlea and newly suggested eABR strategy in the context of the determination of treatment strategy, especially in subjects with cochlear nerve deficiency.

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Poster Session II

SP.54 – Thu 22 Aug, 15:20-17:20

Potential of self-conducted speech audiometry with smart speakers

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Speech audiometry in noise based on matrix sentence tests (Kollmeier et al, 2015) is an important diagnostic tool to assess the speech reception threshold (SRT) of a subject, i.e., the signal-to-noise ratio corresponding to 50% intelligibility. Although the matrix test format allows for self-conducted measurements by applying a visual, closed response format, these tests are mostly performed in open response format with an experimenter entering the correct/incorrect responses (expert-conducted). Using ASR enables self-conducted measurements without the need of visual presentation of the response alternatives (Ooster et al, 2018). A combination of these self-conducted measurement procedures with signal presentation via smart speakers could be used to assess individual speech intelligibility in an individual listening environment. Therefore, this paper compares self-conducted SRT measurements using smart speakers with expert-conducted lab measurements. With smart speakers, the experimenter has no control over the absolute presentation level, mode of presentation (headphones vs. loudspeaker), potential errors from the automated response logging, and room acoustics. We present the differences between lab and Alexa settings for normal-hearing and hearing-impaired subjects and two different rooms (low/high reverberation).

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SP.55 – Wed 21 Aug, 15:00-17:00

Using a spatial speech test to assess the speech discrimination and relative localisation performance of bilateral hearing aid users

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The spatial speech test was developed to simultaneously assess the ability of listeners to identify speech and determine the relative location of target sound in the presence of noise (Bizley et al., 2015). During the task, listeners hear two sequentially presented words from adjacent speakers with a 30 ° separation and they identify the word and the relative direction of the 2nd word presentation relative to the 1st. The task is performed in the presence of multiple independent noise sources at an individually determined signal-to-noise ratio. Current findings showed that the spatial location of the words have a significant effect on relative localisation performance for both normal hearing and hearing aid users. Hearing aid user's relative localisation performance was significantly worse at peripheral locations compared to normal hearing listeners. For the normal hearing listeners, performance on the word identification aspect of the test was moderated by spatial separation of the words from the noise sources. This was not the case for the hearing aid users. Most recently, the adapted Spatial Speech Test was used to assess the effects of Oticon's OpenSound Navigator on relative localisation and word identification performance in the presence of multi-talker babble.

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SP.56 – Thu 22 Aug, 15:20-17:20

Audio-Visual speaker identification

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Speech audio signals are naturally correlated with facial features of the speaker. In this study, we explore statistical relationships between different facial features and different speech audio features to identify features that can be used to match individual speakers with their corresponding audio stream in a multi-speaker scenario. Using videos with a single speaker, we extracted visual features related to movements of the mouth and other parts of the face. From the corresponding speech audio, we extracted various temporal envelope features. We then used Canonical Correlation Analysis (CCA) to learn linear transformations of the features from both the audio and the visual domain into a shared component space where features are maximally correlated. The learned feature transformations were then used to match speech audio and faces in new video data. Using this approach, we show that faces could be reliably matched to the corresponding audio streams for time segments down to 0.25 seconds. Since these features and their linear transforms are relatively fast to compute it is possible to implement these algorithms in real-time systems for audiovisual speaker identification.

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SP.57 – Wed 21 Aug, 15:00-17:00

Neural sensitivity to the statistics of natural sound textures in the inferior colliculus of rats

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Previous studies have identified a hierarchy of statistical parameters which determine the identity of auditory textures, such as running water and buzzing bees. By imposing these statistics on noise, realistic sounding textures can be synthesized. Presumably neurons in the auditory pathway must be sensitive to such statistical parameters to facilitate texture discrimination, and sensitivity to higher order statistical parameters may emerge gradually as one ascends the auditory pathway. We analyzed a database of over 200 natural textures to create 13 sound textures which span the space of parameters observed in nature. For each texture, we generated sounds, which, step-by-step, introduce parameters of increasingly higher order to morph noise gradually to natural textures. Multiunit activity was recorded from the inferior colliculus (IC) of anesthetized rats, and the responses around the transitions where new parameters were introduced were analyzed to see whether IC neurons were sensitive to the change. We observed that, while IC neurons are commonly sensitive to marginal parameters (power, variance, skew and kurtosis) and to changes in modulation power, sensitivity to cochlear correlations was relatively rare. We hypothesize that sensitivity to cochlear correlations may not become wide-spread before auditory cortex, and we intend to test this hypothesis in follow-on experiments.

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SP.58 – Thu 22 Aug, 15:20-17:20

Hearing aids in the drawer: Usage time as a function of auditory and non-auditory factors

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The BEAR project is a Danish national audiological project scheduled to run for 5 years from 2016. The overall aim of the project is to improve hearing rehabilitation in Denmark through a revision of current clinical practice. Based on results obtained in the BEAR project, a revised clinical protocol will be suggested, if possible. Potential benefits resulting from the project may have important impact both nationally and internationally. In Denmark, approximately 500,000 – 800,000 people have a treatable hearing loss and around 300,000 people own a hearing aid. However, a large portion (20%) of owners do not use their HAs regularly and the underlying reasons for this are not well understood, but one must conclude that these owners do not sufficiently benefit from their devices. This results in wasted clinical resources and a lack of rehabilitation for people with hearing-impairment. Literature suggest possible causes as to why a significant number of patients do not use their HA regularly: insufficient awareness of hearing difficulties, alternate coping strategies, personality, low trust in the benefit from hearing aids, cognitive and functional restrictions and social stigma. In this study, these literature claims were investigated with the help of the BEAR database.

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SP.59 – Wed 21 Aug, 15:00-17:00

Using the BEAR data to obtain shortened version of the SSQ-12 and IOI-HA

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The Speech, Spatial and Qualities of Hearing scale (SSQ-12) and the International Outcome Inventory for Hearing Aids (IOI-HA-7) are questionnaires containing 12 and 7 items, respectively. They are designed to subjectively assess hearing ability and are complementary to behavioral measures. Both questionnaires have been applied across a range of clinical and clinical research-related contexts, for example for assessing outcomes of e.g. cochlear implants and hearing aids. However, due to time constraints neither of the questionnaires seem to be an inherent part of standard clinical quality control. The Better Hearing Rehabilitation (BEAR) database contains SSQ-12 and IOI-HA-7 scores of around 2000 subjects. Applying an explanatory factor analysis (EFA) allowed us to reduce the SSQ-12 to 5 questions and the IOI-HA to 3 remaining questions. The SSQ-5 explains 77% of the variance in the SSQ-12 data while the IOI-HA-3 accounts for 69% of the variance in the original IOI-HA-7 dataset. We judge that these new versions can be used more efficiently by shortening time and focusing on the items that are most effective to reflect individual benefit. Furthermore, the analysis seems to confirm the validity of such a reduction from similar findings in the literature that were done on different datasets.

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SP.60 – Thu 22 Aug, 15:20-17:20

Auditory cortical representation of music favors the perceived beat

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Musical beat perception is widely regarded as a high-level ability involving widespread neural coordination across motor, cognitive, and sensory areas. However, the question of how low-level auditory processing must necessarily shape these dynamics, and therefore perception, has remained relatively underexplored. Here, we present surprising evidence that the auditory cortical representation of music, even in an absence of motor or top-down activations, already favors the beat that will be perceived. In response to twenty musical excerpts diverse in tempo and genre, firing rates in the rat auditory cortex were on average higher on the beat than off the beat as reported by 40 human listeners. This "neural emphasis" distinguished the beat that was perceived from other possible interpretations of beat, was predictive of the degree of tapping consensus across human listeners, and was accounted for by a spectrotemporal receptive field model. These findings strongly suggest that the "bottom-up" processing of music performed by the auditory system facilitates the extraction of musical beat and may substantially influence its location and clarity.

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Poster Session I

SP.61 – Wed 21 Aug, 15:00-17:00

Using response times to speech in noise to measure the influence of noise reduction on listening effort

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Single microphone noise reduction (NR) can lead to a subjective benefit even when there is no objective improvement in speech intelligibility. A possible explanation lies in a reduction of listening effort. In a previous study, we showed that response times (a proxy for listening effort) to a simple arithmetic task with spoken digits in noise were reduced (i.e., improved) by NR for normal-hearing (NH) listeners. In the current study we complemented the data set with data from twelve hearing-impaired (HI) listeners, the target group for NR. Subjects were asked to add the first and third digit of a digit triplet in noise. Response times to this task were measured and subjective listening effort was rated. Stimuli were presented at three signal-to-noise ratios (SNR) (-5, 0, +5 dB) and in quiet. Stimuli were either unprocessed or processed with ideal or non-ideal NR. Results indicate a decrease in response times and listening effort ratings for increasing SNRs for both listener groups, even when speech intelligibility is maximum. On an individual level we saw a large variation in learning effects and repeatability. We will discuss the applicability of response time measures and subjective rating for both scientific and clinical applicability.

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SP.62 – Thu 22 Aug, 15:20-17:20

Individualization of noise reduction strength: A trade-off between noise attenuation and signal distortion

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Individualization of hearing aid settings becomes increasingly more important as research has shown that preferences show large inter-individual differences. Modern hearing aids contain single microphone noise reduction (NR) of which the maximum amount of gain reduction (NR strength) can often be altered by the clinician. We assume that individual preferences for NR strength are governed by a trade-off between noise attenuation and signal distortions. In this study, we investigate this trade-off. We use a NR algorithm which allows us to separate the positive effects of NR (noise attenuation) from the negative effects (signal distortion). We use paired comparison testing to investigate how hearing-impaired listeners perceive both effects with increasing NR strength and how these effects combine into an individual preference for NR strength. We also investigate if there is a relation between this preference and other speech-in-noise outcome measures (SRT, ANL). Preliminary results will be presented and discussed at ISAAR.

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SP.63 – Wed 21 Aug, 15:00-17:00

The effect of hearing aid use on the perceptual learning of rapid speech in older-adults

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Older people, specifically ones with age-related hearing loss (presbycusis), often find it difficult to understand rapid speech in everyday life. Parts of this difficulty were attributed to age-related declines in rapid perceptual learning, but the effects of hearing-aid use on the perception and the perceptual learning of rapid speech have not been explored (to our knowledge). We compared the perception of natural-fast speech, time-compressed speech (TCS) and the rapid perceptual learning of TCS between older adults with and without experience with hearing-aids (N=43). Participants were matched in mean age, hearing thresholds and cognitive parameters, and were assessed in their individual most comfortable levels. Although there were no significant effects of hearing-aid use on either perception or learning, we did find significant rapid perceptual learning of TCS within the first 20 sentences and a significant positive correlation between this rapid learning and the perception of natural-fast speech. We also found that the perception of natural-fast speech is positively associated with vocabulary and memory and negatively correlated with hearing thresholds. These findings are consistent with the idea that age-related declines in rapid auditory learning may have an independent contribution to speech perception which is not sensory in origin.

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SP.64 – Thu 22 Aug, 15:20-17:20

A study of reliability and response patterns in self-administered audiometry for adult first-time hearing-aid users

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Out-of-clinic diagnostics offer the advantage of pre- and post-clinical screenings and potential benefits of increased user ownership, but at the possible cost of accuracy and reliability. The present study examines the determination of a classical audiometric threshold through a custom-made self-administered tablet test, utilizing off-the-shelf Bluetooth headphones. The test includes an initial familiarization session allowing characterization of the user's response time, and integrates a standard usability assessment (System Usability Scale, SUS) accompanied by the experimenter's observations and exit-interview responses. The study compares self-administered thresholds determined in the waiting room of Aalborg University Hospital for 16 potential hearing-aid users with the thresholds determined by the professionals in the subsequent session. The study also includes a comparison of thresholds determined in potential users' own homes, in which case the test is performed both with a standard transportable audiometer and with the out-of-clinic tablet system. The study is on-going.

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SP.65 – Wed 21 Aug, 15:00-17:00

Robust auditory profiling: Improved data-driven method and profile definitions for better hearing rehabilitation

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Currently, clinical characterization of hearing deficits for hearing-aid fitting is based on the pure-tone audiogram. Implicitly, this assumes that the audiogram can predict performance in complex, supra-threshold tasks. Sanchez-Lopez et al. (2018) hypothesized that the hearing deficits of a given listener, both at threshold and supra-threshold levels, result from two independent types of auditory distortions. The authors performed a data-driven analysis of two large datasets with results from several tests, which led to the identification of four auditory profiles. However, the definition of the two types of distortion was challenged by differences between the two datasets in terms of the tests and listeners used. In the Better hEAring Rehabilitation (BEAR) project, a new dataset was generated with the aim of overcoming these limitations. A heterogeneous group of listeners was tested using measures of speech intelligibility, loudness perception, binaural processing abilities and spectro-temporal resolution. Consequently, the auditory profiles of Sanchez-Lopez et al. (2018) were refined. The resultant findings are discussed in connection to previous approaches for hearing-loss classification. The updated auditory profiles, together with the investigation of optimal hearing-aid compensation strategies, may form a solid basis for efficient hearing-aid fitting.

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SP.66 – Thu 22 Aug, 15:20-17:20

Hearing-aid settings in connection to supra-threshold auditory processing deficits

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Plomp (1986) described the consequences of a hearing impairment in speech communication as the sum of two components: attenuation and distortion. Recent studies have shown that the sensitivity to spectro-temporal modulations (STM) might be linked to speech intelligibility in noise, suggesting that supra-threshold, or "internal", distortions would affect both speech and STM perception similarly. Furthermore, a reduced sensitivity to STM may also affect a listener's preference for a hearing aid (HA) compensation strategy. Here, speech intelligibility and STM sensitivity were measured in 14 hearing-impaired (HI) listeners. One group of the listeners (group A) showed an inability to detect STM, whereas the other listeners (group B) exhibited similar thresholds as listeners from a control group with young normal-hearing (NH) listeners. The three groups participated in a perceptual evaluation experiment using multi-stimuli comparisons (MUSHRA). The audio files were processed by a HA simulator fitted to the individual hearing loss, and the performance was rated in terms of four attributes: clarity, comfort, preference and listening effort. A principal component analysis showed that clarity and preference were correlated in group A whereas comfort and listening effort were correlated in group B. The classification of HI listeners in auditory profiles might be valuable for efficient HA fitting.

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SP.67 – Wed 21 Aug, 15:00-17:00

Auditory adaptation to spectral slope

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From the source to the ear drum natural sounds undergo a cascade of filtering transformations due to factors such as reverberation or sound processing in hearing devices, all of which alter the spectral shape of sounds. The auditory system must hence learn to separate source and filter while facing the fundamental uncertainty as to whether the strength of components is due to source or filter properties. Only little is known about the malleability of this aspect of auditory perception: in what way does the auditory system adapt to changes in the filter response? Here we present an experiment that tests adaptation to differences in spectral slope. Normal-hearing listeners are presented with a new (unheard) excerpt of natural speech or music per trial and judge whether the excerpt is more bright or dull than their internal reference. A filter modifies the spectrum in the range of 125-8000 Hz with linear slopes ranging from -2 to 2 dB per octave. Pilot data suggest that listeners are fairly accurate in identifying alterations of the spectral slope of unheard excerpts, but there is also rapid contrastive adaptation: excerpts are perceived as brighter when preceded by excerpts filtered with negative slopes (sounding dull) and vice versa. This work provides a window into how the auditory system separates source and filter and bears implications for the fitting of hearing devices.

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SP.68 – Thu 22 Aug, 15:20-17:20

Are adult cochlear implant recipients using the correct sound processor settings in specific listening environments?

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Previously we performed a scoping review and found only a small number of studies on the use of automatically switching devices in hearing aid users, while not a single study was identified that reported on the use of these devices in cochlear implant (CI) users. The objective of the study was to investigate the use of manually and automatically switching programs in various listening environments by adult CI users during daily use. We performed a prospective study with a crossover design in which each of the 15 CI users used manual program selection for 3 weeks with one program with an omnidirectional microphone (speech in quiet) and another program with directional microphones (speech in noise). In a different 3-week period the CIs were programmed with one automatic program (SCAN). An extensive counselling session with simulated listening situations preceded the manual program trial period. Datalog information from the sound processor was used to compare the chosen listening program to the real-life listening situations as retrieved from the datalogs. On average the CI users switched manually between programs only once per day. The program chosen by the CI users matches the listening environment approximately 50-60% of the time.

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Poster Session I

SP.69 – Wed 21 Aug, 15:00-17:00

Non-intrusive speech intelligibility prediction using harmonic beamforming

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It could be advantageous for hearing aid users, if the signal processing of their devices could automatically be optimized with respect to the speech intelligibility in the specific acoustic environment. However, most speech intelligibility metrics are intrusive, i.e. in addition to the degraded signal they require access to a clean reference signal, which is rarely available in daily use of the hearing aids. As such, this work proposes to estimate the speech intelligibility non-intrusively by using an existing intrusive objective intelligibility metric without requiring access to the clean reference signal. Instead, the principle behind the proposed method is to replace the original clean signal with an estimated reference signal as input to the intrusive metric. The reference signal is obtained from the degraded signal using a harmonic beamformer, which utilizes both the spatial characteristics and fundamental frequency content of the desired source. The results show a high correlation between the proposed non-intrusive harmonic beamforming-based metric and the original intrusive metric scores indicating that the proposed non-intrusive speech intelligibility metric is suitable for online assessment of speech intelligibility in hearing aids.

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SP.70 – Thu 22 Aug, 15:20-17:20

Tinnitus management in the digital age: The efficacy of the ReSound Relief Tinnitus app

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With the introduction of mobile applications (apps) in recent years, tinnitus sufferers have direct access to more tinnitus tools than ever before. Very few of these apps are based on clinical standards, and many offer little more than a library of sounds to play, primarily offering masking benefits, but limiting the actual tinnitus management that is offered. ReSound Relief is a tinnitus-focused app that includes not only a library of high definition sounds, but also interactive and informational exercises and meditations. In addition, counseling information is provided to educate and guide the user on how to appropriately manage their tinnitus over time. The goal of ReSound Relief is to support both the Hearing Care Professional (HCP) and patient with convenient access to tinnitus management tools and education, as they collaborate to create the best plan of action. This presentation will demonstrate the efficacy of the ReSound Relief tinnitus app and review clinical trials that demonstrate measurable benefit, using valid clinical tools, to those who use it for tinnitus management.

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SP.71 – Wed 21 Aug, 15:00-17:00

Ear canal sound recordings as a screening tool in the clinical management of patients with pulsatile tinnitus

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Ear canal sound recordings can register an objective tinnitus. Most objective tinnitus are pulsatile and in synchrony with the heartbeat, due to a vascular origin. An example of a pulsatile tinnitus is a dural arteriovenous fistulas (dAVFs). This microvascular malformation is a potentially lethal condition but can be occult on non-invasive imaging. Hence, diagnosis is important. The current golden standard to rule out a potentially hazardous dAVFs is digital subtraction angiography (DSA). However, DSA carries a procedural risk of 1-2%. We performed ear canal sound recordings with a sensitive microphone placed in the external ear canal of patients who were referred for DSA to rule out dAVFs. We found a perfect sensitivity for ear canal recordings in identifying dAVFs. Thus, a relatively simple screening tool may improve and simplify the critically-important diagnosis of pulsatile tinnitus.

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SP.72 – Thu 22 Aug, 15:20-17:20

Learning about perception of temporal fine structure by building audio codecs

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The goal of audio coding is to efficiently describe an auditory experience while enabling a faithful reconstruction to the listener. The subjective quality compared to the original is measured by established psychoacoustic tests (BS.1116, BS.1534) and the description cost is measured in number of bits. As it is much cheaper to describe coarse scale signal properties than Temporal Fine Structure (TFS), tools like noise fill, spectral extension, binaural cue coding, and machine learning have increased performance of audio codecs far beyond the first generation based on masking principles (e.g. mp3). In this evolution, implicit knowledge on hearing has been acquired by codec developers, but it has become increasingly difficult to construct tools to predict subjective quality. For example, we still don't know which aspects of the TFS that are essential for the listening impression to be preserved. To explore this issue, we perform controlled experiments with models of auditory representations with the mindset from audio coding. As in the research on summary statistics of audio textures, the goal of the synthesis is to solve the inverse problem of creating a signal with a specified representation. However, our focus is on TFS aspects of nonstationary signals, and we show how evaluating a representation by listening immediately reveals strengths and weaknesses of candidate models.

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Poster Session I

SP.73 – Wed 21 Aug, 15:00-17:00

Provoking and minimising potentially destructive binaural stimulation effects in auditory steady-state response (ASSR) measurements

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Auditory steady-state response (ASSR) measurements can be used as a clinical tool to validate hearing aid fittings. However, during a sound-field binaural measurement, unintentional dichotic stimulation may occur, e.g., due to interaural time differences (ITD's) imposed by spatial effects or processing by a single aided ear. This may reduce the ASSR response compared to monoaural measurements, possibly extending testing time or causing test false-negatives. Multiple clinically conceivable dichotic conditions were simulated using adapted three-band CE-Chirp™ stimuli each consisting of a two-octave wide chirp centred at 707 Hz presented at a 40 Hz repetition rate, and two one-octave wide chirps centred at 2 and 4 kHz both presented at similar 90Hz repetition rates. ASSRs were recorded with a clinical 2 channel EEG system for 15 subjects. Large ITDs equal to half the stimulation repetition period were found to reduce the noise corrected ASSR response to the 707 Hz two-octave chirp repeated at 40Hz by about 6dB. No reduction was seen in the ASSR to the other two stimulus chirp bands presented at 90Hz. No effect on a ASSR response was detected for an interaural signal polarity inversion or simulated lateralisations of the stimulus. Additional full-scalp recordings will be considered as a measure to optimise the electrode montage during diotic and dichotic stimulation for a clinical two-channel system.

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Poster Session II

SP.74 – Thu 22 Aug, 15:20-17:20

Sensory cue fitness in spatial auditory perception - A concept

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The research of the basic processes of spatial listening is essential for the development of immersive and plausible spatial audio reproduction systems. A current result of such developments is the adaptation of audio reproduction to the listening conditions especially in the field of augmented reality. Various studies have shown that the basic processes of hearing are still not sufficiently understood. For example, it remains open why a correlation between the localization performance and the externalization of auditory events could be determined when using a dummy head, although this correlation disappears when presenting signals with weak spectral characteristics. It has also been observed that when using dummy head and behind-the-ear BRIRs, there seems to be change in the critical directions for quadrant errors. A further conspicuity occurs when the listening room is visible and there is a disproportionate increase in the externalization of rear compared to frontal directions. These observations raise the question, which sensory cues under which circumstances and with which weightings are relevant for the creation of an auditory event and how this composition can change. This paper will provide a summary of such irregularities and presents a concept which investigates the suitability and the fitness of sensory cues in consideration of the scenario and the listening person.

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SP.75 – Wed 21 Aug, 15:00-17:00

On the trade-off between signal-to-noise ratio improvement and interaural cue preservation for binaural spatial filtering

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Designing a binaural noise reduction algorithm for hearing-aids often involves a compromise between the benefits of signal-to-noise ratio (SNR) improvement and interaural cue preservation. This study investigated the effect of a partial cue-preserving binaural beamformer on spatial perception and speech intelligibility for 14 experienced hearing-aid users with moderate to moderately severe hearing loss. Monaural beamforming and diotic binaural beamforming served as reference conditions. Speech intelligibility was assessed using the Danish DAT speech corpus. Two types of background noise were used: two-interfering-talkers and cocktail party noise. The partial cue-preserving binaural beamformer provided the listeners with the same spatial perception as monaural beamforming for the front hemisphere. For speech-on-speech with competing speakers from ±90 degrees azimuth, the benefit in terms of speech recognition score was 19.6% ($p<0.001$) for the cue-preserving binaural beamformer relative to the monaural beamformer, while the benefit was 18.4% ($p<0.001$) for the diotic binaural beamformer. There were no statistically significant differences between the obtained scores for the two binaural beamformer conditions. The results suggest that partial preservation of interaural cues at the cost of reduced SNR improvement makes it possible to maintain both good speech performance and spatial perception.

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Poster Session II

SP.76 – Thu 22 Aug, 15:20-17:20

Health-related quality of life within a cohort of hearing impaired danish adults before and after hearing aid rehabilitation

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15D is a standardized, self-administered, generic questionnaire that provides a profile (D1-15 score) and a single index score (D15-score) as a measure of health-related quality of life (HRQoL) on a scale which ranges from zero to one. 15D also includes a question related to hearing (D3). When completed before and after hearing aid (HA) treatment the questionnaire can be used to assess the potential benefits of HA treatment related to HRQoL. Overall health status was collected by questionnaires (15D and a basic health-related questionnaire) before and two months after HA fitting. The study population (n=1536) comprised first time HA users (n=1096) and experienced HA users (n=440) enrolled in the BEAR project. HA resulted in improved mean score of D3 for both first time (ΔD3: mean, SD (0.102; 0.19)) and experienced (ΔD3: mean, SD (0.083; 0.20)) HA users after 2 months of HA use. Patients with "moderate to severe" and "severe" hearing loss, experienced a significant improvement in D3-HRQoL. The study supports that HA usage has a positive effect on HRQoL when looking at the hearing dimension. Degree of hearing loss alone does not explain the positive effect observed on D3-HRQoL. Therefore, additional parameters need to be studied in order to explain essential factors for patients with HL to be able to achieve an improvement of HRQoL following HA fitting.

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Poster Session I

SP.77 – Wed 21 Aug, 15:00-17:00

Evaluation of six hearing-aid processing strategies from the perspective of auditory profiling: Insights from the BEAR project

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The current study forms part of the Better hEAring Rehabilitation (BEAR) project, which aims at developing new clinical tools for characterizing individual hearing loss and for assessing hearing-aid (HA) benefit. Its purpose was to evaluate the interaction between four auditory profiles and three measures of HA outcome obtained for six HA processing strategies. Measurements were carried out in a realistic noise environment at signal-to-noise ratios that were set based on individual speech reception thresholds ('test SNRs'). Speech recognition scores and ratings of overall quality and noise annoyance were collected in two spatial conditions. The stimuli were generated with the help of a HA simulator and presented via headphones to 60 older habitual HA users who had previously been profiled based on a data-driven approach (Sanchez-Lopez et al., Trends in Hearing 2018). The four auditory profiles differed significantly in terms of the test SNRs and interacted significantly with the HA processing strategies for speech recognition in one spatial condition. Moreover, the correlations between the speech recognition scores and subjective ratings differed among the auditory profiles. However, the HA processing strategies leading to the best outcomes were similar across the four auditory profiles.

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SP.78 – Thu 22 Aug, 15:20-17:20

Investigating the relationship between spectro-temporal modulation detection, aided speech perception, and noise reduction preference

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Hearing-aid amplification is typically adapted to the individual listener's hearing loss based on their pure-tone audiogram to restore audibility. While amplification is necessary to make sounds audible, additional supra-threshold processing deficits often result in severely degraded speech understanding and increased listening effort in adverse conditions with multiple sounds sources. To provide additional help, powerful noise reduction (NR) approaches that combine classical single-channel noise reduction with beamforming may be employed. However, while aggressive NR typically improves speech intelligibility, it may also impair the perceived spatial fidelity etc., depending on the listener. The present study investigated how a measure of supra-threshold auditory perception, namely spectro-temporal modulation detection (STMD), relates to aided speech intelligibility with different degrees of NR aggressiveness in a spatial multi-talker set up. Furthermore, it was assessed whether the individual listeners' performance levels in these tasks were related to their NR preference, assessed via a field study. The results indicate a strong connection between speech intelligibility scores and STMD performance. Further analyses will be conducted to evaluate the relationship between the laboratory-based measures and the listeners' preferences in terms of NR.

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Poster Session I

SP.79 – Wed 21 Aug, 15:00-17:00

Comparative study: Gene replacement versus antisense oligonucleotide therapy to restore hearing and balance in a mouse model of Ush1c

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Usher syndrome type 1 is associated with congenital sensorineural hearing loss, vestibular areflexia, and progressive retinitis pigmentosa (RP). A recessive Ush1c c.216G>A mutation, identified in French-Acadian patients, creates a cryptic splice site which reduces production of harmonin. Harmonin is essential for normal inner ear hair cell development and function. Our goal is to identify novel biological tools to treat auditory and vestibular deficits associated with this mutation. Adeno-associated virus (AAV) gene augmentation therapy is a promising approach to target recessive mutations of genes expressed in the inner ear. An antisense oligonucleotide (ASO) that corrects the splicing defect also partially restores auditory and vestibular function in Ush1c c.216AA mouse mutants. In this report, we describe and compare results obtained with local deliveries of either therapeutic to the inner ear through the round window membrane. Our work shows that both treatments lead to recovery of the Ush1c gene and protein expression along with restoration of hair cell function. This cellular repair promotes increased hair cell survival, rescues complex auditory function, and recovers hearing and balance behavior to near wild-type levels. The data represent unprecedented recovery of inner ear function. Comparative benefits and shortcomings of both treatments will be discussed.

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Poster Session II

SP.80 – Thu 22 Aug, 15:20-17:20

The effect of hearing loss on sound texture perception

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Sound textures, such as “rain” and “fire”, are comprised of many similar superimposed acoustic events. Imposing time-averaged statistics of real textures, measured through a model of the auditory system, onto a noise seed provides a useful means of probing the auditory system’s representation of such signals in a systematic way. Normal hearing (NH) listeners’ ability to identify sound textures increases as the number of statistic classes used in the synthesis stage increases. Discrimination tasks have also shown that normal-hearing listeners tend to use the fine temporal detail where possible, but will instead switch to a time-averaged statistical representation when discriminating textures of longer duration. This study extends the existing research to investigate how hearing-impaired (HI) listeners with age-related hearing loss perform in the same sound texture identification and discrimination tasks. Identification scores increased as the number of statistic classes used in the synthesis increased as with NH listeners, but a significant reduction in overall identification accuracy was observed. Texture discrimination performance was also reduced with hearing loss, but particularly for sounds of shorter duration. Interestingly, reduced texture identification and discrimination was not well correlated with absolute hearing thresholds, suggesting an effect of suprathreshold hearing or other factors such as cognitive decline.

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