VCCA2021

Virtual Conference on Computational Audiology

Computational Audiology
VCCA 2021 June25

SCIENTIFIC PROGRAM

Foreword

Welcome to the **Virtual Conference on Computational Audiology 2021**,

VCCA2021 provides a platform for presentation and discussion of research with focus on applications of computational and technological approaches (big data, machine learning, AI, signal processing, smart tech) to audiological interventions and hearing science (hearing research, hearing loss, hearing devices, tinnitus).

This year, the VCCA is hosted by the Cambridge Hearing Group. The program consists of 4 keynotes by research leaders, 5 featured talks by early-career scientists, 2 special sessions with invited experts and 34 scientific submissions covering a wide range of applications and research in Computational Audiology.

The goal of VCCA is to boost progress and to build an inclusive research community with a free scientific conference. VCCA aims to connect experts and researchers across disciplines and to stimulate interaction and networking in times of social distancing and travel restrictions across the globe. We hope that this will continue also after the conference on the 25th of June to maximise opportunity and impact.

It has been a great journey to organise VCCA2021 and we are very excited about the numerous contributions. We look forward to what we believe will be an outstanding program of presentations and lively discussions.

Get ready for a dense program of talks, discussions and chats, make sure you have snacks & refreshments available, and join us on **25 June 2021** virtually! We look forward to see you there.

The VCCA organising committee

The VCCA2021 organisers:

Chair:	Tobias Goehring	
Co-Chair:	Jan-Willem Wasmann	
Organisation:	Francois Guerit Charlotte Garcia Ben Williges	Alan Archer-Boyd Saima Rajasingam Andrew Harland
Advisors:	Dennis Barbour Volker Hohmann	Astrid van Wieringen Jan de Laat

Sponsors for VCCA2021

Thanks to the sponsors for their support of VCCA2021!

VCCA Scientific Program

On the next page you will find an overview schedule, followed by detailed information on all scientific contributions of **VCCA2021**.

All of the contributions, such as abstracts with additional figures and video pitches, can be found on the VCCA website.

The Conference Program:
https://computationalaudiology.com/vcca2021-conference-program/

The Abstracts:
https://computationalaudiology.com/category/vcca2021-abstracts/

We encourage you to use the VCCA website actively, please do add your feedback and/or questions via comments on the website!

SESSION TIMES		London	Paris	New York	Sydney
		GMT+1	GMT+2	GMT-4	GMT+10
KICK-OFF: VCCA2021	**10 min**	**08:00**	**09:00**	**03:00**	**17:00**
MAIN SESSION 1	**50 min**	**08:10**	**09:10**	**03:10**	**17:10**
15-MIN BREAK	15 min	09:00	10:00	04:00	18:00
PARALLEL SESSION 1	**90 min**	**09:15**	**10:15**	**04:15**	**18:15**
15-MIN BREAK	15 min	10:45	11:45	05:45	19:45
SPECIAL SESSION 1: Global Burden	**60 min**	**11:00**	**12:00**	**06:00**	**20:00**
60-MIN BREAK	60 min	12:00	13:00	07:00	21:00
SPECIAL SESSION 2: Big Data	**120 min**	**13:00**	**14:00**	**08:00**	**22:00**
30-MIN BREAK	30 min	15:00	16:00	10:00	00:00
MAIN SESSION 2	**50 min**	**15:30**	**16:30**	**10:30**	**00:30**
10-MIN BREAK	10 min	16:20	17:20	11:20	01:20
PARALLEL SESSION 2	**75 min**	**16:30**	**17:30**	**11:30**	**01:30**
45-MIN BREAK	45 min	17:45	18:45	12:45	02:45
MAIN SESSION 3	**50 min**	**18:30**	**19:30**	**13:30**	**03:30**
10-MIN BREAK	10 min	19:20	20:20	14:20	04:20
PARALLEL SESSION 3	**90 min**	**19:30**	**20:30**	**14:30**	**04:30**
AWARD CEREMONY	5 min	**21:00**	**22:00**	**16:00**	**06:00**
WRAP-UP: VCCA2021	10 min	21:05	22:05	16:05	06:05
SOCIAL & CHATS	90 min	21:15	22:15	16:15	06:15

Overview schedule

			BST
KICK-OFF: VCCA2021 (ZoomRoom A)		**10 min**	**08:00**
MAIN SESSION 1 (Room A)	Chair: Goehring	**50 min**	**08:10**
Keynote 1: **Brian CJ Moore**		30 min	
Featured 1: **Maartje Hendrikse**		20 min	
15-MIN BREAK >> JOIN PARALLEL SESSION 1/2		15 min	09:00
PARALLEL SESSION 1		**90 min**	**09:15**
P1 - Hearing loss (Room A)	**P2 - Listening (Room B)**	6x	
Chair: De Sousa	Chair: J Valderrama-Valenzuela	15 min	
15-MIN BREAK >> JOIN MAIN SESSION		15 min	10:45
SPECIAL SESSION 1: Global Burden (Room A)	Chair: Wasmann	**60 min**	**11:00**
Keynote 2: **Nicholas Lesica**		30 min	
Discussion 1: Future devices (Room A)	**Discussion 2: Future services (Room B)**	2 disc	
Chair: Archer-Boyd	Chair: Rajasingam	30 min	
60-MIN BREAK (FOOD and RELAX)		60 min	12:00
SPECIAL SESSION 2: Big Data (Room A)	Chair: Nogueira	**120 min**	**13:00**
Featured 2: **Raul Lopez-Sanchez**		20 min	
Featured 3: **Niels Pontoppidan**		20 min	
Invited expert talks: **Eikelboom, Nee & Marschollek, Vanpoucke, Kludt, Neitzel**		50 min	
Panel discussion		30 min	
30-MIN BREAK (COFFEE and RELAX) >> JOIN MAIN SESSION		30 min	15:00
MAIN SESSION 2 (Room A)	Chair: Goehring	**50 min**	**15:30**
Keynote 3: **Josh McDermott**		30 min	
Featured 4: **Josef Schlittenlacher**		20 min	
10-MIN BREAK >> JOIN PARALLEL SESSION 3/4		10 min	16:20
PARALLEL SESSION 2		**75 min**	**16:30**
P3 - Deep Learning (Room A)	**P4 - Tinnitus (Room B)**	5x	
Chair: Guerit	Chair: Salorio-Corbetto	15 min	
45-MIN BREAK (FOOD and RELAX)		45 min	17:45
MAIN SESSION 3 (Room A)	Chair: Billig	**50 min**	**18:30**
Keynote 4: **Mounya Elhilali**		30 min	
Featured 5: **Simone Graetzer**		20 min	
10-MIN BREAK >> JOIN PARALLEL SESSION 5/6		10 min	19:20
PARALLEL SESSION 3		**90 min**	**19:30**
P5 - Auditory processes (Room A)	**P6 - Auditory modelling (Room B)**	6x	
Chair: Billig	Chair: Encina-Llamas	15 min	
Award ceremony: *Young Scientist* and *Video Pitch Awards*		5 min	**21:00**
WRAP-UP: VCCA2021		10 min	

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Keynote talks

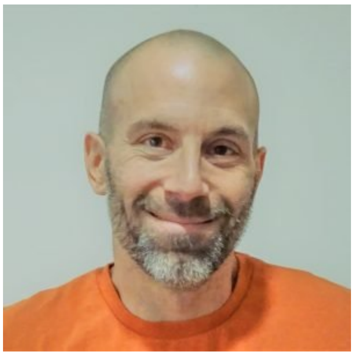
K1: Time-efficient hearing tests and their use in the fitting of hearing aids

Prof Brian CJ Moore

Emeritus Professor of Auditory Perception
Dept. of Experimental Psychology
University of Cambridge

Brian's research focuses on hearing and hearing loss, especially the perceptual analysis of complex sounds. He has played a central role in the development of models of masking and of loudness. He has made contributions to the design of hearing aids, especially amplitude compression systems. He also led the development of a method for fitting wide bandwidth hearing aids. Recently he has contributed to the development of efficient diagnostic tests of hearing. He is a Fellow of the Royal Society, the Academy of Medical Sciences, the Acoustical Society of America and the Audio Engineering Society.

K2: Harnessing the power of AI to combat the global burden of hearing loss: opportunities and challenges



Prof Nicholas Lesica

Professor of Neuroengineering
Wellcome Trust Senior Research Fellow
Ear Institute
University College London

Nick's research is focused on the study of hearing and hearing loss from the perspective of the neural code — the activity patterns that carry information about sound along the auditory pathway. He uses large-scale electrophysiology in animal models to study how hearing loss distorts the neural code and to develop new ideas for how assistive devices might correct these distortions.

K3: New models of human hearing via deep learning

Prof Josh McDermott

Associate Professor
Dept of Brain and Cognitive Sciences
Faculty Member, Program in Speech and Hearing Bioscience and Technology
MIT

Josh is a perceptual scientist studying sound, hearing, and music. His research addresses human and machine audition using tools from experimental psychology, engineering, and neuroscience. He is particularly interested in using the gap between human and machine competence to both better understand biological hearing and design better algorithms for analyzing sound.

K4: Auditory salience

Prof Mounya Elhilali

Professor and Charles Renn Faculty Scholar
Dept of Electrical and Computer Engineering
Dept of Psychology and Brain Sciences
Johns Hopkins University

Mounya's research examines sound processing by humans and machines in noisy soundscapes, and investigates reverse engineering intelligent processing of sounds by brain networks with applications to speech and audio technologies and medical systems. Her work examines neural and computational underpinnings of auditory scene analysis and role of attention and context in guiding perception and behavior.

Featured talks

F1: Virtual audiovisual environments for hearing aid evaluation (and fitting)

Dr Maartje Hendrikse[1,2]

[1] Auditory Signal Processing Group, Carl von Ossietzky University Oldenburg, Oldenburg, Germany
[2] Department of Otorhinolaryngology and Head and Neck Surgery, Erasmus University Medical Center, Rotterdam, the Netherlands

In everyday-life situations, people typically move their head while listening. This movement behavior is highly individual and can affect the performance of directional hearing-aid algorithms. In our group at Oldenburg University, we developed virtual audiovisual environments representing everyday-life listening situations. These virtual environments were used to study self-motion, i.e., focusing on the physical rather than psychological aspects of the movement behavior, of normal-hearing and hearing-impaired participants and hearing-aid users. Individual differences in self-motion were found, as well as group differences between younger normal-hearing, older normal-hearing, and hearing-impaired listeners. Wearing a hearing aid with or without an adaptive differential microphone did not affect the self-motion. Moreover, using acoustic simulations with the measured self-motion data, it was predicted how a set of common hearing-aid algorithms would perform in terms of noise-suppression in the virtual environments under the measured self-motion. Results showed that individual differences in self-motion can affect the noise-suppression performance of directional hearing-aid algorithms and that the performance of such algorithms is reduced on average compared to static laboratory-like conditions. The group differences in self-motion related to hearing-impairment were beneficial for the noise-suppression performance of an adaptive differential microphone, but hearing-aid users did not adapt their self-motion when using this algorithm. Besides explaining these results, the talk will also focus on the challenges faced when developing the virtual environments, the limitations of the virtual environments, and possibilities for the future, such as the usage in the clinic for the fitting of hearing aids and cochlear implants.

F2: Hearing deficits and auditory profiling: data-driven approaches towards personalized audiology

Dr Raul Lopez-Sanchez[1,2]

[1] Interacoustics Research Unit, Kgs. Lyngby, Denmark
[2] Hearing Systems Section, Dep. Health Technology, Technical University of Denmark, Kgs. Lyngby, Denmark

The hearing deficits are the perceptual consequences of different impairments in the auditory system. The sensorineural hearing loss is complex, and it is difficult to connect the perceptual deficits with specific impairments such as cochlear amplifier dysfunction or disruptions in the neural encoding of the incoming acoustic signals. Moreover, a detailed characterization of the auditory perceptual deficits might be more valuable than a detailed characterization of the impaired mechanisms, at least, for personalized audiology.
Previous studies have relied on correlations or regression analyses to shed light to the association between speech perception and other perceptual measures. In contrast, data-driven approaches, where an entire dataset formed by different of auditory perception is analyzed as a whole, can effectively provide patient subpopulations with distinct differences in terms of hearing abilities. This is the definition of "auditory profiling" used here.
In this contribution, the concept and principles of knowledge discovery from databases are revisited and applied on the scope of two recently published studies about auditory profiling. Here, the importance of tailoring the data-driven methods to a hypothesis towards solving a well imposed research question is stressed. The main aims were 1) to identify and characterized four clinically relevant subpopulations of hearing-impaired listeners, and 2) to explore hidden patterns of benefit in data from questionnaires about activity limitations and participation restrictions. Furthermore, these studies are analyzed in terms of findings, limitations, considerations, and perspectives towards personalized treatments.
As a final contribution, general opportunities and possible scenarios for personalized audiology are presented in association with auditory and/or audiological profiling, patterns of benefit, auditory ecology, genetics, preference profiles and other possible insights that data-driven approaches can offer for a better hearing rehabilitation.

F3: Learning from audiological data collected in the lab and the real world

Dr Niels Pontoppidan

Eriksholm Research Centre, Denmark

While audiological data is often discussed as a single matter, it encompasses a wide range of data that addresses independent steps in the research, development, and usage of hearing devices. Early large-scale collections of audiograms are a good example of the start of this era of audiological data giving rise to the first models for normal hearing and age-adjusted normal hearing, and later creating insights on how different occupations affect hearing.

On the instrument side, around 20 years ago, hearing instruments started storing aggregated information about the sound environments which they had been worn in and, the corresponding sound pressure levels, and the use of programs programmed into the hearing instruments. This audiological data along with recordings of everyday sounds, and insights from the wearers of hearing instruments suggested to increase the signal to noise ratios in audiological test paradigms. In the past few years hearing instruments connected to phones to store timestamped records of sound environments, operations, and user feedback. With sufficient coverage of everyday use, the detailed logging data enables a new type of studies where the participants evaluate the alternatives in their natural environments, while the researchers can also get some information about when and which alternative was tried in which situation.

Moreover, yet another type of audiological data goes directly into developing and tuning of audiological algorithms for the hearing instruments. Here the audiological data encompasses speech, everyday sounds, music, and various noises, and by defining the desired signals tuning the algorithms to prioritize speech, segregation, music, and or comfort.

Research in the last decades with the audiological data led to many important discoveries, and today, as the area of data emerges the focus turns to maturing those discoveries along the dimensions of coverage, applicability, bias, and privacy into solutions that improve the lives for people with hearing problems.

F4: Machine learning for models of auditory perception

Dr Josef Schlittenlacher

University of Manchester, UK

Background: Various machine-learning techniques have created new possibilities in hearing healthcare and auditory modelling. With their ability to quantify uncertainty, learn from datasets and efficient computation on parallel hardware, machine-learning techniques are particularly useful for perceptual models that are complex or incorporate individual parameters. We present three different applications to models of auditory perception: Knowledge distillation to speed up computation, combination of first principles and deep learning to model speech recognition, and Bayesian active learning of individual model parameters.

Methods: (1) A three-layer perceptron with simple rectified linear unit activation functions was trained on about 1.7 million spectra of speech and artificial sounds to predict the same loudness as the Cambridge loudness model but considerably faster. (2) An automatic speech recognition (ASR) system was built to allow for modelling of impairments in the spectral domain such as lesions but also in the time domain such as the size of temporal processing windows. It consists of causal and non-causal neural networks and a Hidden Markov Model to predict phonemes (see figure). (3) The edge frequency of a dead region and outer hair cell loss at that place were learned in a Bayesian active-learning hearing test to determine these parameters of an individual model for audibility.

Results: (1) Predictions were accurate for all kinds of sounds, and a 24 hour-recording can be processed within minutes on a graphics processor unit; the reference model takes about 50 times real time. (2) The ASR system is a good predictor for the speech-recognition and phoneme-specific performance of cochlear-implant users. (3) The test was able to identify the individual parameters within 15 minutes.

Conclusions: Machine learning has various applications in auditory modelling and the approaches combined will transform individual testing and processing in hearing devices.

F5: Clarity: machine learning challenges for improving hearing aid processing of speech in noise

Dr Simone Graetzer

Acoustics Research Group, University of Salford, UK

Background: In recent years, rapid advances in speech technology have been made possible by machine learning challenges such as CHiME and Hurricane. In the Clarity project, the machine learning approach is applied to the problem of hearing aid processing of speech-in-noise. In the first round, the scenario is a simulated cuboid-shaped living room in which there is a single static listener, target speaker and interferer. The entrants' tasks are to improve the target speech intelligibility at the listener's ear and to predict it. The software provided includes a generative tool and baseline hearing aid, hearing loss and speech intelligibility models. The databases provided include a large speech database. This is the first machine learning challenge to consider the problem of hearing aid speech signal processing.

Method: The target is within +/- 30 degrees azimuth of the listener, inclusive. The interferer is an omnidirectional point source (speech or domestic noise) at the same elevation. Signals are processed by the baseline hearing aid model, a configuration of the openMHA system for a simple Behind-The Ear (BTE) model. Subsequently, signals are passed through the baseline hearing loss model and the speech intelligibility model (Modified Binaural Short-Time Objective Intelligibility or MBSTOI). Challenge entrants replace the baseline hearing aid or the hearing loss and/or speech intelligibility models.

Results: Baseline system performance measured for the development dataset as MBSTOI values as a proxy for measured intelligibility scores varies according to, for example, SNR and distances between the listener and target.

Conclusion: The first round of the Clarity challenges aims to produce improved hearing aid algorithms for speech signal processing. The first enhancement challenge involves developing hearing aid models that improve on the baseline, while the first prediction challenge involves improving on the baseline speech intelligibility prediction models.

Special Sessions

Special Session 1: Global burden of hearing loss

We will host two interactive discussion sessions (in parallel) after the keynote talk by Prof Lesica on the global burden of hearing loss and how future services and devices could help. Please join the session and provide your ideas and thoughts, we look forward to thought-provoking discussions.

Discussion 1: Hearing services of the future

Session Chair: Dr Saima Rajasingam, Anglia Ruskin University, UK

Discussion 2: Hearing devices of the future

Session Chair: Dr Alan Archer-Boyd, MRC CBU, UK

Special Session 2:
Big Data and data-driven analyses

The second special session builds on the two featured talks by Dr Sanchez-Lopez and Dr Pontoppidan to present novel applications of Big Data and Data-based analyses for Computational Audiology. It will feature five invited talks by experts from academic and industrial research backgrounds to showcase a range of data-focused applications and investigations. After the talks, Session Chair Prof Nogueira will host a panel discussion with the presenters to conclude the special session.

Session Chair: Prof Waldo Nogueira-Vazquez

B1: We have lots of data – now what?

Rob Eikelboom

Ear Science Institute Australia

There is no doubt that the move away from paper-based health records to electronic health records has been important for many reasons. Over time the amount of data, also in ear and hearing health, has grown immensely. However, when researchers have started to examine these data, a number of challenges have become apparent.
These challenges include: (i) Ceiling values in audiometry and in non-adaptive speech perception tests. For example, how do we handle the thresholds that are recorded at the limit of audiometers? (ii) Accessing data in proprietary data databases. For example, there is public and private data in NOAH that is available when viewing clients, but multiple records cannot be exported. (iii) Linking records of clients that reside in different databases. For example, in the absence of a common identification code, variations and errors in the recording of names and other identifying information hamper linking records. (iv) Limitations in the available data fields to answer key questions. Large databases are attractive to big data/AI projects, but are predictive models improving? (v) Correct application of statistical tools; and (vi) Ethics of using client data. How do researchers deal with the growing awareness of data ownership and confidentiality, and satisfy the requirement of legislation and ethics committees?
The presentation will illustrate each of these challenges with real-life examples, and potential strategies to overcome these.

B2: The HiGHmed approach for FAIR use of clinical and research data with openEHR – Focusing on interoperability

Sarah Nee, Michael Marschollek

Peter L. Reichertz Institute for Medical Informatics, TU Braunschweig and Hannover Medical School, Germany

Patient-centric medical research benefits from the sharing and integration of complex and diverse data from different sources such as care, clinical research, and novel emerging data types. The integration and use of the data is typically impeded by its diversity or even incompatibility in terms of quality, type, and format. To facilitate internationally interoperable data integration and methods, HiGHmed is developing a generic and scalable open health data platform. The architecture is based on relevant existing standards such as openEHR (an open standard specification for electronic health records) and the FAIR (findable, accessible, interoperable, reusable) principles. The vendor- and technology-neutral standardized framework openEHR, ensures semantic interoperability of data with the separation of the data model from the content model through multi-level modeling.

B3: Towards Smarter and Connected Hearing Implant Care

Filiep Vanpoucke

Cochlear Technology Centre, Mechelen, Belgium

Connectivity is transforming the hearing care model significantly. Wireless links turn hearing instruments and implants effectively into Internet of Medical Things (IoMT) devices, connecting the user 24/7 to their caregivers and peers. This opens unprecedented routes to higher value creation. Patients may benefit from better hearing outcomes and user experiences. Health care professionals may take better care decisions and access new capabilities such as remote monitoring. And providers and payers may benefit from higher efficiency. Data – and turning these data into useful insights and tools - is at the heart of this digital transformation. AI has the potential to support HCPs – augmented intelligence – and to significantly empower non-expert users in taking on more responsibility. Bigger and richer connected data and machine learning (ML) will allow to make progress on hard problems such as managing outcome variability. Although the future is bright, major hurdles exist. Establishing and maintaining trust among all parties is arguably the biggest one. To truly reap these benefits, users must be willing to share more data about how they use their device and how well they perform, such that these can serve as inputs to smart algorithms. Without a deeper, more precise understanding of their needs, only shallow group-level solutions can be offered. Also, clinicians need the assurance that ML-powered tools are trustworthy. Where risks exist, regulators must play a role in establishing credibility, e.g. by extending traditional design controls to include the data sets underlying machine learning. The role of manufacturers is evolving. In response to the rightful expectation - accelerated by the pandemic - to provide connected care solutions, they are building digital platforms where hearing data can be shared among selected stakeholders in the hearing journey. This is happening in a rapidly evolving context, where governments are trying to keep up with the AI space updating ethical, privacy and medical device regulations and standards. Given this complexity, the first digital platforms will likely be relatively closed. However, given the need for true collaboration and deep partnerships among stakeholders, including the research community, an evolution is expected towards more open ecosystems. Standards can play an enabling role to accelerate our learnings.

B4: Twenty-five years of clinical data collection: from a single site relational database towards multi-site interoperability

Eugen Kludt, Andreas Büchner

Department of Otolaryngology, Medical University of Hannover, Hanover, Germany

Although noise and music exposures are extensive and widespread, there is a paucity of objective, large-scale data available regarding personal music and noise exposure levels and patterns, as well as the health impacts of these exposures. The University of Michigan School of Public Health has partnered with Apple Inc. to use advances in smart device and wearable technology to evaluate the levels of sound at which iPhone users listen to music and other media, as well as how long and how often they listen. Our study will also evaluate hearing threshold levels among participants, and, among the subset of participants who wear Apple Watches, will measure environmental noise levels and collect heart rate information, as well. The unique, crowdsourced big data resulting from this study will allow us to create national-level estimates of adult exposures to music and environmental sound. The information collected will also help give us a clearer picture of the impacts of music and noise exposures on hearing and cardiovascular health in adults, and ultimately inform efforts to address and reduce the public health impacts of these exposures.

B5: Big Data and the Apple Hearing Study

Richard Neitzel

Department of Environmental Health Sciences, School of Public Health, University of Michigan, US

Although noise and music exposures are extensive and widespread, there is a paucity of objective, large-scale data available regarding personal music and noise exposure levels and patterns, as well as the health impacts of these exposures. The University of Michigan School of Public Health has partnered with Apple Inc. to use advances in smart device and wearable technology to evaluate the levels of sound at which iPhone users listen to music and other media, as well as how long and how often they listen. Our study will also evaluate hearing threshold levels among participants, and, among the subset of participants who wear Apple Watches, will measure environmental noise levels and collect heart rate information, as well. The unique, crowdsourced big data resulting from this study will allow us to create national-level estimates of adult exposures to music and environmental sound. The information collected will also help give us a clearer picture of the impacts of music and noise exposures on hearing and cardiovascular health in adults, and ultimately inform efforts to address and reduce the public health impacts of these exposures.

Scientific Submissions (Parallel sessions)

Session P1: Hearing loss detection, monitoring & prevalence

Session Chair: Karina de Sousa, University of Pretoria, South Africa

P1-1: Detecting hearing loss from children's speech using machine learning

Jessica Monaghan, David Allen

National Acoustic Laboratories, Sydney, Australia

Background: Undiagnosed hearing loss in children can have serious consequences for the development of speech and language skills, ultimately impacting educational and social outcomes as well as quality of life. In many countries, including Australia, newborn screening for hearing loss is common or even mandated. However, over 50% of children diagnosed with hearing loss are identified subsequent to newborn hearing screening, either because their hearing loss was too mild to detect at birth or developed later. Since no widespread screening of preschool or school-aged children exists, hearing loss during the early years may go undetected. In addition to sensorineural hearing loss, preventable and potentially reversible hearing loss in the form of 'glue' ear affects 1 in 5 preschool children at any given time, and is especially prevalent in Aboriginal and Torres Strait Islander children. Evidence indicates parents are not able to identify reliably even severe hearing loss in their children, so relying on their suspicions of hearing impairment is not sufficient to address the problem of undiagnosed childhood hearing loss. Testing preschool children, particularly in the age range 2-3 years, is time consuming and requires specialist equipment and expertise, making large-scale testing unfeasible.

Results & Conclusion: Although the trajectory of speech development varies widely across children, hearing loss will nevertheless affect the perception and, therefore, the production, of different phonemes differently, depending on spectral frequency. Other characteristics such as pitch or rate of speech may also be altered by hearing impairment. These differences may be too subtle for humans to observe, but our preliminary analysis indicates that deep-learning techniques can be used to differentiate the speech of children with and without hearing loss. This approach has the potential to be used as a convenient screening tool for children or to monitor speech development in children at risk of otitis media.

1 sentence summary:
Here we explore the potential for using machine learning to detect hearing loss from children's speech.

P1-2: Hearing test using smart speakers: Speech audiometry with Alexa

Jasper Ooster[1,2], Melanie Krueger[2,3], Jörg-Hendrik Bach[2,3,4], Kirsten C. Wagener[2,3,4], Birger Kollmeier[2,3,4,5], Bernd T. Meyer[1,2,3]

[1] Communication Acoustics, Carl von Ossietzky Universität, Oldenburg, Germany
[2] Cluster of Excellence Hearing4all, Germany
[3] HörTech gGmbH, Oldenburg, Germany
[4] Hörzentrum GmbH, Oldenburg, Germany
[5] Medizinische Physik, Carl von Ossietzky Universität, Oldenburg, Germany

Background: Speech audiometry is an important tool to quantify impaired hearing of listeners. Established tests often require substantial resources since an experimenter is required to evaluate the subjects' responses. In addition, calibrated audiometric hardware is required. Smart speakers provide a speech-based interface and could increase the accessibility of tests but could also introduce errors in the response logging. In our study (Ooster, et. al. 2020, TiH, doi:10.1177/2331216520970011), the speech recognition threshold (SRT) is measured with a matrix sentence test implementation for Amazon's Alexa, which is compared to a clinical setup.

Methods: 46 German subjects from 4 different groups are measured in 3 different acoustic conditions to evaluate the influence on the SRT accuracy with a smart speaker. The subject groups range from young, normal-hearing (NH) to elderly, moderately hearing-impaired (HI) subjects. The acoustic conditions are generated in a room with variable, electro-acoustically controlled, reverberation times, ranging from low to high reverberation.

Results: The SRT results were mostly constant over the acoustic conditions. The bias between the SRT measured with the smart speaker and the clinically measured SRT varied with the subject groups from +0.7 dB for elderly, moderately HI listeners to +2.2 dB for young NH listeners. The intra-subject standard deviation was close (0.2 dB) to the test-retest accuracy of the clinical test. An ROC analysis showed that the approach produces a high accuracy (AUC=0.95) when making a binary decision (normal or elevated SRT).

Conclusions: We developed a skill for performing a speech-based listening test, which is currently available on all Alexa devices in German-speaking countries (activation with "Alexa, starte Hörtest"). Although it is not as accurate as the clinical test, an AUC of 0.95 is achieved for deciding if the SRT is normal or not. This test could complement clinical testing.

1 sentence summary:
We present an Alexa skill that performs a speech-in-noise listening test with matrix sentences. The skill is evaluated with four subject groups and in three different acoustic conditions.

P1-3: Evaluation of multivariate classification algorithms for hearing loss detection through a speech-in-noise test

Marta Lenatti[1], Edoardo M. Polo[2,3], Martina Paolini[3], Maximiliano Mollura[3], Marco Zanet[1], Riccardo Barbieri[3], Alessia Paglialonga[1]

[1] Consiglio Nazionale delle Ricerche (CNR), Istituto di Elettronica e di Ingegneria dell'Informazione e delle Telecomunicazioni (IEIIT), Milan, Italy
[2] DIAG, Sapienza University of Rome, Rome, Italy
[3] Politecnico di Milano, Dipartimento di Elettronica, Informazione e Bioingegneria (DEIB), Milan, Italy

Background: Online speech-in-noise screening tests are becoming increasingly popular as means to promote awareness and enable early identification of age-related hearing loss. To date, these tests are mainly based on the analysis of a single measure, that is the speech reception threshold (SRT). However, other features may provide significant predictors of hearing loss.
The aim of this study is to address a hearing screening procedure that integrates a novel speech-in-noise test that can be used remotely in individuals of unknown language and artificial intelligence (AI) algorithms to analyze features extracted from the test.

Methods: In addition to the SRT, estimated using a newly developed staircase, our system extracted features such as percentage of correct responses, average reaction time, and test duration from 177 tested ears (including 68 ears with slight/mild or moderate hearing loss). These features were fed into a collection of AI algorithms including both explainable (XAI, e.g. Decision Tree) and conventional methods (e.g., Logistic Regression, Support Vector Machines) to train a multivariate classifier identifying ears with hearing loss.

Results: Our AI-based multivariate classifiers achieved better performance and sensitivity (e.g., Logistic Regression: AUC=0.88; sensitivity = 0.80; specificity = 0.77) when compared to a conventional univariate classifier based on SRT (cut-off: -8.87 dB SNR; AUC=0.82; sensitivity = 0.75; specificity = 0.81).
According to XAI methods, in addition to SRT, other features like the number of correct responses and age were relevant in identifying slight/mild or higher degree of hearing loss.

Conclusion: The proposed hearing screening procedure showed good performance in terms of hearing loss detection. Ongoing research includes the implementation of an icon-based module to assess additional features, specifically risk factors for hearing loss (e.g., noise exposure, diabetes), that will be validated on a large population.

This study was partially supported by Capita Foundation (project WHISPER, Widespread Hearing Impairment Screening and PrEvention of Risk, 2020 Auditory Research Grant).

1 sentence summary:
A new online hearing screening procedure integrated with artificial intelligence for the identification of slight/mild hearing loss in older adults.

P1-4: Model-based selection of most informative diagnostic tests and test parameters

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Background: Given the complexity of most brain and body processes, it is often not possible to relate experimental data from an individual to the underlying subject-specific physiology or pathology. Computer simulations of these processes simulate experimental data but rarely address the inverse problem, i.e. to identify a pathology from experimental data. Even if the inverse problem is addressed, typically a single model parameter is varied to fit previously recorded experimental data. After the fit, confidence intervals are given in the units of the experimental data but usually not for the model parameters that are the ultimate interest of the diagnosis. Our long-term goal is that models allow for a quantitative diagnosis and guide the diagnostic process.

Methods: We propose a likelihood-based fitting procedure, operating in the model parameter space and providing confidence intervals for the parameters under diagnosis.

Results: The procedure is capable to run in parallel with the measurement and can adaptively set stimulus parameters to the values that are expected to provide the most diagnostic information. By predefining the acceptable diagnostic tolerance, i.e. the confidence intervals, the experiment continues until the goal is reached. As an example, the procedure is tested with a simplistic three-parameter auditory model and a psychoacoustic binaural tone in noise detection experiment. The model-based measurement steering provided 80% more information for a given number of trials, compared to a conventional maximum likelihood measurement.

Conclusion: We conclude that model-based experiment steering is possible and has at least theoretical advantages over sequential measure-and-fit approaches. Practical problems and the lack of sufficiently accurate models are going to prohibit most diagnostic applications for the time being.

1 sentence summary:
The model should conduct the experiment because it knows best which condition is going to be the most informative for confining its free parameters, at least in theory.

P1-5: Examining the association of standard threshold shifts for occupational hearing loss among miners exposed to noise and platinum mine dust at a large-scale platinum mine in South Africa

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Background: The prevalence of occupational noise induced hearing loss is associated with various risk factors that affect workers employed by the South African mines. We aimed to examine the association of standard threshold shifts (STS) with exposure to noise and platinum mine dust (PDM) for miners at a platinum mine in South Africa, using demographic data and five years (2014 to 2018) of annual screening audiometry results, noise and dust exposure data (averaged exposure data).

Methods: This was an analysis of secondary retrospective data collected by the mine. Miners' age, sex, PLH, and dust and noise exposure data were described, after which we developed a linear mixed effects regression model to predict STS. Data that had been collected from 2014 to 2018 were analysed (N=12 692); average occupational exposure levels to noise and dust were calculated from recorded measurements.

Results: Most miners were male (89.6%) and more than 50% younger than 41 years. More than 70% were exposed to > 85 dBA noise and 58% exposed to PMD from 1.5 to 2.99 mg/m3. Changes in hearing levels ranged from 8.3 dBHL at baseline (2014/2015) to 10 dBHL in 2016 (high frequency average, HFA234); with no changes thereafter. Linear mixed effects model estimated exposure associated with sex (male) was 27% and 21% for the left and right ear, respectively. The estimated effect of age, PLH, noise exposure and years of exposure was < 10% for each variable. There was no statistically significant association between PMD and STS in hearing.

Conclusion: Age, sex, years of exposure to noise, and noise exposure levels combined effects and strength of association can be used to predict STS for this group of miners. Our findings may be used to measure the mine's HCP efficiency and prevention of ONIHL among miners.

1 sentence summary:
The association of standard threshold shifts for occupational hearing loss among miners exposed to noise and platinum mine dust at a large-scale platinum mine in South Africa

P1-6: Prevalence statistics of hearing loss in adults: Harnessing spatial big data to estimate patterns and trends

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Background: Hearing loss is estimated to affect over eight million adults aged over 50 in England. These estimates are based on the prevalence data reported in the 'Hearing in Adults' study by Davis, who collected audiological data for 1,538 subjects 50 years old and above in the 1980s. The prevalence (%) per age group from this study's samples is being applied to the most recent English census. We aimed to (a) explore regional patterns and trends of hearing loss in a representative longitudinal prospective cohort study of the English older population, and (b) identify potential regional differences with the current prevalence estimates.

Methods: We utilised the full dataset (74,699 person-years) of the English Longitudinal Study of Ageing (ELSA). We used local spatial statistics to analyse spatial distributions, patterns and trends of the geographical data, examining both self-reported and objective hearing data. The objectively identified hearing loss was defined as greater than 35 dB HL at 3.0 kHz, in the better hearing ear, as measured by a handheld audiometric screening device (HearCheck Screener).

Results: There was a wide variation in hearing loss prevalence in representative samples from different regions in England with similar age profiles (Fig 1.). In a period of 15 years (2002-2017) the increase rate of hearing loss ranged between regions from 3.2% to 45%. The Getis-Ord Gi* spatial statistic showed marked regional variability, and hearing health inequalities between Northern and Southern England that were previously unknown.

Conclusion: The profound multidisciplinary professional and experimental care in the 'Hearing in Adults' study by Davis is broadly recognised; however, our study showed that this data does not remain valid and generalisable. The time of small-scale research should be consigned to the past; applying computational approaches in audiology might offer promising solutions to generate large-scale epidemiological inferences to improve population's hearing health.

1 sentence summary:
Harnessing spatial big data to estimate patterns and trends of hearing loss

Session P2: Listening effort, behaviour & intervention

Session Chair: Dr Joaquin Valderrama-Valenzuela, NAL, Australia

P2-1: A classification approach to listening effort: combining features from the pupil and cardiovascular system

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Background: Physiological markers of autonomic nervous system activity are evaluated to reflect listening effort (LE). Different physiological measures are often uncorrelated with one another and with participants' subjective ratings. This mismatch may arise because the measures reflect different aspects of LE. Here we trained a classifier using pupil and cardiovascular features to predict the experimental condition. Another classifier was trained with the same features, to predict the participant's subjective perception during the experiment.

Methods: 29 hearing-impaired listeners undertook a speech-in-noise task (Danish HINT) at two signal-to-noise ratios, individually adapted to 50 and 80% intelligibility levels. The task was performed alone and in the presence of two observers. Per sentence, seven features were extracted, four from the cardiovascular system (inter-beat interval, pulse transit time, blood volume pulse amplitude and pre-ejection period) and three from the pupil (peak pupil dilation, mean pupil dilation and baseline pupil size). After the experiment, participants gave a semi-structured interview describing the experience. The interview transcripts were reviewed to determine whether each participant was affected by the the observers' presence (yes/no). The seven physiological features were fed into k-nearest neighbor classifiers with 50-fold cross validation, to predict 1) the social state and intelligibility and 2) if the participant was affected by the observers' presence.

Results: The social state (alone/observed) was predicted with an accuracy of 77.5% and the intelligibility (50/80%) was predicted with an accuracy of 61.2%. Different features contributed to these models. The participants' response to the observers was predicted with an accuracy of 92.9%.

Conclusion: A combination of features may be preferable to a single dependent variable during LE studies. Some features were better suited to predicting intelligibility and others to predicting observers' presence.

1 sentence summary:
Training k-nearest neighbor classifiers to predict intelligibility, social state and participant perception of a listening task

P2-2: Assessing listening effort, using EEG and pupillometry, in response to adverse listening conditions and memory load.

Loes Beckers[1,2], Svetlana Gerakaki[3], Marc van Wanrooij[3], Birgit Philips[1], Wendy Huinck[2], Emmanuel Mylanus[2]

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[3] Department of Biophysics, Donders Institute for Brain, Cognition and Behaviour, Radboud University, Nijmegen, NL

Background: Profoundly deaf adults are currently treated with a cochlear implant (CI) yielding substantial hearing benefits. Nevertheless, speech understanding of CI users remains perturbed, especially in noisy environments and for those CI users suffering from considerable auditory-nerve degradation. Under these demanding situations, non-acoustic sources of information, such as prediction and memory, may become important. Here, we studied how effortful listening becomes, when neurocognitive mechanisms need to be activated while listening to a distorted speech signal through a CI vocoder.

Methods: Previous studies by Obleser et. al., (J. NeuroSci. 2012 vol:32) and Petersen et. al., (F. Psych. 2015, vol:6), implemented a suitable protocol to investigate inhibition, a mechanism involved in ignoring task-irrelevant information, and working memory load in listeners with normal hearing and various degrees of hearing loss. Neuronal alpha-band activity, associated with inhibition and storage, was assessed using EEG during an auditory digit working memory task, in which memory load and acoustic degradation were manipulated. We replicated this paradigm in normal hearing listeners and extended the paradigm by measuring pupil dilation.

Results: This is an ongoing study.

Conclusion: Based on the replicated study, we expect that alpha activity in the parietal lobe increases with increasing memory load and acoustic degradation. Furthermore, we expect that listening effort, and pupil dilation as its proxy, has a non-monotonic relationship to memory load and acoustic degradation. And that individual differences in coping strategies are reflected in pupil dilation and oscillation measures. The results of our study will help to aid audiological assessment and revalidation of CI users, as we will apply a similar paradigm to adult CI users to explore variability in speech perception performance due to top-down neurocognitive factors.

1 sentence summary:
This ongoing study aims to investigate how effortful listening becomes, when neurocognitive mechanisms need to be activated while listening to a distorted speech signal through a CI vocoder, using EEG and pupillometry during an auditory digit working memory task.

P2-3: Automatic detection of human activities from accelerometer sensors integrated in hearables

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Background: Human activity tracking has gained momentum in recent years driven by the development of powerful mobile devices with integrated movement sensors. Among these are ear-level devices that have proven to produce rich data regarding physical motion and vital parameters. It seems likely that functionalities of hearing devices and personal health systems will continue to merge in the near future. As of now, many research areas investigate the informational content of movement sensors in small, body-worn devices. With this project, we add to the growing fields of health monitoring and ambulatory assessment by exploring a method for activity detection using hearables manufactured by Dopple BV (NL).

Methods: Pilot data from two individuals were collected using a 3-axis accelerometer integrated in the hearables. We used research prototypes with direct access to raw sensor data. Data were streamed via Bluetooth to a PC for processing. Participants performed 12 activities for 5 minutes each. After outlier removal, balancing between classes and scaling, data were classified using a Naïve Bayes classifier implemented in sklearn using a 5-fold stratified cross-validation procedure.

Results: Using mean acceleration of all three channels computed on subsequent windows of 1 second, activities were detected with 72 % and 67 % accuracy on average for each participant. The averaged confusion matrix across folds shows that accuracy varies over activities, which might be explained by the different statistical properties of the activities. In addition, systematic confusion between similar activities may occur.

Conclusion: We found accelerometer data from sensors integrated in a hearable and worn in the ear canal to be suitable for the detection of human activities in our pilot sample. More work is needed to further investigate real-time capabilities of our approach, but also the generalisation and robustness of our machine learning approach.

1 sentence summary:
Using a Naive Bayes classifier, we could show that twelve different activities were classified above chance.

P2-4: Sound-Level monitoring earphones with smartphone feedback as an intervention to promote healthy listening behaviors in young adults

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Background: More than a billion adolescents and youngsters are estimated to be at risk of acquiring recreational noise-induced hearing loss (RNIHL) due to the unsafe use of personal audio systems (PAS). RNIHL is preventable, therefore, the present study aimed to determine (i) the accuracy and reliability of dbTrack (Westone) sound-level monitoring earphones and (ii) the effect of sound-level monitoring earphones with smartphone feedback and hearing health information on listening behaviors in young adults.

Methods: The study consisted of two phases, phase 1 investigated the accuracy and reliability of sound-level monitoring earphones. Accuracy was determined by comparing earphone measurements to sound level meter measurements. Intra-device and within-subject reliability were determined by comparing sound measurements. For phase 2 of the study, a single-group pretest-posttest design was utilised. Participants utilized sound-level monitoring earphones with an accompanying dbTrack smartphone application for 4 weeks. The application's smartphone feedback was disabled during the first 2 weeks (pretest condition) and enabled during last 2 weeks (posttest condition).

Results: Phase 1 dbTrack earphone measurements were within 1 dB when compared to sound level meter measurements. Earphones were also within 1 dB in repeated measures across earphones and across participants. Phase 2 posttest average daily intensity decreased by 8.7 dB (18.3 SD), duration decreased by 7.6 minutes (46.6 SD) and sound dose decreased by 4128.4% (24965.5% SD). Differences in intensity and sound dose were significantly lower with a small and medium effect size, respectively.

Conclusions: Sound-level monitoring earphones with a calibrated in-ear microphone can reliably and accurately measure PAS sound exposure. Feedback on sound exposure using accurate sound-level monitoring earphones with an accompanying app can promote safe listening behavior in young adults and reduce the risk of acquiring RNIHL.

1 sentence summary:
In a first of its kind study, we aimed to determine the accuracy and reliability of sound-level monitoring earphones and the effect of smartphone feedback as an intervention to encourage safe listening use among young people.

P2-5: How variation in cochlear implant performance relates to differences in MAP parameters

Enrico Migliorini[1,2], Bastiaan Van Dijk[1], Birgit Philips[1], Emmanuel Mylanus[2], Wendy Huinck[2]

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Background: Cochlear Implants are effective devices for the restoration of hearing capabilities in profoundly deaf people. However, CI recipients with similar age, aetiology, clinical history, implanted device and residual hearing may have vastly different speech perception outcomes. The objective of this study is to investigate the relation between this unexplained variability and mapping levels through a retrospective analysis of the fitting data from the ENT department of Radboud university medical center Nijmegen.

Methods: The anonymized database contains both CVC speech audiometry tests (NVA) collected from CI recipients with a Nucleus™ CI and corresponding mapping information from the CustomSound™ fitting software. First a principal component analysis (PCA) was performed to reduce the dimensionality of the MAP dataset, and the subjects were split into tertiles based on performance (lowest tertile;<66% phoneme score, highest tertile; >87% phoneme score). Consequently two test were performed on the data: 1) Differences in the PCA components between subjects in the highest and lowest tertile was checked with the Mann-Whitney-Wilcoxon's u test; 2) Spearman's and Pearson's correlation coefficients were calculated in order to find correlations between performance and the maps' components.

Results: We found statistically significant differences between the average maps of subjects in the highest and lowest tertile.

Conclusion: We found clinically significant differences between the populations, with an effect size large enough to potentially allow to develop a fitting intervention to address poor speech recognition. In this conference contribution we will discuss the results, the interpretation and suggest follow-up work.

1 sentence summary:
Statistical analysis of how fitting parameters relate to speech recognition scores finds meaningful differences between the highest- and lowest-scoring tertiles of recipients.

P2-6: Designing the BEARS (Both Ears) virtual reality training suite for improving spatial hearing abilities in teenage bilateral cochlear implantees

Lorenzo Picinali[1], Kevin Sum[1], Jordi Albanell Flores[1], Bhavisha Parmar[3], Sandra Driver[2], Chris Rocca[2], Marina Salorio-Corbetto[3], Dan Jiang[2], Deborah Vickers[3]

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Background: teenagers with bilateral cochlear implants (CI) often suffer from poor spatial hearing abilities, which arise from their difficulties to fuse sounds from the two ears. This deficit jeopardises speech and language development, education and social well-being, and is worsened by the lack of protocols for fitting bilateral cochlear implants and of resources for spatial-hearing training. A large body of research demonstrates that sound localisation can improve with training, underpinned by plasticity-driven changes in the auditory pathways. Maximal benefit for generalising training to non-trained auditory skills is best achieved by using a multi-modal (audio-visual) implementation and multi-domain training tasks (localisation, spatial speech-in-noise and spatial music). The goal of this work was to develop, using an action research protocol, a package of VR games (BEARS, Both EARS) to train spatial hearing in teenagers with bilateral CI.

Method: formalised cycles were used for patient participants to experience a prototype of the BEARS games and provide feedback, which was then transformed in system requirements for further developments.

Results: the main areas which were modified based on participatory feedback were the variety of immersive scenarios to cover a range of ages and interests, the organisation of levels to ensure small improvements were measured and rewarded, and specific provision for participants with balance issues, who had difficulties when using head-mounted displays. The effectiveness of the finalised BEARS suite will be evaluated in a large-scale clinical trial.

Conclusions: interventions such as the BEARS allow patients to take control of their own management thus reducing the reliance upon outpatient-based rehabilitation programmes. Specifically for teenagers, a VR implementation is more engaging than traditional rehabilitation methods, and the participatory design approach has ensured that the BEARS games are relevant and engaging.

1 sentence summary:
Teenagers with bilateral cochlear implants (CI) often suffer from poor spatial hearing abilities; a set of multi-modal (audio-visual) and multi-domain training tasks (localisation, spatial speech-in-noise and spatial music) was designed by involving teenage CI users as co-creators during the development process.

Session P3:
Deep learning applications and models

Session Chair: Dr François Guérit, MRC Cognition and Brain Sciences Unit, UK

P3-1: Estimating the distortion component of hearing impairment from attenuation-based model predictions using machine learning

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Background: Hearing impairment affects the ability to understand speech. This can be described according to Plomp (1978) by an attenuation and a distortion component. The attenuation component affects speech recognition in quiet and is linked to the absolute hearing threshold. The supra-threshold distortion component affects speech recognition in quiet and noise while its origin remains speculative. Such supra-threshold deficits cannot be compensated by amplification and no “simple” measurement method exists. Yet, separating the attenuation component from the distortion component appears possible by using speech recognition models: When only using the attenuation component for modeling, differences between predicted and measured SRTs can be interpreted as an estimate of supra-threshold deficits.

Methods: Published speech recognition thresholds (SRTs) in noise of 315 hearing-impaired ears were predicted with the machine-learning-based framework for auditory discrimination experiments (FADE), the speech intelligibility index (SII), and a modified SII with a hearing-loss-dependent band importance function (PAV). Their attenuation-component-based prediction errors were interpreted as estimates of the distortion component.

Results: Overall, the models showed root-mean-square errors (RMSEs) of 7 dB, but for steeply sloping hearing loss FADE and PAV were more accurate (RMSE=9 dB) than the SII (RMSE=23 dB). Prediction errors of FADE and PAV increased linearly with the average hearing loss. This linear relation was used as distortion component estimate whose application significantly improved the accuracy of FADE’s and PAV’s predictions.

Conclusion: Simulations with FADE and PAV imply, that the supra-threshold distortion component increases linearly with average hearing loss. Accounting for a distortion component improves the model predictions and implies a need for effective compensation strategies for supra-threshold processing deficits with increasing audibility loss.

1 sentence summary:
Attenuation-component-based model predictions of speech recognition thresholds like FADE seem to facilitate an estimation of the supra-threshold distortion component of hearing impairment.

P3-2: Comparing phonemic information transmission with cochlear implants between human listeners and an end-to-end computational model of speech perception

Tim Brochier[1], Josef Schlittenlacher[2], Iwan Roberts[1], Tobias Goehring[3], Chen Jiang[1,4], Deborah Vickers[1], Manohar Bance[1]

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Background: Speech perception in CI listeners is affected by the degradation of spectral and temporal information through the CI. Computational models of CI speech perception can be used to rapidly and objectively evaluate strategies that may improve information transmission in CIs. To succeed, these models must make use of similar phonemic cues to CI listeners. Our research aims to replicate phoneme-level CI speech perception patterns using an end-to-end computational model.

Methods: We combined a finite element model of a cochlea, a computational model of the auditory nerve, and an automatic speech recognition neural network (ASR) to generate predictions of CI speech perception. The ASR was trained and tested on neural activation patterns generated by the initial stages of the model, and phonemic information transmission was evaluated. Results were compared to data measured in CI listeners (Donaldson and Kreft, 2006). Consonant features assessed were manner, place, and voicing, and vowel features were the first and second formant, tenseness, and duration. The model was also used to investigate information degradation through the CI signal processing chain.

Results: No significant differences were found between the model and the CI listener data for any consonant or vowel feature. For consonants, manner and voicing cues were transmitted better than place cues. Model predictions and CI listener data for consonant recognition accuracies were correlated ($R = 0.641$, $p = 0.001$), suggesting that the model captures between-consonant differences in perceptibility. For vowels, both the model and CI listeners prioritized the first and second formant cues. The bottleneck of information flow occurred at the electrode-neural interface.

Conclusion: A computational model replicated CI speech perception patterns and quantified information degradation through the CI. The model will help to develop, optimize, and predict the efficacy of new CI processing strategies.

1 sentence summary:
A finite element model of a cochlea, a computational model of the auditory nerve, and an automatic speech recognition neural network were combined to replicate CI speech perception patterns.

P3-3: Hearing-impaired artificial neural networks replicate speech recognition deficits of hearing-impaired humans

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[4] Program in Speech and Hearing Biosciences and Technology, Harvard University, USA

Background: Damage to peripheral auditory structures is known to alter cochlear signal processing. However, relating these changes to the real-world listening difficulties of humans with hearing loss has remained a challenge. Computational models capable of performing real-world tasks could provide insight. Artificial neural networks optimized to perform auditory recognition tasks from simulated cochlear input have recently been shown to replicate aspects of human auditory behavior. Here, we extend this approach to investigate how outer hair cell (OHC) and auditory nerve fiber (ANF) loss can account for difficulties recognizing speech in noisy environments.

Methods: We trained deep neural networks to recognize words from simulated healthy cochlear representations of speech in noise. We then simulated OHC and ANF loss in the cochlear model and measured the effects on network performance. To investigate how plasticity in the central auditory system might allow hearing-impaired listeners to adapt to their damaged cochleae, we also trained networks with impaired cochlear input.

Results: Networks with either OHC or ANF loss introduced at test time replicated behavioral deficits of hearing-impaired listeners: speech recognition performance was degraded (especially at low SNRs) and the fluctuating masker benefit was reduced. In most cases, optimizing networks to handle impaired peripheral input produced remarkably unimpaired performance, provided sounds were presented at an audible level.

Conclusion: The results suggest that a perfectly plastic auditory system could almost fully compensate for hearing loss-related changes in the periphery due to outer hair cell or auditory nerve fiber loss. Our model illustrates how deep learning can provide insight into both normal and abnormal sensory function.

1 sentence summary:
We developed a deep learning model of hearing loss by training artificial neural networks to recognize words in noise from simulated auditory nerve input.

P3-4: Binaural prediction of speech intelligibility based on a blind model using automatic phoneme recognition

Jana Roßbach[1,3], Saskia Röttges[1,2], Christopher F. Hauth[1,2], Thomas Brand[1,2], Bernd T. Meyer[1,3]

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Background: Models for speech intelligibility (SI) prediction are important tools in the development of signal processing algorithms and could be used to estimate the benefit when using hearing aids. We explore the use of using binaural information for modelling SI in reverberant conditions. We propose a model that is blind w.r.t. the separate speech and noise signals, which is in contrast to prior information required by intrusive models.

Methods: The model borrows an algorithm from automatic speech recognition (ASR) and is referred to as BAPSI (for binaural ASR-based prediction of speech intelligibility) and was first introduced in (Roßbach et al. (2021). "Non-intrusive binaural prediction of speech intelligibility based on phoneme classification," in Proc. ICASSP). The model receives a stereo signal (speech in noise) and uses a binaural frontend based on an equalization-cancellation mechanism (Hauth et al., doi:10.1177/2331216520975630). The result is used as input for a deep neural network for phoneme classification. The uncertainty about this classification is used for the prediction. The model is evaluated using data from normal-hearing listeners in three different room conditions (anechoic, office, and cafeteria) and several azimuth angles of the noise. We compare the speech recognition threshold (SRT) of listeners to BAPSI and two intrusive baseline models: the binaural SI model (BSIM06) (Beutelmann and Brand (2006) and HASPI (Kates and Arehart (2014) doi:10.1016/j.specom.2014.06.002) combined with better ear listening (HASPI+BE).

Results: The root mean squared errors (RMSEs) of BAPSI (0.6-2.1 dB) are similar to the RMSEs of BSIM06 (0.3-1.8 dB) and lower than the RMSEs of HASPI+BE (3.1-3.7 dB). Additionally, the correlation coefficients are high (0.71-1.00) for all three models.

Conclusion: The consideration of classifiers based on deep learning seems to be promising for predicting speech intelligibility in binaural acoustic scenes.

1 sentence summary:
In this study, we show that phoneme probabilities from a DNN can produce good estimates of speech intelligibility when combined with a blind binaural processing stage.

P3-5: Use of a deep recurrent neural network to reduce transient noise: Effects on subjective speech intelligibility and comfort

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Background: Hearing aid users often complain about discomfort and reduced speech intelligibility caused by transient sounds such as a knife hitting a plate. Here, a deep recurrent neural network (RNN) for reducing transient sounds was developed and its effects on subjective speech intelligibility and listening comfort were evaluated. The RNN was trained using many sentences spoken with different accents and corrupted by different transient sounds. It was then tested using sentences spoken by unseen speakers and corrupted by unseen transient sounds.

Methods: A paired-comparison procedure was used to compare all possible combinations of three conditions for subjective speech intelligibility and listening comfort for two relative levels of the transients. The conditions were processing using the RNN, processing using a multi-channel transient reduction method (MCTR, [1]), and no processing (NP). Ten native English-speaking participants with normal hearing and ten with mild-to-moderate hearing loss were tested.

Results & Conclusion: For the normal-hearing participants, processing using the RNN was significantly preferred over that for NP for both subjective intelligibility and comfort, processing using the RNN was significantly preferred over that for MCTR for intelligibility, and processing using the MCTR was significantly preferred over that for NP for comfort but only for the higher transient level. For the hearing-impaired subjects, processing using the RNN was significantly preferred over that for NP for subjective intelligibility and comfort, processing using the RNN was significantly preferred over that for MCTR for comfort, and processing using the MCTR was significantly preferred over that for NP for comfort. Overall, the results indicate that the RNN was more effective than the MCTR.

[1] Keshavarzi, M., Baer, T. & Moore, B. C. J. (2018) Evaluation of a multi-channel algorithm for reducing transient sounds. Int J Audiol, 57, 624-631.

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1 sentence summary:
Transient noise reduction using a deep recurrent neural network improves the subjective speech Intelligibility and comfort.

Session P4:
Interventions and diagnosis of tinnitus

Session Chair: Dr Marina Salorio-Corbetto, University of Cambridge, UK

P4-1: Outcomes and experiences of delivering an internet-based intervention for tinnitus during the COVID-19 pandemic

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Background: Internet-based cognitive behavioral therapy (ICBT) for tinnitus has the potential of improving accessibility to tinnitus care. As the efficacy of ICBT in the US is unknown as study was planned. As the COVID-19 pandemic unfolded, this study was ready to be launched. Together with identifying the intervention effects, insights where obtained regarding delivering ICBT during the COVID-19 pandemic.

Methods: A randomized controlled trial including 158 participants using a delayed treatment design was undertaken. The primary outcome was a change in tinnitus distress as measured by the Tinnitus Functional Index (TFI). Secondary outcome measures included measures of anxiety, depression, insomnia, tinnitus cognitions, hearing-related difficulties, and health-related quality of life. Treatment engagement variables included monitoring engagement regarding the number of logins, number of modules opened and number of messages sent.

Results: Undertaking the ICBT led to large improvements of tinnitus severity which were maintained at follow up. There was furthermore a greater reduction for secondary effects such as insomnia and tinnitus cognitions. Results were maintained 2 months post-intervention. The intervention was very beneficial to help individuals cope, and helped them better manage the anxiety associated with the pandemic.

Conclusion: ICBT supported tinnitus individuals during the pandemic. Although results are comparable to results found using ICBT in Europe, participant engagement was much lower. For some participants, the pandemic or having COVID-19 impacted on their ability to focus on the trial. Ways of improving intervention compliance and engagement are required as well as evaluating the long-term effects that both the intervention and COVID-19 has had requires monitoring. Creative solutions are required to improve accessibility of ICBT due to its proved efficacy.

1 sentence summary:
A RCT was undertaken on a US populations indicating the value of an CBT internet-intervention for both reducing tinnitus distress, tinnitus comorbidities and managing the anxiety associated with the pandemic.

P4-2: A data-driven decision tree for diagnosing somatosensory tinnitus

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Background: The influence of the somatosensory system on the perception of tinnitus is well-documented. International experts have defined somatosensory tinnitus (ST) as a tinnitus influenced by the cervical or temporomandibular somatosensory system, and have recently agreed on a set of diagnostic criteria. However, a straightforward diagnosis of ST has proven elusive. Here, we present an uncomplicated model for diagnosing ST, based on the results of a large-scale online survey.

Methods: A survey was launched on the online forum Tinnitus Talk, managed by Tinnitus Hub, in September 2019. The survey included 42 questions, both on the presence of diagnostic criteria for somatosensory tinnitus and on other potentially influencing factors. Individuals were identified as having ST if they reported a known diagnosis of ST and/or experienced considerable influence from cervical spine or temporomandibular problems on their tinnitus. A decision tree was constructed to classify participants with and without ST using the rpart package in R. Tree depth was optimized during a five-fold cross-validation. Finally, model performance was evaluated on a subset containing 20% of the original dataset.

Results: In total, 7981 participants filled out the questionnaire completely, of whom 681 individuals (8.5%) were identified as having ST. The final model classified participants with an accuracy of 82.24% (82.54% sensitivity, 79.02% specificity). This decision tree was based on only four parameters: simultaneous improving or worsening of tinnitus and neck or jaw complaints, tension in the suboccipital muscles, the ability to modulate the tinnitus sound with certain movements, and the presence or absence of bruxism.

Conclusions: Based on a large dataset, we developed a straightforward tool to diagnose ST with over 80% accuracy. The final decision tree can be readily used in the clinic, and may improve diagnostic flow and expedite the start of appropriate treatment for somatosensory tinnitus.

1 sentence summary:
Based on the results of an online survey, we developed a decision tree to classify somatosensory tinnitus patients with an accuracy of over 80%.

P4-3: What can we learn about tinnitus from social media posts?

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Background: Individuals with tinnitus are highly heterogeneous in terms of etiology, manifestation of symptoms, and coping's mechanisms. Most of these patients are likely to seek hearing health information and social support online via various websites or social media platforms. Indeed, information is easily accessible online. In the absence of evidence-based care, patients with similar symptoms regroup, share experiences, and exchange tips. The present study examines such discussions around tinnitus in Reddit free-texts posts.

Methods: The study uses a cross-sectional design. 130,000 posts were extracted from Reddit's application programming interface. The 101,000 unique posts were analyzed using automated NLP techniques: hierarchical cluster analysis; unsupervised Machine Learning -- Latent Dirichlet Allocation (LDA) -- algorithm; and supervised sentiment analysis. After identifying the main topics in the corpus and assessing their sentiment scores, sub-themes are probed in selected topics with LDAs. Reddit users' interactions are used to measure topics' cooccurrence in messages and threads.

Results: The cluster and LDA analyses both result in a 16-topic solution with comparable clusters. The sentiment analysis shows heterogeneity among different factors. LDAs ran, e.g., on the Temporo-Mandibular Joint (TMJ) topic, highlights three distinctive sub-themes: teeth and jars; neck and back stress; and muscular and neuralgic pains. Finally, themes overlap emerges from their cooccurrence as suggested by the users' activity. For instance, the TMJ topic appeared to be often discussed along with the Music Volume topic.

Conclusion: The study maps topics discussed on social media, some not explored in the literature (e.g., supplements, personal timeline). It also investigates topics interconnection in spontaneous discussions. The findings enrich the understanding of patients' interplay with their conditions and inform the development of appropriate patient-centered strategies to support individuals with tinnitus.

1 sentence summary:
Exploiting spontaneous messages of Reddit users discussing tinnitus, this work identifies the main topics of interest, their heterogeneity, and how they relate to one another based on cooccurrence in users' discussions; to enhance patient-centered support.

P4-4: Behavioral and electrophysiological evaluation of loudness growth in clinically normal hearing tinnitus patients with and without hyperacusis

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Background: Tinnitus and hyperacusis are widely assumed to be associated with the central gain mechanism, and this mechanism also has control over loudness perception. Although central gain increases with the attempts to compensate hearing loss, reduced input can also be observed in those with clinically normal hearing. This study aimed to evaluate the loudness growth function of tinnitus patients with and without hyperacusis using behavioral and electrophysiological methods.

Methods: The study consists of 3 groups with a total of 60 clinically normal hearing subjects, including the control group (10 men and 10 women; mean age, 39.8; SD, 11.8 years), tinnitus group (10 men and 10 women; mean age, 40.9; SD, 12.2 years) and hyperacusis group (also have tinnitus) (7 men and 13 women; mean age, 38.7; SD, 14.6 years). Loudness discomfort levels (LDLs), categorical loudness scaling (CLS), and auditory cortical evoked potentials (ACEP) were used for the evaluation of loudness growth. N1-P2 component amplitudes and latencies were measured.

Results: LDL results of 500, 1000, 2000, 4000, and 8000 Hz showed a significant difference between the hyperacusis group and the other two groups ($p < 0.001$). In the loudness scale test performed with 500 Hz and 2000 Hz narrow-band noise (NBN) stimulus, a significant difference was observed between the hyperacusis group and the other two groups in the “medium,” “loud,” and “very loud,” categories ($p < 0.001$). In the cortical examination performed with 500 Hz and 2000 Hz NBN stimulus at 40, 60, 80 dB nHL intensities, no significant difference was observed between the groups in the N1, P2 latency and N1-P2 peak-to-peak amplitude.

Conclusion: The hyperacusis group is significantly different between the groups in behavioral tests, but not in electrophysiological tests. In our attempt to determine hyperacusis with objective tests, N1 and P2 response was not seen as a suitable method. However, CLS can also be used in addition to LDLs used in behavioral tests.

1 sentence summary:
The hyperacusis group differs significantly from the control and tinnitus groups in behavioral tests, but not in electrophysiological tests.

P4-5: Systematic monitoring of Meniere's disease: A smartphone-based approach for the periodical assessment of audiometric measures and fluctuating symptoms

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Background: Meniere's disease is a progressive disorder of the inner ear, characterized by spontaneous vertigo, fluctuating hearing loss and tinnitus. Pure tone audiometry is the principal tool for following up the disease progression and treatment. Unfortunately, lack of sufficient data emerges from this measurement. Therefore, it is crucial to seek for reliable self-testing methods of audiometry that overcome the barrier of audiometric device calibration. In this study, a software application for mobile devices is proposed, in an attempt to record the course of patients with Meniere's disease or fluctuating hearing loss.

Method: Binaural loudness and pitch matches were measured by presenting sinusoid and amplitude-modulated noise stimuli, using an originally developed software application for smartphone devices. Subjects reported the potentially infected ear in the beginning of the process. Participants conducted the provided tests every day for 14 days via their smartphones using typical headphones in a quiet environment. The procedure was automatically controlled by Randomized Maximum Likelihood sequential procedure.

Results: Subjects with Meniere's disease required significantly higher intensities to obtain equal loudness than normal-hearing participants ($p<0.05$). Binaural pitch matches of the patient group did not differ remarkably from the normal, but showed larger daily fluctuations at tests of low- frequency stimuli (median>6%, IQR>7%). Perceptual pitch ability is better in sine-measurements comparing to those with SAM noise for both groups. In general, SAM noise results were similar between the two groups ($p=0.88$).

Conclusions: The proposed application may constitute the beginning of collecting sufficient and varied data on the peculiar Meniere's disease, absent until now. Frequent and long-term recordings are expected to contribute to the timely diagnosis, delay and personalized treatment of the disease.

1 sentence summary:
This study developed an application software that periodically records the course of Meniere's disease patients through the performance of automated, binaural audiometric tests.

Session P5:
Auditory attention and processes

Session Chair: Dr Alexander J Billig, Ear Institute, UCL, UK

P5-1: Using active inference to model selective attention during cocktail party listening

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Background: Selective attention enables listeners to focus on a talker among a mixture of talkers. Yet, it does not appear to be all-or-none: when an attention-directing cue is presented, preparatory attention builds up over time. For example, reaction times progressively improve as an instructional cue is presented longer in advance of the target talker. Also, EEG activity increases in amplitude before the target talker speaks. The computational processes underlying this slow induction of attentional set are not fully understood.

Methods: Here, we take a theoretical stance, based on active inference (Friston et al., 2017). We introduce a new generative model of selective attention during cocktail party listening, and treat selective attention as an inference problem. We model a simple paradigm in which two spatially-separated talkers each speak a different colour and number word. A visual cue directs attention to the left or right talker, and the task is to identify the words spoken by the cued talker. We used this model to test competing hypotheses about how time-sensitive changes in precision affect simulated reaction times and EEG responses.

Results: Temporal changes in precision were unnecessary to explain the improvement in reaction times with longer cue-target intervals, but were needed to explain the increase in EEG responses before the target talker speaks. An exponential-shaped increase in precision fit the EEG data better than alternative functional forms tested.

Conclusion: Overall, this work contributes to our understanding of selective attention. The model generates quantitative (testable) predictions about behavioural, psychophysical and electrophysiological responses, and underlying changes in synaptic efficacy.

1 sentence summary:
We introduce a new generative model of selective attention during cocktail party listening, and treat selective attention as an inference problem.

P5-2: Cortical tracking of a distractor speaker modulates the comprehension of a target speaker

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Background: To comprehend speech in busy environments such as pubs and restaurants, human listeners need to selectively attend a target voice while ignoring interfering voices. Such speech-in-noise processing has recently been found to involve vortical tracking of the target speaker. In particular, altering this cortical entrainment through transcranial alternating current stimulation (tACS) in the theta band has been found to modulate the comprehension of speech in noise. However, the functional role the cortical tracking of an ignored speech signal and its impacts on the behaviour remain unclear. Here we therefore sought to investigate the impact of tACS with the envelope of a distractor voice on the comprehension of a target speech signal.

Methods: Eighteen right-handed native English speakers with self-reported normal hearing and no history of mental health problems or neurological disorders took part in the experiment. The individual SNR was determined such that the participant understood 50% of a target sentence in the presence of a distractor voice. We then employed tACS with either the target or the distractor envelope filtered in theta band, while evaluating the speech comprehension at the individual SNR. As a control condition, we further evaluated the speech comprehension when a sham stimulus was employed as a current waveform.

Results & Conclusion: We found that tACS with the distractor envelope influenced comprehension of the target speaker, suggesting that the cortical tracking of the ignored speaker plays a functional role in speech processing. tACS with the target envelope also influenced speech comprehension in a very similar manner. Particularly, both types of neurostimulation caused a significant modulation of speech comprehension that varied sinusoidally, at the longest possible period, with the applied phase shifts. Moreover, both types led to the modulation of speech comprehension that had a significantly consistent phase dependency across different participants.

1 sentence summary:
Cortical tracking of the ignored speaker plays a functional role in speech processing.

P5-3: Correlates of linguistic processing in the frequency following response to naturalistic speech

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Background: Comprehension of spoken language requires rapid and continuous integration of upcoming acoustic information. Most of the studies investigating neural correlates of natural language comprehension focus on comparatively slow cortical activity. However, fast neural activity in subcortical and cortical areas can also track the fundamental frequency of voiced speech (f0). Whether this fast neural tracking plays a role in linguistic aspects of speech processing remains unclear. Here, we investigated whether this neural response is influenced by linguistic cues.

Methods: We measured EEG while participants listened to audiobooks. We then used a language model to compute linguistic features describing each word from the stories. Each word was characterized by its frequency out of context and by context-dependent surprisal and precision. We used a linear model to find a mapping between the fundamental waveform, which oscillated at the f0 of the speech signal, and the EEG. The model quantified the neural tracking of the fundamental waveform through a reconstruction score. Finally, we established a multiple regression model that predicted the reconstruction score for each word from its linguistic features.

Results: The neural response estimated by the linear model had a low latency (11 ms) and a high-frequency (above 50 Hz), characteristic for the neural tracking of the f0. The coefficients of the multiple regression model indicated that the single-word neural phase-locking to the f0 was significantly influenced by the context-dependent linguistic features: word precision and surprisal.

Conclusion: We showed that the neural response to the f0 of continuous speech in naturalistic narratives is modulated by context-dependent linguistic cues. Due to the low latency of the response, our findings suggest that it is under top-down control from higher processing centers. Our results show that the neural response at the f0 plays an active role in the rapid and continuous processing of spoken language.

1 sentence summary:
Our findings suggest that the frequency following response tracking the fundamental frequency of voiced speech plays an active role in the rapid and continuous processing of spoken language.

P5-4: The effect of selective loss of auditory nerve fibers on temporal envelope processing: a simulation study

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Background: Cochlear synaptopathy is a selective loss of auditory nerve fibers with low spontaneous rates (SRs) due to sublimit noise exposure or aging and has been proposed to affect temporal envelope (TE) processing at suprathreshold levels. However, due to insufficiently sensitive measures and highly variable patients' background, evidence of cochlear synaptopathy in humans has been unconvincing. This study examines the role of low SR fibers in suprathreshold TE processing and the choice of task parameters to manifest cochlear synaptopathy through computational simulation.

Methods: Selective deafferentation of auditory nerves was simulated in a physiologically inspired model of the auditory system. The model output was then transformed into sound and tested on normal-hearing listeners. Loss of low SR fibers was compared to loss of both low and medium SR fibers, loss of high SR fibers, or no fiber loss. TE processing tasks included amplitude modulation detection (500-Hz carrier, modulation rates of 16, 32 and 64 Hz), speech recognition in modulated noise, and recognition of unvoiced speech in modulated noise.

Results: The performances of the tasks were consistently affected by loss of low SR fibers and loss of low and medium SR fibers, but not by loss of high SR fibers. For amplitude modulation detection, the performance was degraded more severely when the modulation rate was 16 Hz than 64 Hz. For speech tasks, the intelligibility of unvoiced speech was impaired more severely than natural speech.

Conclusion: The simulation supports the role of low SR fibers in coding suprathreshold TE and shows that a sensitive TE measure requires careful selection of the task parameters. The current study recommends 16 Hz over 64 Hz for amplitude modulation detection and unvoiced speech over natural speech for testing cochlear synaptopathy.

1 sentence summary:
Computer simulation supports the hypothesis of cochlear synaptopathy selectively damaging low-spontaneous rate auditory nerves and impacting temporal envelope processing, and shows that some tasks are more sensitive to this issue than others.

P5-5: Functional hearing and communication deficits (FHCD) in blast-exposed service members with normal to near-normal hearing thresholds

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Background: Military audiologists routinely see Service members (SMs) with clinically normal-hearing thresholds (NHT) who present with hearing difficulties not consistent with their audiogram. Many of these SMs also report being exposed to one or more blasts during deployments. The purpose of this study was to investigate sensory and cognitive factors that might lead to such functional hearing deficits, and to provide insight into potential rehabilitative strategies.

Methods: Two groups of blast-exposed SMs – those with or without suspected FHCD – were evaluated. FHCD was identified using a rapid screener consisting of two auditory tests and a 6-question survey. Once enrolled, subjects were further evaluated using a battery of behavioral, electrophysiological, and cognitive tests. These tests were selected to provide insight into the likely factors contributing to FHCD. Results for blast-exposed SMs were compared with an age-matched control group of SMs with audiometric thresholds ≤ 20 dB HL, no history of blast-exposure, and no evidence of FHCD.

Results: A 3-way comparison (i.e., age-matched controls, and blast-exposed SMs with or without FHCD) showed that SMs with FHCD performed significantly worse than control subjects on behavioral and electrophysiological metrics sensitive to sensory deficits in auditory processing. There was also evidence of some cognitive deficits, especially on tests of speed of language-processing. By comparison, SMs without FHCD performed more like the control group.

Conclusions: Blast-exposed SMs with clinically normal audiograms and evidence of FHCD appear to have poorer peripheral auditory processing and a poorer internal signal-to-noise ratio when compared with control subjects or with blast-exposed SMs without FHCD. Results suggest that blast-exposed SMs with NHT and FHCD are likely to have sensory processing deficits, findings consistent with the growing practice to prescribe low-gain hearing aids for this population.

1 sentence summary:
Difficulties integrating binaural cues and understanding speech in noise amongst blast-exposed Service members with audiometric thresholds within clinical norms are most likely due to sensory, and not cognitive deficits, as indicated by a poorer signal-to-noise ratio in the neural encoding of sound in the peripheral auditory system.

P5-6: Visualization of speech perception errors through phoneme alignment

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Background: While clinical speech tests assess the ability of people with hearing loss to hear with a hearing aid or cochlear implant, these tests usually examine speech perception at the word or sentence level. Because few tests analyze perceptual errors at the phoneme level, there is a need for an automated program to compute and visualize the accuracy of phonemes in response to speech tests.

Method: Stimulus-response pairs are read in by the program to obtain phonetic representations from a digital pronouncing dictionary. Global alignment between response and stimulus phonemes is achieved through the use of a Levenshtein Minimum Edit Distance algorithm with costs for insertion, omission and substitutions. Accuracies for each phoneme are based on a modified F-score, which are then averaged and visualized with respect to place and manner (consonants) or height (vowels). Confusion matrices of stimulus-response phoneme pairs undergo information transfer analysis based on ten prescribed phonological features, from which a histogram of the relative information transfer for the features is shown as a phonemegram.

Results: The program was applied to one dataset, where stimulus-response sentence pairs from 6 volunteers (each with varying degrees of hearing loss) were analyzed. 4 volunteers listened to sentences from a mobile auditory training app while 2 listened to sentences from a clinical speech test. Stimulus-response word pairs from 3 word lists were also analyzed. In all cases, visualization of phoneme accuracy was obtained in real-time.

Conclusion: It is possible to automate the alignment of phonemes from stimulus-response pairs from speech tests in real-time, which then makes it easier to visualize the accuracy of responses via phonetic features. Such visualization of phoneme alignment and accuracy could aid speech language pathologists and audiologists either in person or virtually.

1 sentence summary:
We present an automated program for aligning stimulus-response phonemes collected from speech testing in order to visualize speech perception errors in individuals with hearing loss

Session P6:
Computational auditory modelling

Session Chair: Dr Gerard Encina-Llanas, DTU, Denmark

P6-1: "Ear in the Clouds"– A web app supporting computational models for auditory-nerve and midbrain responses

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Background: Complex sounds are encoded in population responses that support discrimination, identification, and detection. Models that simulate neural responses are widely available; however, they often require access to and familiarity with computer programming environments, such as MATLAB or Python. Additionally, models often focus on single neurons - users interested in population coding must extend the models to visualize responses of tonotopic arrays of neurons. We developed a cloud-based web app to make simulation and visualization of population responses more accessible.

Methods: The MATLAB-based App Designer was used to implement our graphical user interface (UR_EAR, University of Rochester, Envisioning Auditory Responses). The interface displays responses of auditory-nerve (AN) (Zilany et al., 2014, and Bruce et al., 2018) and midbrain models (Mao et al., 2013; Nelson & Carney, 2004; Carney & McDonough., 2018). Midbrain models include two major types of inferior colliculus (IC) neurons, with band-enhanced and band-suppressed modulation transfer functions. Stimuli include user-uploaded audio files or standard psychophysical stimuli with adjustable parameters. Time-varying rate functions for a selectable range of frequency channels are displayed, alongside average-rate responses computed over an adjustable analysis window.

Results: The web app is at https://urhear.urmc.rochester.edu, with links to a User Manual, FAQs, and a contact button for the authors. Open source code is available at https://osf.io/6bsnt/, including executable versions for Win10, Mac, and Linux.

Conclusions: The current Web App supports visualizations of AN and IC model population responses for several standard psychophysical stimuli, as well as responses to user-provided audio files. Future efforts will be focus on efficient visualization of responses to longer stimuli (music, speech) and apps that estimate psychophysical thresholds based on model responses. (NIH-R01-001641)

1 sentence summary:
A cloud-based web app provides an accessible tool for simulation and visualization of population responses of model auditory-nerve and midbrain neurons.

P6-2: Predicting fusion of dichotic vowels in normal hearing listeners with a physiologically-based model

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Background: Fundamental frequency (F0) is an important cue for speech segregation. Listeners with normal hearing are more likely to correctly identify both vowels of a dichotically-presented pair if the F0s of the two vowels are different; however, they overwhelmingly identify two vowels with the same F0s as a single vowel (i.e., fuse the two vowels) (Reiss & Molis, 2021). A physiologically based model was applied to explore the underlying neural mechanism.

Method: Six subjects with normal hearing participated. First, to obtain individualized vowel identification maps, subjects identified 90 single synthetic vowels with first and second formant values varied evenly across the vowel space. Next, subjects listened to dichotic vowel pairs (selected from four exemplars), and identified the vowel(s) heard. Vowel identification was modeled with a phenomenological auditory nerve (AN) model (Zilany et al., 2014), a relay cochlear nucleus model, and a same-frequency-inhibition-excitation inferior colliculus (IC) model (Carney et al., 2015). AN model output was used to determine whether the F0s of the dichotic vowel pair were the same or different. For the same F0s, the IC responses of the two vowels were averaged to simulate a fused vowel percept and predict a single vowel response. Otherwise, the IC responses of the two vowels were used to predict two separate vowel responses. Template responses were obtained for the 90 vowels used for the vowel identification map. Predicted vowel identification was based on the similarity between those templates and model output (e.g., compare Fig. 1D to Fig. 1A-C).

Result: Consistent with previous studies, subjects often fused dichotic vowel pairs with the same F0, but not different F0s. The model predictions for fused vowels were similar to those of human subjects, especially for low F0s.

Conclusion: A physiologically-based model utilizing binaural averaging can simulate some dichotic vowel fusion percepts in human listeners.

1 sentence summary:
A psysiologically-based model can simulate some dichotic vowel fusion percepts in human listeners.

P6-3: A computational single-fiber model of electric-acoustic stimulation

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Background: Cochlear implant users with residual acoustic hearing in the implanted ear strongly benefit from combined electric-acoustic stimulation (EAS). However, EAS also introduces interferences between the two modalities that are not fully understood. The goal of this project is to provide a computational framework for the investigation of peripheral electric-acoustic interaction to allow for deeper insights into the underlying physiological mechanisms.

Methods: A computational model of a single auditory nerve fiber (ANF) excited by EAS was developed to study the interaction between electric and acoustic stimulation. Technically, two existing models of sole electric or acoustic stimulation were coupled to simulate responses to combined EAS. The model of acoustic stimulation is a phenomenological model of the auditory periphery (Bruce et al., 2018). The model of electric stimulation simulates direct electroneural stimulation of the ANF (Joshi et al., 2017). Different coupling methodologies between both models were investigated.

Results: The model was validated with single-ANF recordings from animal experiments. Simulated measures included threshold and dynamic range, spike rate, latency, jitter, and vector strength. The model reproduces the reported spike statistics including effects such as the lowering of electrical thresholds and dynamic ranges in deaf ANFs, or the reduction of phase locking by a second stimulus of the other modality. The refractoriness of the model leads to an inhibitory interaction between electrically and acoustically evoked spiking in accordance with published data.

Conclusion: We presented a model of an ANF responding to EAS that largely reproduces animal data. The presented model forms a basis for future investigations of EAS interaction.

Acknowledgements
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1 sentence summary:
We present a computational model of auditory nerve fiber responses elicited by combined electric and acoustic stimulation which can be used to investigate peripheral electric-acoustic interaction.

P6-4: A computational model of fast spectrotemporal chirp sensitivity in the inferior colliculus

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Background: Recent physiological results have demonstrated that inferior colliculus (IC) neurons in rabbits and gerbils can be sensitive to fast frequency chirps. This sensitivity was shown using harmonic stimuli with chirps within each pitch period caused by the phase spectrum. Chirp sensitivity is not present in state-of-the-art computational models of the IC. We evaluate several models hypothesized to replicate these physiological results. While mechanisms for frequency-modulation (FM) sensitivity have been proposed in previous IC studies, those studies focused on FM specialists, such as bats, rather than generalist mammals. This work has implications for speech coding because fast chirps are contained in vowels. Here, we present a novel computational model of the IC that accurately portrays the influence of chirp sensitivity on responses to common stimuli, while also retaining known IC properties such as amplitude-modulation tuning.

Methods: We use the auditory-nerve model from Zilany et al. (2014) and build upon the same-frequency inhibition and excitation (SFIE) model for IC neurons (Nelson and Carney, 2004). The proposed models are evaluated based on their ability to reproduce physiological responses to chirp stimuli.

Results: The proposed models of IC chirp sensitivity are based on the premise that a neuron selective for chirp direction or velocity receives input from multiple frequency channels. A model that incorporates coincidence detection and delayed inhibition has the best mix of chirp selectivity and preservation of known IC properties. We can vary model configurations, as well as the relative strength, timing, coincidence window size, and duration of inputs.

Conclusion: We tested several models for the neural mechanism of fast chirp sensitivity in the IC. These models allow us to test hypotheses related to chirp sensitivity. This work contributes new insight into the neural representation of speech at the level of the midbrain. (NIDCD 001641)

1 sentence summary:
We compare several computational models of the inferior colliculus in their ability to accurately simulate physiological selectivity to fast frequency chirps.

P6-5: Modeling the effects of inhibition and gap junctions on synchrony enhancement in bushy cells of the ventral cochlear nucleus

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Background: Auditory nerve fibers (ANFs) tend to synchronize to low-frequency stimuli, and this synchrony is increased in bushy cells of the ventral cochlear nucleus (VCN) (Joris & Smith 2008). Synchrony enhancement in globular bushy cells (GBCs), receiving many ANF synaptic inputs, can be explained by a coincidence-detection mechanism. However, the possible mechanisms behind spherical bushy cell (SBC) synchrony enhancement remain unclear, since they receive very few excitatory inputs. Gomez-Nieto & Rubio (2011) found that bushy cells are also connected to each other via gap junctions (i.e., electrical synapses), which are known to influence synchrony in other neural systems.

Methods: We have developed biophysically detailed models of GBC and SBC microcircuits in the VCN based on Manis & Campagnola (2018), with inputs provided by the ANF model of Bruce et al (2018). The effects of broadband and narrowband inhibition, coming from D-stellate (DS) and tuberculoventral (TV) cells respectively, on synchronization enhancement are investigated, as well as the effects of gap junction conductance strength between a pair of bushy cells.

Results: Inhibition from model DS and TV cells appears to fill the gaps between peaks in firing of model ANFs, which tends to eliminate some of the spontaneous firing in SBCs caused by ANFs. This enhances synchronization in model SBCs but not GBCs. Similarly, gap junctions increase synchronization for a pair of model SBCs but not for a pair of model GBCs.

Conclusion: The results with a pair of model bushy cells connected via gap junctions suggest that both the inhibition and the gap junctions affect synchrony enhancement for SBCs but not GBCs, indicating that coincidence detection fully explains synchrony enhancement in GBCs. In ongoing work, we are increasing the number of model SBCs connected with gap junctions and are exploring effects on synchrony of different cluster structures and of the gap junction strength between these clusters.

1 sentence summary:
This study explores the effects of inhibition and gap junctions on the synchrony enhancement seen in ventral cochlear nucleus bushy cells by using biophysically detailed neural network models of bushy cells microcircuits.

P6-6: Modeling formant-frequency discrimination based on auditory-nerve and midbrain responses: normal hearing and sensorineural hearing loss

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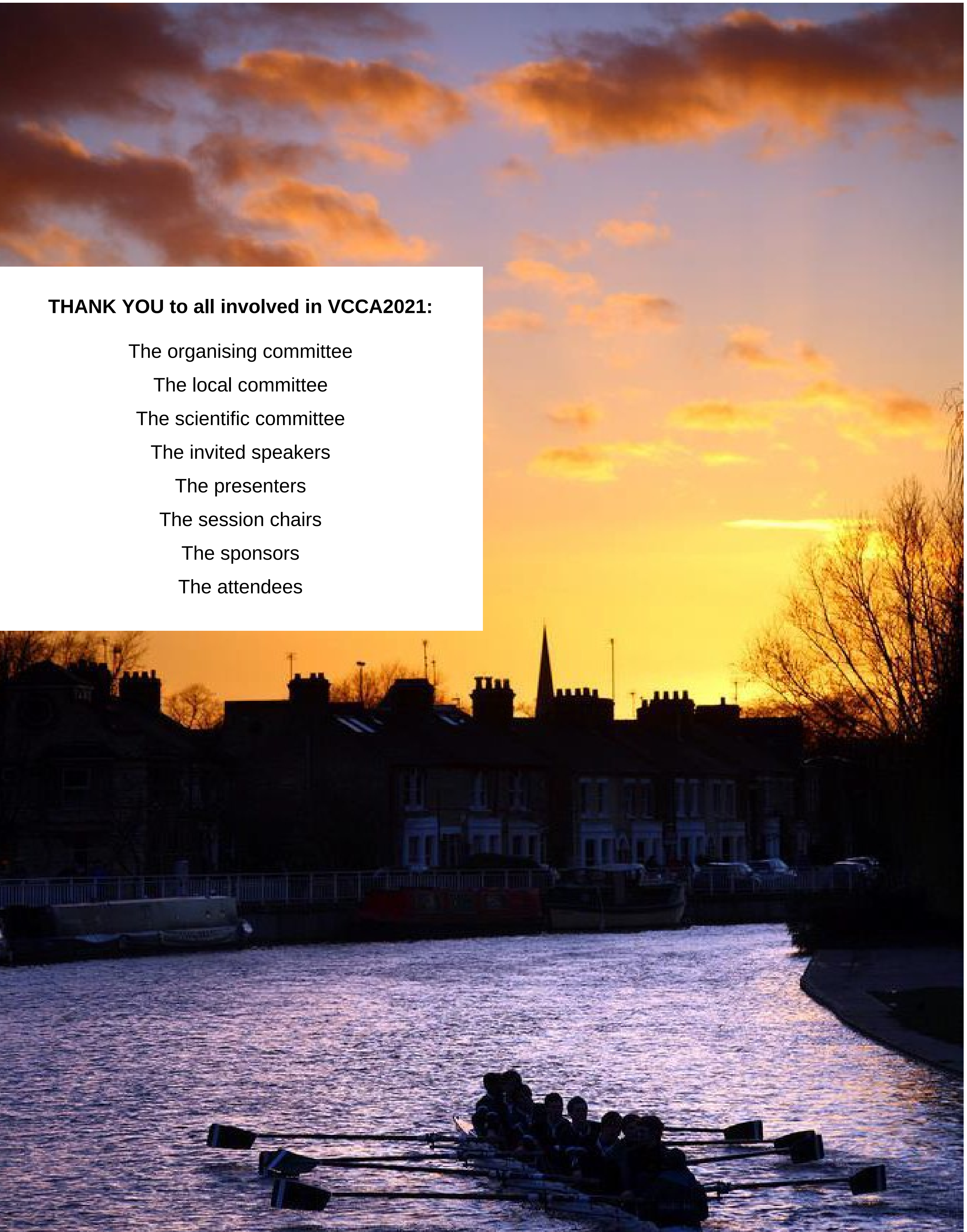
Background: Formant-frequency difference limens (FFDLs) are increased in listeners with sensorineural hearing loss (SNHL) when compared to normal hearing (NH) controls. Preliminary results show that thresholds of the SNHL group is more affected by broadening the formant bandwidths (BW). We tested the hypothesis that response profiles of neurons in the inferior colliculus (IC), based on the neural fluctuation model (Carney, 2018), could predict FFDLs in listeners with NH and SNHL.

Methods: Behavioral thresholds were estimated in a two-down-one-up, four-interval, two-alternative-forced-choice paradigm. Computational models for auditory-nerve (AN, Zilany et al., 2014) and IC neurons (Mao et al., 2013) were used to simulate response profiles across neural populations (characteristic frequency=150-3000 Hz). FFDLs were estimated based on the Mahalanobis distances between model response profiles to each stimulus interval and a template based on the standard stimulus. Noise in the simulations was based on spontaneous activity of AN fibers. Model thresholds were compared to the behavioral results.

Results: FFDLs were estimated for seven listeners by including pure-tone thresholds in the peripheral model. Trends across BW and different degrees of SNHL agree with behavioral results. Threshold estimates based on AN population responses were generally higher than behavioral thresholds, while IC-based thresholds were lower. Thus IC-rate profiles can potentially explain human thresholds for this task.

Conclusion: The formant-frequency discrimination task is an alternative to intelligibility testing for quantifying sensitivity to changes in speech sounds in listeners with SNHL. Thresholds for all listeners increased with BW, as predicted by computational models of AN and IC rate profiles. IC responses are driven by neural fluctuations; therefore, results suggest strategies for enhancing intelligibility by manipulating neural fluctuations. (NIH-R01-DC001641)

1 sentence summary:
Formant-frequency difference limens in human with normal hearing or mild sensorineural hearing loss were estimated based on models for neural fluctuation profiles of neurons in the inferior colliculus.



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