IHCON 2022

International Hearing Aid Seminar (IHAS)
August 9 – 10, 2022

International Hearing Aid Research Conference (IHCON)
August 10 – 14, 2022

Granlibakken Conference Center
Tahoe City, California

In memory of Sigfrid D. Soli, PhD
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Conference Support: Kevin O’Connor and Barbara Serrano
# Student Scholarship Recipients

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<th>Name</th>
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<td>Fotios Drakopoulos</td>
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<td>Caitlin Frisby</td>
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<td>Chi Yhun Lo</td>
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<td>Hasan Saleh</td>
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<td>Lipika Sarangi</td>
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<td>Dana Urbanski</td>
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<td>Soumya Venkitakrishnan</td>
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<td>Kathryn Wiseman</td>
<td>Boys Town National Research Hospital</td>
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IHAS PROGRAM

Monday, August 8

4:00 p.m. – 10:00 p.m.  IHAS Arrival / Registration
5:30 p.m. – 6:30 p.m.  IHAS Welcome Reception
6:30 p.m. – 8:00 p.m.  Dinner
8:00 p.m. – 10:00 p.m.  Social

Tuesday, August 9

7:00 a.m. – 8:30 a.m.  Breakfast
8:30 a.m. – 9:50 a.m.

IHAS ORAL SESSION 1

S1-1. Cognition and On-Beat Rhythmic Ability Support Speech-In-Noise Perception for Older Adult Hearing Aid Users

Chi Yhun Lo*, Ella Dubinsky1, Gurjit Singh2, Frank Russo1

1Toronto Metropolitan University, 2Toronto Metropolitan University, Phonak Canada, University of Toronto

Musicians tend to perform better than non-musicians on a range of auditory tasks, such as speech-in-noise (SIN) perception. However, the relative contributions to this advantage (i.e., musicians’ innate abilities and/or learned experience) remains contentious. Associations between working memory, frequency discrimination, and SIN performance are relatively well explored and understood, while the role of rhythmic abilities has been investigated to a much lesser extent—particularly in populations with hearing loss. Nonetheless, multiple studies suggest that the motor system and rhythmic abilities are correlated with better SIN abilities. This may have implications for individuals with hearing loss, who are typically characterised by poor frequency discrimination (i.e., pitch and timbre representations) but intact rhythmic abilities.

The present study is part of a larger study investigating the benefits of choir-based training and active music listening for adults with hearing aids. The present dataset explored pre-training correlations between outcome measures that were broadly stratified as speech perception, cognition, and psychoacoustic/musical tests. The aim of this study was to investigate the relative contributions of psychoacoustic outcomes on speech perception.
Participants were 38 older adult hearing aid users with a moderate bilateral hearing loss (M = 46.7 dB 4FAHL) aged between 57 and 90 years (M = 72.8 years, 25 female and 13 male). All participants passed the Montreal Cognitive Assessment (MoCA) screening for mild cognitive impairment and Alzheimer’s disease.

As expected, MoCA and SIN scores were moderately correlated, r(36) = -.49, p < .01, with better MoCA scores associated with improved speech reception thresholds. Interestingly, better on-beat rhythm perception was also moderately correlated with better SIN performance, r(36) = -.45, p < .01. To the authors’ best knowledge, the association between on-beat rhythm accuracy and SIN perception has not been previously reported for adults with hearing aids. Frequency- and spectral-psychoacoustic measures were not correlated with SIN.

In conclusion, cognition and on-beat rhythmic ability are associated with better speech-in-noise outcomes for older adults with a moderate hearing loss who use hearing aids. [This research was funded by a Mitacs training grant awarded to the second author that was partially sponsored by Unitron Canada.]

S1-2. Co-Production of Text Message Content to Help NHS Audiology Patients when They Are First Prescribed Hearing Aids

Emma Broome*, Katrina Copping†, Sian Calvert†, Helen Henshaw†
†NIHR Nottingham Biomedical Research Centre, Hearing Sciences, University of Nottingham, UK

Introduction: In the United Kingdom, approximately 12 million people have permanent hearing loss, and 355,000 adults are fitted with hearing aids each year via the National Health Service (NHS). However, the non-use and infrequent use of NHS-prescribed hearing aids is high (18-31%). There can be many barriers to overcome when hearing aids are first prescribed and patients tell us that they would benefit from additional support.

The NHS-approved text-message service ‘Florence’ has been shown to help patients self-manage many different long-term conditions. However, it has not yet been used by patients with hearing loss. We are working in partnership with patients to coproduce a Florence intervention protocol for new hearing aid users, using qualitative participatory techniques. The intervention protocol, designed in accordance with the Medical Research Council guidance for the development and evaluation of complex interventions, will use health behaviour theory to improve patients’ capability, opportunity and motivation, when they are first prescribed hearing aid(s). Florence will deliver text-messages to individuals’ mobile telephones, with the aim of improving hearing aid use and benefit. Text-message content, drawing upon established behaviour change techniques, will be coproduced with patients in order to support the adaptation to and use of new hearing aids by addressing key barriers.

Objective: This study aims to create and refine theory-driven text-message content for a Florence intervention protocol.

Design: This work-in-progress qualitative study comprises a two-stage process to intervention development:

1. Maximum variation sampling will be used to recruit people with hearing loss and their communication partners for three workshops. Participants will be invited to coproduce and refine text-message content using PhotoVoice, a qualitative participatory method that documents and reflects reality. A further workshop will be held with audiologists to ensure that intervention content is aligned to clinical care.

2. To assess the acceptability of Florence to patients we will pilot a prototype Florence protocol with approximately five participants and gather in-depth feedback on the text-message content, language and framing, via semi-structured interviews. The protocol will be iteratively refined to ensure that it is accessible, engaging and aligned to patient needs.
Conclusion: This approach to intervention development will ensure that the Florence intervention addresses issues that are important to new hearing aid users. The findings of this study will ensure that the Florence protocol is fully functional and can be successfully implemented within future studies of intervention effectiveness and cost-effectiveness to improve outcomes for patients and the health service.

S1-3. Ecological Momentary Assessment App for the Open-source Speech processing Platform (OSP)
Vy Nguyen*, Wayne Phung1, Dhiman Sengupta1, Martin Hunt1, Varsha Rallapalli2, Harinath Garudadri1
1University of California - San Diego, 2Northwestern University

Background: EMA incorporates multiple experience sampling methods, including real-time outcomes assessments with human subjects. These assessments are typically done via surveys sent to participants multiple times a day across a trial period to gather relevant, in situ feedback about their experiences. EMA has been proven to improve many drawbacks of retrospective recall, diary studies, and other traditional methods. Researchers adopting EMA implementations within the clinical psychology domain and recent hearing-related topics suggest the critical need for developing and supporting EMA within open-source ecosystems. We have developed an EMA system enabling researchers to (i) design EMA surveys using a simple text-based authoring tool and share with others for extensions; (ii) incorporate contexts (e.g., background conditions, GPS, number of talkers, etc.) to dynamically control the surveys in real time, and (iii) log audio parameters (raw, spectra, subband energies, etc.) before, during, and after a given survey.

Approach: cEMA, is an interactive graphical user interface that runs on OSP’s embedded web server framework. We developed an offline authoring tool for Windows, Mac, and Linux operating systems based on YAML, an acronym for “Yet Another Markup Language,” and also a recursive version “YAML Ain’t a Markup Language.” Researchers can leverage many YAML online tutorials and cEMA examples to become proficient in designing EMA investigations for hearing aids research. cEMA has configurable settings that allow researchers to specify surveys’ push logic (e.g., number of surveys per day, specific times, and combinations) and surveys’ context logic (e.g., number of talkers, GPS, background noise, etc.). After researchers have finalized an EMA logic, it is ported to OSP enabling participants to complete surveys either on their smartphones or tablets. Survey results reside within OSP’s internal memory for added security and privacy. In addition, cEMA can passively sense and capture audio and environment characteristics during surveys. This provides researchers additional information on the number of social interactions, speaker turn-taking in discourses, etc., while addressing privacy and security concerns. Further, cEMA results serve as the ground truth to create labeled environments for machine learning and artificial intelligence tools related to emerging hearing aids.

Results: We present cEMA design methodology and provide hands-on training to IHCON audience. Researchers can use existing configuration files, edit them to add new functionalities, or create entirely new cEMA investigations using example YAML files. We will also provide training to analyze EMA surveys based on individual and group-level responses.
Background: Individuals with similar audiograms may receive fairly similar hearing aid (HA) gain settings, validated and fine-tuned with real-ear measures for better precision. Still with a “perfect” fit, some perceive benefit in speech understanding, while others do not, suggesting untargeted inter-individual differences. Hearing loss may result in one or more deficits in auditory perceptual abilities that are heterogenous among individuals and, along with cognitive abilities, may be important factors affecting HA benefit. The goal of this study was to identify “auditory perceptual profiles” based on individual differences in auditory perceptual abilities and to determine any associations with subsequent HA benefit. A goal, with precision audiology, is to leverage perceptual profiles to target compromised perceptual abilities in hearing aid fittings.

Methods: A condensed test battery using the Portable Automated Rapid Testing (PART) platform assessed the auditory perceptual and cognitive abilities of twenty older adults with mild to moderately-severe hearing loss. Assessments included measures of frequency selectivity, spectro-temporal processing, temporal fine structure (TFS) and binaural processing, temporal envelope perception, spatial release from masking (SRM), and measures of working memory and fluid intelligence. A step-up design was used in which listeners were evaluated unaided for two weeks followed by three weeks aided. Ecological momentary assessment (EMA) surveys reflecting real day-to-day listening experiences were collected four times per day for the first (unaided) and last (aided) two weeks. Individuals were fit with HAs using clinical best practices. An objective hearing in noise test and standard HA questionnaires were administered prior to and after HA use.

Results: Cluster analyses identified three auditory perceptual profiles from individuals’ auditory perceptual and cognitive abilities. One profile (A) indexed poor performance on all perceptual and cognitive tasks and was associated with minimal HA benefit. Profiles B and C were associated with good perceptual performance and greater HA benefit. Profile B differed from C in which measures of HA benefit were significant. Working memory and SRM were major factors separating the “best” profile (B) from the other two (A and C); whereas the other two profiles were separated by performance on frequency selectivity, spectro-temporal sensitivity, and TFS processing.

Conclusion: Auditory perceptual profiles were associated with different degrees of HA benefit, especially as measured under ecologically valid conditions using EMA. Auditory perceptual abilities may be important precursors in identifying realistic expectations and, in the long-term, HAs may precisely target auditory percepts to improve benefits and successful HA adoption.
Ryan Corey*, Andrew Singer¹
¹University of Illinois Urbana-Champaign

Some of the most challenging listening environments, especially for people with hearing loss, are group conversations in crowded spaces, such as restaurants. Conventional hearing aids perform poorly in noisy environments because their microphones capture a mixture of speech and unwanted background noise. Remote microphones can improve intelligibility in noise by transmitting speech directly from a talker to the ears of the listener. However, commercial remote microphones are unsuitable for group conversations because they work with only one talker at a time and do not preserve spatial cues such as interaural time and level differences. These spatial cues are especially important for group conversations because they help the auditory system to follow speech from multiple talkers.

We have recently proposed an immersive multitalker remote microphone system that combines the low noise of remote microphones with the realistic spatial cues of earpiece microphones [1]. A set of adaptive filters processes the remote microphone signals to match the magnitude and phase of the earpiece signals. Because it relies on the earpieces as references, the system does not need to explicitly localize or track the talkers and the enhanced remote signals can be seamlessly mixed with the live sound at the earpieces. A basic version of the system was implemented in real time on the Tympan open-source hearing aid development platform [2].

In this presentation, we extend the previous implementation to improve performance in real-world environments. In particular, we consider robustness against talker and listener motion, crosstalk between two or more talkers, delayed auditory feedback of the listener’s own speech, and double-talk. We also demonstrate a cooperative conversation enhancement system for multiple users with listening devices. We compare experimental results from laboratory equipment with the real-time implementation on the Tympan hardware platform. [This work was supported in part by an appointment to the Intelligence Community Postdoctoral Research Fellowship Program at the University of Illinois Urbana-Champaign, administered by Oak Ridge Institute for Science and Education through an interagency agreement between the U.S. Department of Energy and the Office of the Director of National Intelligence.]

References:

S2-2. Realtime Multirate Multiband Amplification for Hearing Aids
Alice Sokolova*, Dhiman Sengupta¹, Martin Hunt¹, Rajesh Gupta¹, Baris Aksanli², Varsha Rallapalli², Fred Harris¹, Harinath Garudadri¹
¹University of California - San Diego, ²San Diego State University, ³Northwestern University

Background: Subband amplification is a basic feature of modern hearing aids (HA) to compensate for individual hearing loss. However, the temporal and spectral nature of hearing loss poses many challenges for the HA signal processing designers. Larger number of frequency
channels offers more precision and accuracy for fulfilling HA prescriptions, especially for unusual hearing loss patterns, but increases the complexity and reduces battery life. There is also a lack of understanding on how to accurately satisfy attack and release time parameters in a real-time system based on ANSI 3.22 guidelines, to facilitate systematic investigation of these parameters on listener outcomes.

**Approach:** We present a novel multiband real-time amplification system for HAs using multirate signal processing to minimize the complexity and power consumption of processing multichannel audio. The system offers precise control of the temporal dynamics of WDRC. The filterbank is based on audiometric frequencies comprising eleven half-octave frequency channels spanning five octaves, from 250 Hz to 8000 Hz, uniformly distributed on the logarithmic scale, with each band having a proportionate bandwidth, mimicking the known properties of cochlear transduction. Further, each octave of audio channels is mapped to a different sampling rate, such that each group of bands is processed at the lowest possible sampling rate. Processing lower sub-bands at the original Nyquist rate results in redundant samples. The proposed multirate system removes this redundancy and offers dramatic reduction in complexity. We implemented a new WDRC subsystem using an automatic gain control (AGC) using a Hilbert Transform for the envelope estimation in each band in the lower sampling domain. This approach accurately satisfies attack and release time specifications for the compression parameters as suggested from ANSI 3.22 guidelines.

**Results:** We implemented the proposed multirate HA algorithm on the Open Speech Platform (OSP) – an open-source system hearing loss research. On-target measurements on the OSP hardware show that the system runs in real-time, offers better frequency resolution, and with less complexity than prior sub-band amplification tools available in OSP. The proposed multirate filter bank provides a 14x reduction in complexity compared to a single-rate implementation, and has an algorithmic latency of 5.4 ms – well within the conventional threshold for real-time operation. The WDRC subsystem satisfies ANSI 3.22 guidelines within 0.5 ms. We present acoustic measurements (e.g., using Verifit Verification Toolbox, HASQI) to confirm the accurate static, and dynamic behavior of the proposed system for the ISMADHA hearing loss profiles.

**S2-3. Designing the Real-Time Master Hearing Aid (RT-MHA) Framework for the Open Speech Platform (OSP)**

*Dhiman Sengupta*1, Martin Hunt1, Harinath Garudadri1, Rajesh Gupta1

1University of California - San Diego

**Background:** The open-source audio processing tools for hearing aid (HA) research community came to a consensus that the best hardware platform for the next generation of HA research tools should be based on mobile computing platforms. These platforms include single board computers (SBC) like the Raspberry Pi, Qualcomm 410c, the Beaglebone, etc. The reasoning behind choosing these SBC platforms is that they are the ideal combination between the available computation, energy efficiency, and programmability. These SBC platforms achieve the ideal combination by using multiple super-efficient CPU cores, usually two or more, among other processing elements, making these computing platforms, unlike traditional computers. Therefore, it is essential to design the research tools with the hardware in mind.

**Approach:** This work makes two critical contributions to this area. The first contribution is a detailed analysis of the best way to set up the operating environment for these SBC platforms to get the most out of the hardware, no matter the open-source HA framework used. Our analysis found that we achieve the best performance when partitioning the computation resources between real-time (RT) and non-RT applications. Using this essential information and others gathered from the analysis, we designed the RT master hearing aid (RT-MHA) framework, the second critical contribution of this work. RT-MHA is a part of the Open Speech Platform (OSP), which has been designed and optimized to utilize the resources available on the SBC.
best. The RT-MHA framework gives HA algorithm researchers the most usable computational resources while minimally impacting the performance of non-RT applications like the embedded web server on the OSP platform.

**Results:** We show the different impact mechanisms on a modern-day SBC has on the real-time performance of the MHA algorithm. These mechanisms include the scheduler on Linux, the partitioning of computing resources, and the idling mechanism on SBC. Then we will describe how the RT-MHA framework can optimize the workload given what we found in our analysis to get up to 2 times more computation while being able to guarantee real-time operation.

11:10 a.m. – 12:10 p.m. **Mentoring Session 1A**
Industry: The job market (interviewing, developing a professional portfolio, negotiations)

11:10 a.m. – 12:10 p.m. **Mentoring Session 1B**
Academia: The job market (interviewing, startup, transition from postdoc, lab management)

12:10 p.m. – 1:30 p.m. Lunch
1:30 p.m. – 5:30 p.m. Leisure
5:30 p.m. – 7:00 p.m. Dinner
7:00 p.m. – 8:00 p.m.

**IHAS ORAL SESSION 3**

**S3-1. Effects of Patient Traits on the Relationship between Hearing Handicap and Readiness to Pursue Hearing Help**

*Lipika Sarangi*¹, *Jani Johnson*¹

¹The University of Memphis

Hearing aid (HA) success is commonly assessed in the later stages of an individual’s hearing health journey in terms of adoption, use, benefit, and satisfaction in daily living. For those in earlier stages, readiness to pursue hearing healthcare could be an indicator of progress towards positive hearing health outcomes. We explored whether patient traits that are predictors of later indicators of HA success also predict readiness to pursue hearing intervention. We further explored the moderating/mediating effects of modifiable patient traits on the relationship between hearing difficulties and readiness to pursue intervention using moderation/mediation analyses.

Sixty-two adults with self-reported hearing difficulties and no prior experience with HAs participated in this descriptive study. Perceived hearing-related handicap was assessed as the primary predictor and readiness to change as the outcome variable. Hearing aid self-efficacy (HASE), personality traits, and affective states were assessed in general and in hearing-related situations. Moderation/mediation analyses were performed to identify significant predictors of readiness and to test the effects of HASE and affective states on the relationship between hearing handicap and readiness to pursue help.

When HASE was considered in the model, individuals with greater hearing handicap, high HASE, more agreeable personality, and those who had hearing loss for a shorter duration were more ready to pursue help (F(9,52)=7.83, p<.0001). Moderation analyses demonstrated that the relationship between hearing handicap and readiness didn’t change as a function of HASE,
when controlling for personality and duration of hearing loss. When affective states were considered, individuals with greater hearing handicap, low Conscientiousness personality, and those who had hearing loss for a shorter duration were more ready to pursue help (F(14,48)=6.79, p<.0001). Mediation analyses demonstrated that the relationship between hearing handicap and readiness could not be explained by their relationships with affective states, when controlling for personality and duration of hearing loss.

This study confirms that HASE, certain personality traits, and duration of hearing loss are important variables that motivate people toward or away from becoming successful in their hearing health journey. Neither HASE and affective states significantly impact the relationship between perceived hearing handicap and readiness to pursue audiologic intervention. Future research should explore the associations among these patient traits and their impact on success at different stages of the hearing health journey to determine if an assessment of these factors is warranted.

S3-2. Hearing Aid Self-Adjustment methods: Adjusting Hearing Aid Gain and Noise Reduction Algorithms

Jonathan Goesswein*, Jan Rennies†, Birger Kollmeier‡
†Fraunhofer IDMT, Oldenburg Branch HSA, Oldenburg, Germany, ‡Department of Medical Physics and Acoustics, Carl von Ossietzky University of Oldenburg, Oldenburg, Germany

Self-adjustment of hearing aids (HAs) is subject of recent research, it should make adjustments more user-friendly and faster. In the current study we review a sequence of experiments with the aim to optimize the methods and user interfaces (UIs).

The fitting process of HAs is very time consuming and does not always lead to a satisfactory result. Typically, the process starts with an audiogram measurement, on which basis a prescription gain rule is calculated. This prescription is then further fine-tuned by the audiologist based on the patient’s reported perception. One possible solution of improvement is to involve the HA user more in the fitting processes by means of self-adjustment. In one study a two-dimensional (2D) graphical UI was utilized to reasonably reduce the available parameter space for the naïve HA user to self-adjust. The results of this study prove to be reliable and fast, while big interindividual variabilities were observed. This variability raises the question if a different starting point than the audiogram-based prescription could lead to similar preferred self-adjustment results. In the current study we aim to replace the audiogram-based prescription with an iteratively designed self-adjustment method using the evaluated 2D graphical UI. The initial parameter space is designed around an estimated prescription based on the sex and age of the HA user. With every iteration the size of the parameter space is reduced according to the HA user’s chosen preferences until – in the smallest and final parameter space – the final HA gain setting is determined.

Another aspect of HA fitting is the adjustment of noise reduction algorithms. Noise reduction algorithms are a double-edged sword: the more noise they suppress, the more distortions they create. Individually fitting these algorithms is crucial because HA users are known to differ strongly in their noise and distortion tolerance. Regarding this tolerance, two personal traits are described in the literature: noise haters who consistently prefer a rather strong noise reduction despite the distortions and distortion haters who consistently prefer a very moderate noise reduction to avoid these distortions. In the current study we aim to distinguish between these two personal traits and predict a preferred noise reduction setting based on a limited set of self-adjusted preference settings.

Overall, we demonstrate feasibility and the relevant factors and interindividual variabilities to be observed when designing self-adjustment strategies for HAs.
The Impact of Changes in Hearing Thresholds on Hearing Aid Fitting in Children who are Hard of Hearing

Kathryn Wiseman*1, Elizabeth Walker2, Jeff Crukley3, Ryan McCreery3
1 Boys Town National Research Hospital, 2 University of Iowa, 3 McMaster University

Children with hearing loss may show increases or decreases in their hearing thresholds as they age. With these changes, children who use hearing aids may need adjustments to their hearing aid programming to accommodate changes in unaided hearing thresholds. However, previous research suggests that hearing aids are often underfit relative to their prescriptive targets, as approximately 50% of pediatric hearing aid fittings deviate from targets by 5 dB or greater. In this study, we examined how changes in hearing thresholds impact pediatric hearing aid fittings and aided audibility in children who use hearing aids. Participants included 190 children with hearing loss who use hearing aids (age 8 months – 11.5 years old) with mild-to-severe hearing loss at baseline measurement. Pure-tone hearing thresholds were assessed and hearing aid verification was performed at annual intervals. We examined relationships between magnitude of change in pure-tone thresholds and: 1) change in deviation from prescriptive targets (i.e., RMS error) and 2) change in aided ability (i.e., aided Speech Intelligibility Index). We found that deviations from target were independent of changes in unaided hearing thresholds. For instance, some children improved their fitting (i.e., smaller RMS error) after a decrease in their hearing and others had poorer fittings after this decrease. Several children saw changes in RMS error despite no changes in hearing thresholds. However, changes in hearing thresholds (specifically higher frequency thresholds) were associated with changes in aided audibility, such that as hearing worsened, aided audibility decreased. These findings suggest that progressive hearing loss, particularly in the higher frequencies, may result in a loss of audibility even for children who use hearing aids. This may be because audiologists are not uniformly making the necessary adjustments to hearing aid programming to compensate for shifts in hearing, or because audibility cannot be fully restored via hearing aids due to the severity of the hearing loss. Clinically, these findings suggest that regular verification and programming of hearing aids is key to supporting a child’s access to audible speech. Ongoing research examines how ear canal acoustics impact longitudinal changes in hearing and verification of hearing aids.

8:00 p.m. – 10:00 p.m.

IHAS POSTER SESSION

SP101 Relationship between Beamformer Patterns, Compression, and Working Memory for Different Noise Configurations.

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Previous research has shown that speech recognition with wide dynamic range compression (WDRC) is associated with individual working memory ability, especially in adverse listening conditions. Our recent work has found that combining stronger directional processing (beamformers) with WDRC in hearing aids may reduce the role of working memory for speech recognition in ideal spatial conditions when the target signal is presented at 00 and the noise is at 1800. However, under realistic spatial conditions, the noise may arrive from multiple locations, rendering the beamformer less effective if the interfering noise is more diffuse and falls outside the directional null. We need to understand the impact of directional processing on the relationship between working memory and speech recognition in realistic spatial conditions. In this project, we extend our work to include different beamformer patterns and multiple noise locations.
Listeners with bilateral mild to moderately severe sensorineural hearing loss repeat low-context sentences mixed with multi-talker babble, presented at a realistic signal-to-noise ratio (SNR) in different spatial configurations. The spatial configurations include two (+90, -90) or three (+90, -90 and 180) noise locations around the listener. Wearable hearing aids, customized to the listener’s hearing level, are used to present four combinations of signal processing available in two different devices: a binaural-cardioid (commercial device) or a bi-directional beamformer (open-source device) combined with fast- or slow-acting WDRC (matched for both devices). Other advanced hearing aid features are turned off. Individual working memory ability is measured using the reading span test. To account for potential differences between devices, speech recognition with omnidirectional processing is measured in quiet. In addition, a signal fidelity metric is used to quantify envelope distortion in the processed signal across experimental conditions with respect to a linearly-processed signal in quiet.

Preliminary acoustic analyses show more change in signal fidelity (re: omnidirectional) between noise configurations with the binaural-cardioid beamformer compared to the bi-directional beamformer. Feasibility data show better speech recognition with the binaural-cardioid versus bi-directional beamformer for noise presented from three locations, but comparable performance with both beamformers for noise from two locations. Performance across conditions is related to individual working memory and signal fidelity using a linear mixed-effects statistical model. The results of this study will guide clinical decisions for fitting directional processing based on individual cognitive abilities and listener environments. [Supported by NIH-K01DC018324].

SP102 Effects of Hearing Status on the Time Course of Vocal Emotion Recognition
Hannah Shatzer*, Marc Pell, Gurjit Singh, Frank Russo

1Toronto Metropolitan University, 2McGill University, 3Toronto Metropolitan University, Phonak Canada, University of Toronto

Prior research has shown that adults with age-related hearing loss demonstrate deficits in auditory emotion recognition, and that the use of hearing aids does not mitigate these losses. However, there may be some nuance in the deficits that hard-of-hearing and hearing-aided individuals experience that is not captured by traditional paradigms where emotional stimuli are presented. The current study investigated the time course of vocal emotion recognition in order to assess differences in emotion recognition based on hearing status. In a remotely-conducted study, adults with normal hearing or mild-moderate unaided hearing loss completed an adaptation of the auditory gating paradigm used by Pell and Kotz (2011). Seven-syllable pseudo-utterances conveying different emotions (anger, fear, sadness, happiness, and neutral) were divided into gate intervals based on the number of syllables. Participants completed a forced-choice task identifying the emotion of each stimulus at each gate interval in a successive, blocked design, beginning with one syllable presented and ending with all seven syllables presented. The emotion identification point was defined as the gate in which responses were correct and remained correct for subsequent gates. Results indicated that participants with hearing loss identified each emotion at later gates than normal-hearing participants and were less accurate overall. At the emotion identification point, individuals with hearing loss also required greater variability in acoustic features (e.g., range of f0 and amplitude) compared to normal-hearing participants, suggesting that challenges with identifying vocal emotions are partially tied to the acoustic characteristics of different emotions. Data collection is currently underway for an in-lab follow-up study using the same paradigm with hearing aid users to assess the effects of hearing aids on the time course and overall accuracy of vocal emotion recognition.

SP201 A Questionnaire Study on the Impact of Face Mask on Individuals Suffering from Hearing Loss during the Pandemic
Preston Ho*

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Objectives: During the COVID-19 pandemic, mask wearing has become a major part of our daily lives. Despite its benefit of limiting transmission of respiratory droplets, face mask can introduce a barrier against communication and can be a source of inconvenience for those wearing hearing aids. We aim to investigate the impact of face mask on hearing aid usage and selection for individuals suffering from hearing loss.

Method: 72 patients aged 54 – 93 completed a questionnaire regarding the impact of face mask on communication, hearing aid usage, selection, and the decisions to pursue hearing aids at a busy Ear, Nose and Throat practice in San Francisco. Questionnaires were answered either at the first hearing evaluation...
Results: The questionnaire showed that face mask caused a substantial impact on communication in more than 70% of the participants. In particular, those with severe hearing loss or greater reliance on lip reading were more severely affected by face mask use. The impact on communication also drove more individuals to seek medical advice on hearing loss and to obtain hearing aids of their choices.

Conclusions: The widespread use of face mask during the pandemic contributed to communication problems among individuals with hearing loss and may explain the rebound growth on the demands of hearing aid products in the post lock-down era. Hearing care providers must take into account the multitude of physiologic and psychological effects of face mask wearing when counseling patients about hearing aid selection.

SP202 Investigation of hearing device technologies for individuals with hearing impairment in noisy workplaces
Solenn Ollivier*, Jérémie Voix1, Christian Giguère2, Rachel E. Bouserhal3, Fabien Bonnet3, Hugues Nélisse3
1École de Technologie Supérieure, Montréal, Canada, 2University of Ottawa, Ottawa, Canada, 3Institut de recherche Robert-Sauvé en santé et en sécurité du travail (IRSST), Montréal, Canada

In the United-States, 12% of the working population suffers from hearing loss and 25% is exposed to hazardous noise. For many workers operating in high-level noise environments, wearing hearing protectors is often necessary to reduce sounds to safe levels. Because hearing loss is characterized by higher hearing thresholds, these protectors often worsen a hearing-impaired user’s ability to perceive and localize sounds. Thus, such users are more likely to suffer from difficulties in communication and performing tasks efficiently and safely. Moreover, they may wear hearing aids at work, with or without hearing protectors, even though consequences of this practice have not been extensively researched. There is a need to develop a protective hearing aid able to amplify sounds of interest thus preserving speech perception yet reducing noise exposure when necessary. This work explores three axes: enhancing communication, managing noise exposure, and maintaining safety.

Communicating in noise is a common struggle, especially for people with hearing-impairment. To enhance communication, the hearing technology should be individualized by adapting amplification parameters according to the user’s hearing loss while preserving speech intelligibility. Multichannel wide dynamic range compression (WDRC) is widely recognized in current hearing aid technologies to personalize the device depending on the user’s hearing profile. Protection is allowed with both amplification of soft sounds and reduction of loud sound levels. However, some studies have highlighted that, depending on the selected parameters and the noise environment, speech intelligibility may be compromised after WDRC processing. Speech intelligibility relies greatly on acoustical cues, i.e. frequency and time components of the speech signal. This research will explore how WDRC parameters affect speech intelligibility in the context of noisy occupational environments. After objective evaluation and optimization, the designed algorithms will be validated on participants with or without hearing loss.

Noise hazard depends both on the level and the duration of noise exposure. The implementation of real-time in-ear noise dosimetry will be combined with adaptive amplification/compression to prevent the worker from being overexposed to excessive noise. This will also pave the way for research on a better understanding of the risks faced by workers with hearing-impairment operating in high-level noise environments.

Attenuating loud sounds combats over-amplification and should prevent further hearing loss but can also lead to safety issues. Alarm sounds and other key sounds are loud by nature but should not be missed by the user. Attenuation/compression algorithms should then be adjusted such that the workers’ safety is guaranteed.

SP203 Development of a new German speech test using synthetic speech
Saskia Ibelings*, Thomas Brand1, Inga Holube1
1Institute of Hearing Technology and Audiology, Jade University of Applied Sciences, Oldenburg, Germany, 2Medizinische Physik, Universität Oldenburg, Oldenburg, Germany

Existing German speech-recognition tests have disadvantages like training effects, outdated words, or a limited number of sentence lists. The goal of this project is to develop a new speech-recognition test using a Text-To-Speech (TTS) system to generate the speech material. First, the quality of different TTS systems was evaluated. A TTS system based on deep neural networks was rated best in the quality dimensions prosody, speech flow, and naturalness. Using
Ultrasound (US) research has grown rapidly in the past decade for noninvasively modulating brain regions with high spatial resolution and as a tool for creating transient openings in the blood-brain-barrier for drug delivery, showing exciting applications of this new technology. While investigating the neuromodulation technique, our lab discovered that US applied to the head readily activates the auditory system through vibrations of cerebrospinal fluid that then directly vibrate fluids within the cochlea (Guo et al., Neuron, 2018). Due to the potential applications of US induced auditory activation, our group is interested in characterizing safe levels of US pressure for the hearing system, particularly when used for neuromodulation, blood-brain-barrier, or hearing applications. To characterize this, we collected auditory brainstem responses (ABRs) and electrocochleography (ECochG) in response to air-conducted acoustic pure tones (2, 4, 8, 12, 20, and 30 kHz) and broadband noise at varying levels (10-70 dB SPL) before and after US stimulation in anesthetized guinea pigs. Both acute and chronic preparations were performed to fully characterize the potential for a parameter to be damaging. Control data was also collected for characterizing the stability of the recording protocol and for standard noise-induced hearing loss (NIHL). We assessed ABR and ECochG thresholds, amplitudes, and latencies over time to identify changes that are associated with hearing damage. Some tested US parameters showed neurophysiological changes associated with hearing loss with most tested parameters outside of the hearing aid modality showing some form of loss. Parameter settings used to effectively send complex information to the auditory system don’t show any changes associated with hearing loss at lower pressures. When changes were seen, threshold shifts were most prevalent in the high frequencies with some more severe cases of US stimuli causing threshold shifts in the middle frequencies as well with a similar pattern to NIHL. A reduction in ABR wave amplitudes can be seen also and including some parameters without significant shifts. Choice of center frequency may have an impact on safe level ranges. Future studies will include an in-depth characterization of the US parameters of interest also in large animal models that better mimic the head size in humans.

**SP204 Hearing Safety of Transcranial Ultrasound Modalities with Emphasis on Parameters for a Novel Hearing Aid Device**

*John Basile*, Gerardo Rodriguez, Hubert Lim

*1University of Minnesota*

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the preferred TTS system, sentences from an existing German speech-recognition test were synthesized and compared to the original regarding listening effort and speech recognition. The synthetic speech resulted in better speech-recognition scores and listening effort was comparable to that for natural speech. In conclusion, the TTS system is suitable for the generation of speech-recognition tests.

In a next step, the requirements for the new speech-recognition tests were compiled. These included that the speech material should be composed of meaningful and nowadays well-known words covering different parts of speech. Three to four words should each be combined in the same natural syntax to form sentences or phrases. A large amount of those combinations and test lists should be available to allow test repetitions and avoid training effects. It was decided that the new speech material is composed of phrases of the structure "article-adjective-noun-infinite" (German: “Den netten Mann grüßen”; English with different word order: “to greet the nice man”) and is therefore called Oldenburger Phrasen (OIPhra).

To compose the speech material, several annotated German corpora were filtered by word categories to obtain as many different words as possible. Since the words should be well-known, their frequency was analyzed with a database. With the selected words, thematic noun categories (e.g., food, people, ...) were formed and adjective and infinitives which are related to the categories were added. Subsequently, each noun per category was combined with all adjectives in the same category and meaningless, discriminatory, or too negative combinations were dismissed. This process was repeated for the nouns and infinitives. The remaining adjective-noun and noun-infinite combinations were merged and the selection process was applied again. Test lists formed of 20 phrases each should be balanced in terms of number of syllables, phoneme frequency, noun categories, articles (German: den, die, and das) and infinitives without prefix (e.g., to inform) and with prefix (e.g., to misinform). This contribution presents the procedure for developing speech-recognition tests and discusses its potential for continuous update of the speech material.

**SP205 Effects of Mandibular Motion for In-Ear Hearing Devices**

*Robert Budinsky*, Nathan Higgins, Erol Ozmeral, David Eddins

*1University of South Florida*
Design of a one-size-fits-all hearing device such as a hearing aid, in-ear monitor, or earbud is difficult due to individual variability in pinna size and shape, ear canal volume and geometry, and cerumen production. Further increasing design difficulty, is the poorly understood interaction between mandibular movement, device placement, and its impact on device performance. Mandibular motion (occurs during talking, singing, chewing, breathing, head movement) provides a nearly continuous source of external ear canal movement resulting from forces applied by the condyle of the mandible against the walls of the ear canal. Here, the effects of mandibular motion on the attenuation provided by a closed ear piece (earplug) were investigated by testing anatomically correct models of human ears using a custom acoustic test fixture developed for this study. Anatomical ear models simulating the entire pinna and canal up to the second bend were obtained from impressions of human ears (N=60 ears) with the jaw open, jaw closed, and jaw open with head turned toward one side. An acoustic test fixture was created that housed a standardized ear simulator with ear canal extension (GRAS RA0045), microphone (GRAS 40AH ¼”), loudspeaker (JBL D220TI), and a single-board microcontroller with stepper motor (Arduino). Mandibular motion was simulated in repeated cycles using a computer-controlled drive-pin with a silicone tip that articulated with the portion of the model ear canal that contacted the condyle of the mandible (i.e., the mandibular bump). The test procedure simulated various levels of mandibular motion by measuring insertion loss in the presence of 130 dB pink noise (attenuation with no earplug minus attenuation before and after various number of mandibular motion cycles) with and without the presence of artificial cerumen. Results indicate that as few as 50 cycles of mandibular motion could significantly compromise hearing device coupling and compromised insertion loss was highly subject-dependent. The methods developed for this study represent a novel approach to simulate the interaction between the physical design and real-life usability of hearing devices. The apparatus and methods used here could be extremely valuable in future investigations of the effects of mandibular motion on the fit and acoustic seal of custom and non-custom ear pieces.

Abstract Text Ultrasound stimulation (US) is an exciting new technique to non-invasively modulate neural activity with high spatial precision. Recently, studies have demonstrated that US, when coupled to the animal, activates the auditory system through an indirect, peripheral pathway (Guo et al., Neuron 2018). The current hypothesis posited by these studies suggests that the method of activation is via a fluid pathway, in which the ultrasound waves vibrate the cerebrospinal fluid and travel into the cochlea via the cochlear and vestibular aqueducts. Due to the differing mechanism and pathway of activation, US induced activity of the auditory system will have different characteristics from traditional air stimuli. Our lab has previously reported on the comparisons between air-evoked and US-evoked neural activity, and how US can effectively evoke frequency specific activity when modulated with pure tones. However, these experiments have not explored the complexity of information which can be encoded.

In this study, we investigated the neural activity in response to complex signals transmitted via air-, bone-, and fluid-conduction (ultrasound). Neural activity was recorded using a two-shank 32-channel NeuroNexus electrode array placed in the central nucleus of the inferior colliculus (ICC) of anesthetized guinea pigs. Air-conducted stimuli consisted of guinea pig vocalizations presented via a speaker. For fluid-conducted stimuli, we extracted the envelope of the vocalizations and then amplitude-modulated a 220 kHz sinusoid. This signal was presented via an ultrasound transducer coupled directly over the brain of the guinea pig with agarose and a focusing cone. Bone-conducted stimuli were presented with a B-81 bone conduction device coupled to the skull via a skull-nut system (Curthoys et al., Exp Brain Res, 2006).

Our results demonstrate that complex auditory information can be encoded in ultrasound, and the ICC reliably responds to vocalizations when presented via air-, bone-, or fluid-pathways. Our analysis demonstrates similarities between the air-, bone-, and fluid-driven stimulation approaches while also highlighting that these paths are different. Further research will investigate how the fluid pathway differs from the air-conduction pathway. Better understanding of how ultrasound stimuli act on the auditory system can help guide the development of next-generation hearing devices, possibly even a combination of technologies leveraging the different conduction pathways.

SP206 UltraHearing: Complex Activation of the Auditory System Using Body-Coupled Ultrasound
Gerardo Rodriguez Orellana1, John Basile1, Hubert Lim1
1University of Minnesota

SP207 Identifying the Cues that Support Intelligibility of Reverberant Speech
People typically communicate in indoor environments where speech is degraded by both noise and reverberation. Particularly, reverberation-related distortions alter the spectral and temporal envelope cues that support speech perception. The effect of these perturbations is more intrusive for people with hearing impairments and hearing aids. Identifying the cues that help such people successfully understand speech in different environments might not only help characterize the impact of environmental factors on their ability to understand speech in different contexts but also help us test the effectiveness of any signal processing that is intended to aid listeners in these environments.

The goal of this experiment is to identify the relative importance of envelope modulation-rate cues in determining the intelligibility of reverberant speech. In order to analyze the relative importance of different modulation-spectral cues, we first obtained sentence-level intelligibility scores from a total of 22 adults. Ten participants (range: 20-32 years; mean: 24.5 years) formed the normal hearing group (NH group) as defined by air conduction thresholds of 20 dB HL or better at octave frequencies 250 Hz through 8 kHz. Twelve participants (range: 53-77 years, mean: 65.3 years) with mild to moderate cochlear hearing loss formed the group with hearing loss (HL). These participants listened to monaural presentations of IEEE sentences in their better ear (or right ear for participants in the NH group) processed through the impulse responses of 4 different rooms. In total, each participant listened to 528 sentences, including envelope expanded versions of these reverberant conditions. Sounds were presented at 70 dB SPL, with listeners with hearing loss receiving NAL-R-based amplification. These sounds were also analyzed to extract the quantitative measures of changes to modulation spectra. The relative importance of different acoustic cues were examined by fitting the different measures to participant intelligibility scores. The effectiveness of the reverberation-specific weighting of cues was also compared to the Hearing-Aid Speech Perception Index (HASPI) version 2 predictions of speech intelligibility. This analysis provides insights into the relative importance of different modulation rate filters and envelope vs fine structure changes on intelligibility of monaural reverberant speech subjected to hearing aid signal processing. [Work supported by a research grant from GN ReSound to the University of Colorado at Boulder.]

The operator-aided audiometry (UAud) project aims at introducing an automated system for user-operated audiometric testing into everyday clinical practice. In that context, the Audible Contrast Threshold (ACT™) test is proposed as a test of supra-threshold hearing ability and as a language-independent alternative to speech-in-noise tests. Here, five user-operated ACT™ (UACT) test-paradigm candidates were evaluated in terms of performance and usability by 28 participants with diverse hearing and cognitive abilities. The five test candidates differed in the task and the procedure. UACT-0 and UACT-1 use a train of consecutive stimuli and the patient must indicate “when” the target was presented. UACT-2 through UACT-4 employ a sequential presentation of three separate intervals; UACT-2 uses a 3IFC task, where the patient must indicate whether the second interval contains target or reference (“which”), while UACT-3 and UACT-4 apply a 3AFC task, where the target is presented in one of the three intervals and the patient must indicate “where” the target stimulus was. The study was divided in two sessions conducted at least one week apart. To investigate the efficacy of non-verbal instructions (using pictograms), the participants underwent a block with the five tests in random order with exclusively non-verbal instructions. Three additional blocks were carried out for the purpose of evaluating test-retest reliability and training effects. Furthermore, the clinical audiologist-operated ACT™ and a 3AFC spectral-temporal modulation (STM, research baseline) test were performed for comparison in each session. The results showed that only 50% of the participants with lower cognitive abilities were able to provide reliable thresholds in the first block. The main factor affecting the thresholds estimated with the different tests was the task. All tests provided a reasonably good test-retest reliability, while the most reliable one was UACT-2 (3IFC task). UACT-2 also showed an excellent agreement with both the clinical ACT™ and the STM baseline tests. In terms of usability, the only significant aspect was the self-perceived duration. All participants reported a good experience with all test candidates with no significant differences among their judgements. Overall, UACT-2, the test based on a 3IFC with a custom
Speech and language training, with consistent use of appropriate amplification devices are crucial factors for successful rehabilitation of hearing-impaired children. The overall rehabilitation journey is economically and socially challenging endeavour for patients in resource-constrained settings. The cost of hearing aids (HA) can range from ₹20,000 (~$260) to ₹3,00,000 (~$3800), making them unaffordable for patients at the base of the economic pyramid. Societal stigma associated with hearing impairment further compounds these economic challenges. An intervention in the form of an affordable HA, coupled with a smartphone application was proposed in our earlier work to address these issues. This work focuses on the design approach of an affordable HA, derived from the open-source Tympan platform.

Functional and performance requirements for the HA design were derived from a WHO guidance document (World Health Organization, 2017. Preferred profile for hearing-aid technology suitable for low-and-middle-income countries.), and benchmarking of commercially available HAs in the Indian market for profound hearing impairment. These specifications were limited to essential parameters for appropriate amplification. To build upon an existing open-source hearing aid platform and a comparison of multiple available platforms was conducted, which pointed to Tympan Rev–D (an open-source hearing aid developed by Tympan) as the most suitable one.

Tympan Rev–D is a body worn hearing aid based on Teensy-3.6 processor, however, technical, and financial constraints within the Indian manufacturing ecosystem led to Design for Manufacturing (DFM) changes. Further changes were also undertaken to meet derived requirements. The developed HA, built upon Tympan Rev–D was subjected to a battery of tests in accordance with relevant standards (ANSI S3.22-2009 Specifications) using the Fyre-Phonix analyser and the results were found to be meeting all derived requirements. In summary, device had OSPL90 of 128dB, Full-On-Gain of 52dB, Equivalent Input Noise of 26dB, Total Harmonic Distortion of <1.4% with suitable Frequency Response. Based on clinical immersion activities, a headband model was designed with children as the target user. The DfM changes led to a reduction in the cost of manufacturing by an order of magnitude (~10x) and enabling local manufacturing in India without compromising any of the technical requirements. The team is currently undertaking efforts for size reduction, and a device embodiment design catering to a child’s lifestyle. These design changes will be made available in the open-source domain. This can facilitate distributed economical manufacturing of open-source hearing aid hardware and its wider community adoption.
found to be significantly different between self-fit and NAL-NL2. In addition, a paired comparison test did not show a consistent preference of the participants toward either of the two fits.

SP302 Over-The-Counter Hearing Aids Challenge the Core Values of Hearing Healthcare
Katherine Menon*, Michelle Hoon-Starr¹, Katie Shilton¹, Eric Hoover¹
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Regulatory changes in the United States introduced over-the-counter (OTC) hearing aids with the goal of increasing the accessibility and affordability of hearing healthcare. These changes represent a shift away from the provision of hearing aids by licensed audiologists and hearing instrument specialists. If the proposed solution does not share the same values as the current model, then it may fail according to existing metrics (e.g., poor fit-to-target) and still be highly successful by its own metrics (e.g., devices on more people). Our recent work identified the values of hearing healthcare service delivery. In this study, we evaluated the relative importance of these values across service delivery models, and the extent to which regulatory changes represent a coherent re prioritization of values.

We performed a qualitative content analysis of two document categories: critique documents representing the motivation to create an OTC model, and regulatory documents governing the implementation of OTC. Team members coded portions of text for the values they expressed. In total, 29,235 words were coded across 72 pages in four documents. Rank-order analyses were performed to determine the relative importance of values within each category of documents, between document categories, and in comparison to the existing model.

We observed a strong association between the rank order of values within each category, indicating that values are internally consistent in both critique and regulatory documents. Comparing between categories, the rank order of values in the regulatory documents was largely inconsistent with the critique documents, suggesting that the OTC model does not address the barriers that motivated its creation. Differences in the rank order of values in the regulatory documents compared to the existing model showed that the OTC model represents a values shift, but it remains unclear what values are prioritized by the OTC model. In order to evaluate the extent to which regulatory changes improve hearing healthcare, we need to establish the values of the new model through a consensus of stakeholders, including underserved individuals from diverse backgrounds.

SP303 What is audiologic counseling and what could it be?: A mixed-methods study of adult hearing aid counseling and possible adaptations for remote service models
Dana Urbanski*, Erin O'Neill², Randi Rankl¹, John Ellison², Peggy Nelson¹
¹University of Minnesota, ²GN Advanced Science

According to a concept promoted by Freston, innovation is taking two things that exist and putting them together in a new way. The emergence of over-the-counter (OTC) amplification underscores this idea, as OTC manufacturers pair technological advances with elements of traditional, brick-and-mortar clinical audiology to create new solutions. Specifically, hearing scientists and professionals have shown interest in leveraging telehealth and mobile applications for delivery of audiologic counseling outside of formal clinic visits. To achieve this aim, we must define the core elements and desired outcomes of audiologic counseling in everyday clinical settings.

Toward this objective, we designed a mixed-methods study to examine audiologists’ implementation of adult hearing aid counseling in dispensing clinics. We are recruiting 20-25 audiologists from a variety of practice settings to complete semi-structured interviews and ecological momentary assessment (EMA) surveys. Interview questions explore how audiologists define, implement, prioritize, evaluate, and individualize counseling activities, along with audiologists’ visions for change/innovation in counseling service delivery. Interviews are conducted using video conferencing and transcription, quality checked for accuracy. After establishing interrater reliability, two research team members will code interview transcripts using inductive thematic analysis. EMA surveys collect quantitative data categorizing the type and frequency of audiologists’ daily counseling activities and barriers/facilitators to counseling. Participants complete daily EMA surveys using ExpiWell, an EMA smartphone application. EMA data will be summarized in descriptive statistics and interpreted relative to identified qualitative themes.

Early qualitative results reveal several key themes: a) counseling is time intensive; b) counseling is a process that takes place over several interactions; c) counseling is most effective when tailored and se-
quenced to patient characteristics including experience and attitudes; and d) counseling has an observable positive impact on adult hearing aid outcomes. Early EMA data confirm the time demands of effective counseling, with several participants citing appointment length as a daily barrier to counseling. EMA results reveal that audiologists spend substantial time reviewing basic device use, care, and maintenance, including smartphone applications/streaming, sometimes to the exclusion of other counseling activities. Audiologists highlighted ways in which mobile wireless technologies might improve and augment in-person counseling, including reminders/notifications for cleaning, care and maintenance, user-friendly availability of instructional videos, and delivery of listening situation-specific communication and hearing aid strategies.

This poster will present results of our full participant sample with data collection and analysis slated for completion in early-mid Summer 2022.

**SP304 Verification and Validation of a Self-Fitting Hearing Device**

*Jiong Hu*1, Jayaganesh Swaminathan1, Jade Kwan1, Mayra Rodriguez1, Alexis Dalager1, Anna Walters1

1University of the Pacific, 2Eargo Inc.

Current developments in advanced consumer electronics have allowed manufacturers to develop hearing-aid self-adjustment algorithms as an integral part of their technology. Such devices may change how the patients with hearing loss, clinicians, and the industry operate and interact with each other in the future. However, questions remain about the clinical efficacy and effectiveness of such devices/approaches compared to current standard of care. One such device is Eargo’s hearing aid that allows users to perform app based in-situ hearing screening and self-select gain parameters based on the results of the hearing screening. The primary goals of this study were twofold: 1) To validate the accuracy of hearing thresholds measured with Eargo’s in-situ hearing screener against the hearing thresholds measured following Audiology best practice methods (i.e., in-booth audiometric testing with a clinical audiometer) and 2) To establish comparability of Eargo hearing aids’ output with NAL-NL2 prescription targets through real ear measurements. As an extended goal, objective and subjective sound quality comparisons will be made between Eargo devices self-adjusted by the user compared with a traditional hearing aid fit-to-target by a clinician following Audiology best practice (ABP) methods.

127 subjects with normal-hearing and varying degrees of hearing loss were recruited. 100 subjects completed the hearing screener study. The results showed that the audiometric thresholds measured using Eargo’s in-situ hearing screener were comparable to the thresholds measured using ABP methods. Furthermore, the results showed excellent accuracy (93%), sensitivity (97%) and specificity (100%) for the thresholds measured with Eargo’s in-situ hearing screener compared to those measured with a clinical audiometer. Overall, the results indicated that, compared to ABP methods, Eargo’s in-situ hearing screener provides accurate and reliable inference about the hearing status of individuals with and without hearing loss.

For the second part of the study, 20 subjects completed the real ear measurements. The real ear aided response showed that the average deviation from NAL-NL2 targets was less than 5 dB SPL for audiograms representing mild-to-moderate hearing loss profiles. Results demonstrated that Eargo’s gain selection approach delivers hearing aid output comparable to a hearing aid fit to NAL-NL2 targets.

Taken together, the results validated the efficacy and effectiveness of a user directed self-adjusting hearing aid in providing adequate amplification for those with mild-to-moderate hearing loss. The results of this study may inform future design and development of self-adjusting hearing aid strategies grounded in principles of hearing science and clinical audiology.

**SP305 Community-Based Adult Hearing Care provided by Community Healthcare Workers Using mHealth Technologies**

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1University of Pretoria, 2Ear Science Institute Australia, 3London School of Hygiene and Tropical Medicine, 4hearX Foundation, 5University of Colorado School of Medicine

**Background:** Rising prevalence rates of hearing loss is of concern due to the global shortage of hearing healthcare services. Task-shifting to community healthcare workers (CHWs) supported by mHealth technologies has been recommended to overcome the lack of services.

**Objective:** This study evaluated the feasibility of a community-based rehabilitation model for providing hearing aids to adults in low-income communities by using CHWs supported by mHealth technologies.
**Method:** An implementation approach evaluated the hearing assessment, hearing aid fitting, and support process for adults with hearing loss in two low-income communities in the Western Cape, South Africa. CHWs facilitated hearing assessments and hearing aid fittings, after training by audiologists, over a period of 13 months from September 2020 to October 2021. Data was gathered using qualitative and quantitative measures and analysed using a mixed methods approach. Hearing aid outcomes were measured using the International Outcome Inventory – Hearing Aids.

**Results:** 148 of 152 adults in the community who self-reported hearing difficulties were successfully tested by CHWs during home visits. Most had normal hearing (39.9%) with 24.3% having sensorineural hearing loss bilaterally, 20.9% with suspected conductive hearing loss and 14.9% with unilateral hearing loss of which 5.4% was a suspected conductive loss. 40 adults met the inclusion criteria to be fitted with hearing aids of whom 19 were fitted with hearing aids bilaterally. Positive hearing aid outcomes and minimal device handling challenges were reported at 45 days post-fitting and was maintained at six months with no significant changes. CHWs were able to support and resolve minimal challenges.

**Conclusions:** Implementing a hearing healthcare service-delivery model in the community facilitated by CHWs is feasible. mHealth technologies used by CHWs enables a scalable service-delivery model for improved access and affordability in low-income settings. Future studies should compare this innovative model to conventional hearing care professional service-delivery models.

**SP401 Predicting real-life listening outcome based on in-situ acoustic and heart-rate measurements for listeners with normal and impaired hearing**

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Ecological momentary assessment (EMA) can provide insights into the real-life auditory ecology of hearing aid (HA) users. In a previous study, we combined EMA with acoustic data-logging and found that this can reveal situation-specific benefits of HA noise management (Andersson et al., 2021, Am J Audiol). Given that physiological measures such as heart rate (HR) are known to be sensitive to acoustic influences (e.g., sound pressure level and signal-to-noise ratio), we hypothesized that the inclusion of such data could provide even more detailed insights into daily-life listening outcome.

In the current study, participants with normal hearing (NH; N = 10) or mild-to-severe sensorineural hearing impairments (HI; N = 16) completed smartphone-based EMAs during a 2-week period. The HI participants were fitted bilaterally with behind-the-ear HAs with a single automatic program. The NH participants received a single HA each that they fastened to their collars. The HAs were used to collect continuous sound pressure level and signal-to-noise ratio data in the participants’ daily surroundings. Wristbands worn by the participants were used to collect complementary HR data.

Linear mixed-effects models with participant and time of day as random intercepts showed that the acoustic data could predict self-reported listening outcome for both participant groups (likelihood ratio tests against NULL models; both chi-squared, i.e., \( \chi^2(2) > 42.2 \), both \( p < 0.001 \)). Including time-synchronized HR estimates in the models improved the predictions further. In addition, the models revealed negative associations between self-reported listening outcome and both sound pressure level and HR. Signal-to-noise ratio was positively associated with self-reported listening outcome for the HI group.

Overall, these findings imply that combining EMA with in-situ physiological and acoustic data can increase the predictive value of real-life listening evaluations.

**SP402 Aiding in small group conversations – effects on behaviour of both the wearer and other group members**

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†University of Nottingham

People with hearing loss exhibit coping strategies to help navigate challenging social situations. It is plausible that wearing hearing aids may not eliminate the additional perceptual and cognitive challenges people with hearing loss may face. Furthermore, behavioural effects are likely to become more diverse and dynamic when there are more than two interlocutors, and group context is the conversation scenario where people with hearing loss are most likely to experience activity limitations and participation restrictions. Interlocutors with normal hearing may also adapt their behaviour, be that consciously or unconsciously, in
response to the difficulty being experienced by interlocutors with hearing loss. Therefore, the impact of hearing loss on conversation behaviour and the benefit of hearing aids should be considered in terms of their effects on all interlocutors in a group conversation setting.

This research explores to what extent the wearing of hearing aids by people with hearing loss alters communication behaviour of all interlocutors (both with and without hearing loss) in complex conversation settings.

Groups of four participants (two with mild-to-moderate symmetrical hearing loss (HL), two with normal hearing (NH)) were recorded having conversations in quiet, moderate, and noisy environments. Environments were presented contrasting bilaterally aided (own devices) and unaided conditions for HL participants. 3D motion capture, 2D video, and audio recordings were combined with self-report measures of conversation fluency, effort and success. Metrics of conversation behaviour were examined with respect to contrasting hearing ability group (NH vs. HL), noise level, and aiding in the HL participants.

In this poster we will present preliminary results, with behavioural metrics grouped not according to their modality, but according to the coping strategy they may be presumed to reflect. This promotes meaningful interpretation of ‘selfish’ and ‘altruistic’ behaviours (examining ‘me’ versus ‘us’ motivation), and of behaviours which superficially provide no perceptual benefit. This research provides a novel exploration of the effects of hearing aid use in small group conversations comparing NH and HL communication behaviours.

**SP403 Development and Feasibility of Using an Open-Source Portable Hearing Aid with EMA in the Real World**

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*The University of Iowa*

**Objective:** The purpose of this study was to investigate the feasibility of using a portable, open-source hearing aid paired with ecological momentary assessment and soundscape recording in the real world. Open-source hearing aids and ecological momentary assessment will enable the transparent and collaborative development and real-world testing of hearing aid processing algorithms. This study describes platform development, compliance, feasibility, participant experiences, and potential challenges and use cases.

**Design:** This study used the Portable Hearing Aid Lab (PHL), a BeagleBone computer with wired receiver-in-the-canal earpieces, running the Open Master Hearing Aid software. Participants (with and without hearing loss) were asked to collect data in 5–10 complex listening environments over a 7–10-day period. In each environment, participants wore the PHL and completed two ecological momentary assessments on a smartphone over the course of 30 minutes. Between each momentary assessment for participants with hearing loss, the smartphone triggered the PHL to switch gain settings from an aided (fit to NAL-NL2 targets) to unaided (acoustically transparent) condition, or vice versa, testing the possibility of within-environment A/B comparisons of hearing aid algorithms in the real world. The PHL also recorded the environment.

**Results:** 12 participants with hearing loss and 10 with normal hearing completed the study. Most participants were able to operate the devices, but not universally. 266 EMA surveys were completed, but only 127 had associated recordings. Absent recordings were determined to typically be the result of user error. Only 1 participant (hearing loss group) declined to complete the study because the devices were too complicated. Participants with normal hearing completed EMAs in, on average, 7 complex listening environments and participants with hearing loss completed EMAs in, on average, 5 environments. Compliance did not differ significantly between groups. Data were collected at home (46%), in restaurants and bars (14%), transportation (10%), shops (8%), work (6%), and other environments, suggesting participants could use the device successfully in different places. Participants also wore the PHL for a variety of listening activities, including passive speech listening (26%), one-on-one conversation (22%), music listening (17%), and group conversation (9%). Compliance tended to increase over the course of the study as training was adapted.

**Conclusions:** Using an open-source hearing aid platform paired with EMA and sound recording for within-environment A/B hearing aid algorithm comparison is feasible. However, pitfalls were noted, particularly user error, user frustration, and device limitations. Training and streamlined interfaces and procedures are critical for real-world use.
**SP404 Head Orienting Behavior When Following a Multi-Talker Conversation**

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**Introduction.** During speech communication and conversational turn-taking, listeners direct their head and eyes to receive meaningful auditory and visual cues and to relay nonverbal cues back to talkers. There is scant understanding of how listeners interact with multiple talkers, how this interaction may convey listener intent, and how hearing aids might take such movement or intention into account. This study investigated head orienting behaviors when participants were in an engaging audiovisual environment: a conversation among four talkers with live-action audiovisual (AV) information. Methods. Ten younger adults with normal hearing and twenty older adults with mild-to-moderate sensorineural hearing loss completed the study. Participants wore a headset snugly fit to the head upon which were small sensors. Movement of the sensors was tracked in real time by an infrared camera system (OptiTrack V120-Trio). Head movements were monitored during blocks of 90-second conversations, in which participants were tasked with following the content of the discussion (American football analysis). Blocks were designed to test head orienting behaviors for different AV source locations in broadband or babble background at +6 dB or +12 dB SNR. Hearing-impaired listeners were tested unaided and with hearing aids using omnidirectional microphones. We investigated the effect of these parameters on self-rated listening effort, which may mediate head movement activity levels, speech comprehension, or both. Results. Movement trajectories showed that hearing-impaired listeners turned much closer to the target location with an increase in latency compared to normal-hearing listeners. However, there was no significant hearing aid effect. Listening effort was significantly lower for normal-hearing listeners compared to hearing-impaired listeners. For hearing-impaired listeners, the listening effort decreased with aided listening. Conclusions. Head movement behaviors differ between hearing-impaired and normal hearing populations when following a conversation. It may be important for hearing aid spatial processing to consider the expectation that listeners are orienting towards target talkers when possible.

**SP405 Remote Psychoacoustic Tests Using Gaussian Processes for Personalised Hearing-Aid Fitting**

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The COVID-19 pandemic highlighted the need of having remote hearing testing and hearing-aid fitting to guarantee service delivery. They are also useful to provide high-quality service in remote or under-developed areas. However, remote tests are susceptible to level calibration and headphone responses. The standard procedures, often using up/down adaptive tracking, in which the stimuli are presented with a systematic combination of frequency/intensity, make many tests too long to be feasible for clinical practice. These problems are addressed in the present study by developing and evaluating audiogram, notched-noise and temporal masking curve tests that use Gaussian processes (GPs) and are implemented to be run on a web server. Since the notched-noise and temporal masking curve tests are based on the perception of the difference between signal and the masker, they are more robust than audiogram when the calibration is not known. The GPs presented stimuli only with informative combinations of frequency and intensity, which led to a reduction in testing time (50 trials in about 4 minutes) without losing accuracy (root-mean-square differences of 5 dB). Participants with previously known audiograms (having normal hearing, mild and moderate hearing loss) performed the remote tests. In particular, the audiogram was measured between 250 to 8000 Hz (Schlittenlacher et al., 2018), notched-noise tests were performed between 500 to 4000 Hz and with a set of 6 notch widths (i.e., 0|0, 0.1|0.1, 0.2|0.2, 0.3|0.3, 0.2|0.4, and 0.4|0.2) (Stone et al., 1992; Schlittenlacher et al., 2020), and temporal masking curves were measured between 500 to 4000 Hz and from -10 to 100 ms gap, with the linear reference (or off-frequency) threshold at 1600/4000 Hz (Lopez-Poveda and Johannesen, 2012; Johannesen et al., 2014). At the time of writing, three participants have finished the experiment. These pilot results have shown that thresholds could be established for audiogram, and for the wider notches in the notched-noise tests. The condition 0|0 (i.e., no notch) was not successful, so we changed the strategy for the first few trials for the next participants. The remote audiograms were compared with those previously obtained in the clinics. The notched-noise tests were used to estimate the auditory-filter shapes. The temporal masking curve tests were used to estimate the inner- and outer-hair cell dysfunction. We have been able to assess the feasibility of remote tests, and to collect comprehensive information on participants’ hearing.
Development and initial validation of the Hearing Aid Feature Importance Evaluation (HAFIE) questionnaire
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In this study, a novel questionnaire called the Hearing Aid Feature Importance Evaluation (HAFIE) was developed and initially validated. The aim of developing this questionnaire was to provide a structured, evidence-based methodology for hearing aid recommendations and selection. This may be achieved by allowing clinicians to gather patient attitude and self-reported importance ratings for different modern hearing aid features.

Initial questionnaire items were designed using the statements generated in a concept mapping investigation of the drivers of user preference between hearing aids at higher and lower technology levels (Saleh et al., 2021). A series of focus group interviews were conducted with experienced hearing care professionals (n=10) to assess these items and gather suggestions regarding further questionnaire design and content. The items were modified based on the focus group results and a scan of currently available hearing aid features.

Validation of this initial 34-item version of the questionnaire was conducted using an anonymous online survey tool (Qualtrics). Respondents were adults self-reporting hearing difficulties (N=218, median age = 48 years). Exploratory factor analysis was used to assess the factor structure of the dataset, using principal axis factoring and an Oblimin rotation. Three factors were identified, dividing the hearing aid features into the subscales: “Advanced connectivity and streaming”, “Physical features and usability”, and “Sound quality and intelligibility”. Seven items did not load well onto any factor or had high factor loadings on more than one factor and were therefore removed. This resulted in a 27-item questionnaire with three subscales. Reliability of each of these subscales was assessed via Cronbach’s alpha and item-total correlation, and each was found to be appropriate.

References:

SP407 Assistive Listening and Sound Quality during a Live Orchestra Performance
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In addition to difficulties listening to speech, music sound quality can be greatly affected by hearing loss and hearing aid processing. Any non linear processing on hearing aids has the potential to distort important elements of music. Live music has additional challenges compared to recorded music, in both volume and the bandwidth of frequencies that are necessary for the best possible sound quality. In order for hearing aid processing to have optimal processing, sound quality for live music needs to be maximized, so ecologically valid studies during live performances are essential to fully characterize sound quality.

Preliminary studies in the LIVELab and an experiment conducted during an orchestra concert showed that while music sound quality judgments in hearing aid users are subjective and variable between subjects, those with high musical sophistication are more critical and consistent in their judgments. Two different assistive listening systems, a telecoil loop and a 2.4 GHz RF streaming system, were tested during the concert to determine if such a system was beneficial for live music. Several different audio mixes from microphones amongst the musicians were rated during the concert for sound quality and loudness to find an optimal mix for the assistive listening systems. No clear optimal mix for improving sound quality was found, but the high musical sophistication group did show a preference for having the assistive listening system compared to the hearing aid microphones alone. In order to have more detailed and paired comparisons, a follow up study using a MUSHRA listening test with recordings from the initial concert will be performed, and initial analysis of results presented.

SP408 Factors Influencing Hearing Help-Seeking and Hearing Device Uptake in Adults with Hearing Difficulties: A Systematic Review of the Past Decade
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Objective: The aim of this systematic review was to examine the audiological and non-audiological factors that influence hearing help-seeking and hearing device uptake in adults with hearing difficulties based on the literature published during the last decade.
Design: Peer-reviewed articles published between January 2011 and February 2022 were identified through systematic searches in electronic databases CINAHL, PsycINFO and MEDLINE.

Results: Forty-two articles were included in the review. These studies investigated 72 and 161 factors respectively in relation to help-seeking and hearing device uptake. Significant non-audiological factors that influence both hearing help-seeking and hearing device uptake include the following categories: demographics (e.g., age, sex), health/cognition, and behaviour. Furthermore, factors in the social category affect hearing help-seeking alone whereas factors in the categories of finances/work, funding/insurance, audiology appointment, motivation, and technology affect hearing device uptake alone. Significant audiological factors that influence both hearing help-seeking and hearing device uptake emerged in the following categories: pure tone audiometry, self-reported hearing difficulties and beliefs, communication, and hearing aids. Additionally, factors in the categories of speech testing, hearing healthcare consultation, and readiness for change influenced hearing device uptake alone. Of the included studies, 28 were classified as level 4 evidence, 12 as level 3 evidence and 2 as level 2 evidence. In terms of quality, 37 studies were rated fair, 1 good and 4 poor quality.

Conclusions: Various factors relating to help-seeking and hearing device uptake have been investigated although limited studies examine factors like the influence of the hearing device cost. Some factors have conflicting findings, like the influence of self-reported health, requiring further exploration. Our findings inform clinical audiological practice for help-seeking and hearing device uptake by patients.

SP409 Hearing Aid use, Benefit and Satisfaction in adults: A Systematic Review
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Importance: Understanding the factors influencing hearing aid outcomes can inform treatment approaches and patient support towards optimal benefit and satisfaction. There has been a growing body of evidence accumulating across various studies over the past decade but with the last review of literature done in 2010.

Objectives: This study systematically review peer-reviewed studies (published between January 2010 and January 2021 on factors influencing hearing aid use, benefit and satisfaction in adults.

Methods: This systematic review was conducted in accordance with the Preferred Reporting Items for Systematic Reviews and Meta-Analyses (PRISMA) guidelines 2020. A systematic literature search was conducted from the following databases (i) Web of Science (ii) Scopus, (iii) PubMed, (iv) EBSCOhost including CINAHL, and Academic Search Complete. All relevant articles were identified, exported to the Ryann Systematic review software and screened by two researchers independently. Full text copies of articles identified during the search and considered to meet the inclusion criteria were obtained for data synthesis. Periodical searches were conducted prior to the completion of the systematic review to assess for any further studies. The snowballing of the reference list method was also used to identify related articles that may have not been found during the initial search. Articles found through this exercise were included except for unpublished and non-peer reviewed publications.

Results: A total of 1022 peer-reviewed articles were identified from the following databases (i) CINAHL, (ii) PubMed, (iii) EBSCOhost including Web of Science, and (iv) Academic Search Complete. After removing 346 articles duplicates, the remaining 676 articles were further reduced based on outcomes, Implantable device, population, timeframe, publication, study design. A total of 33 articles were included for data extraction and analysis. From the articles reviewed, audiological factors that had significant influence on hearing aid outcomes included hearing sensitivity, speech perception, self-reported hearing disability, ear, tinnitus and balance, hearing aid acoustics and features, hearing aid candidate factors, hearing aid fitting and follow up. Non-audiological factors that showed a significant influence on hearing aid outcomes includes demographics, social networks, mental health, psychosocial, health and socio-economic factors.

Conclusion: Factors other than hearing loss and hearing technology are important for hearing aid use, benefit and satisfaction. The result of this study places an emphasis on patient-centered care, where non-audiological factors such as social networks, mental health, health, socio-economic should be considered to optimize hearing aid outcomes.

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The use of auditory models is important for designing speech and audio processing algorithms for hearing assistive devices. These auditory models are often parameterized by a set of parameters relating to auditory function, e.g., hair cell loss or synaptopathy. In practice, the computational load of these auditory models can be very high thus limiting the feasibility of using the models as bio-inspired loss functions for deep learning based hearing loss compensation strategies or denoising strategies. Previous efforts have addressed this problem by training a neural network [1] for each parameter configuration of the auditory model which greatly reduces the computation time of the auditory model and allows for direct and efficient computation of the gradient when using the auditory model as a loss function but requires a new network to be trained whenever the parameterization changes. We propose an approach where a single neural network is trained, once and for all, to accurately simulate auditory models across their parameter spaces by conditioning the weights of the network on the parameters of the respective auditory models. This approach enables greater flexibility than training a single model for each parameter configuration, as any parameterization can be acquired on the fly. We showcase the approach on two different auditory models, the filterbank model of the cochlea and inner hair cells by Zilany et al [2], and the transmission line model of the cochlea by Verhulst et al [3]. The accuracy of the neural network is shown to be robust across both unseen inputs and different hearing losses.

References:

SP502 A Systematic Evaluation of Combined Noise Reduction and Dynamic Range Compression in Adverse Conditions

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Noise reduction (NR) and wide dynamic range compression (WDRC) are two essential building blocks in modern hearing aids to increase listening comfort and restore audibility. However, NR and WDRC commonly counteract each other. For example, the improvement in signal-to-noise ratio (SNR) after NR can be reduced by WDRC due to the amplification of residual noise. In addition, the extent of interaction between the NR and WDRC building blocks depends on their configuration and processing arrangement. A stronger NR system can provide a higher output SNR and a low level of residual noise compared to a more conservative NR system. However, it might attenuate soft speech components to a point where they cannot be amplified by dynamic range compression. A fast-acting compression system can increase audibility compared to a slow-acting compression system but might amplify the residual noise after NR. A serial arrangement can provide an increased amount of compression but also amplifies more residual noise compared to a parallel arrangement. This study investigated the influence of these choices by evaluating a large set of systems using noise reduction methods (including ideal, model-based, and convolutional neural networks) combined with either fast-acting, slow-acting, or adaptive compression settings in either parallel or serial arrangements. The systems were tested with noisy speech and evaluated using objective metrics (e.g. the effective compression ratio and the change in SNR). Each system was compared to a reference system that used ideal ratio-mask for noise reduction combined with fast-acting compression in a serial arrangement. The reference system was considered to provide the highest amount of both noise reduction and compression. Each system was compared to the reference system in terms of the similarity of their objective metrics. This was done by calculating the likelihood that the objective metrics from each system came from the same distribution as those of the reference system. The results showed that the NR stage had the largest effect in terms of similarity to the reference system, followed by WDRC and the processing arrangement. In addition, the results suggested that a sufficiently effective NR method was more similar to the reference system when fast-acting compression was used in a serial arrangement, while the model-based NR method was most similar to the reference system when adaptive compression was used.

SP503 Automatic analysis tools for exploring own-voice and near-ear sound pressure level distributions

IHAS August 9-10, 2022
Exploring acoustic conditions and listening experiences of people in their everyday environments has drawn a lot of attention. Our laboratory developed the smartphone-based ecological momentary assessment (EMA) system olMEGA which stores smoothed acoustical features (root mean square, RMS; auto and cross-power spectral density, PSD; zero crossing rate, ZCR) on a reduced time scale to preserve the privacy of the test participants. The recordings were performed by two head-worn, near-ear microphones attached to glasses. Simultaneous to the recorded acoustical features, subjective assessments are taken in situ by surveys on the smartphone.

This contribution shows analysis results of 13 near-ear recordings of hearing-impaired participants (aged 50 to 75 years) in an EMA study. The collected acoustical features and subjective assessments of approx. 4 days per participant were extracted from the smartphones and transmitted to a database server for further analysis. The database server facilitates data management and enables international collaboration. Furthermore, new features can be extracted automatically that are of interest for studying the impact of natural, acoustic environments on the communication abilities of elderly people. For this contribution, an algorithm detecting the test participant’s own voice (Own Voice Detection, OVD) and an algorithm that extracts the sound pressure level (SPL) in order to measure the sound exposure were applied to the EMA data. The OVD is based on a machine learning algorithm and a set of acoustical features derived from the olMEGA features. The evaluation of the OVD with a manually labeled real-world recording of one full day showed reliable and robust detection results.

The analysis of the EMA data shows that the grand mean percentage of predicted own-voice audio segments (OVS) during one day was approx. 10% which corresponds well to other published data. The grand median SPL across all participants and recording days was 50.3 dB(A). The OVS had a small impact on the median SPL over all data. However, for short analysis intervals, significant differences of up to 30 dB occurred in the measured SPL, depending on the proportion of OVS and the SPL of the background noise.

**SP504 Predicted Selection of Listening Programs Based on Sound Exposure**

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User-centric adaptation of audiological preferences across different contexts is a challenging task, as traditional clinical measurements of audibility do not reflect the cognitive perception of speech nor the binaural loudness of sounds in different contexts. Listening programs enable hearing aid users to adapt device settings for specific listening situations, which increases the personalization of their listening experience. However, studies investigating the real-world use of listening programs are lacking.

This study aims to investigate whether the selection of a specific listening program can be predicted based on the sound exposure of a user. Specifically, a binary classifier predicts whether the default (“General”) or the most frequently used additional listening program (“Speech in Noise”) is selected based on real-world, time-series sound environment data. Thus, we defined a time-series classification framework and compared multiple machine learning (ML)-based prediction models.

A state-of-the-art feature extraction model, MiniRocket, was used to transform environment features for the classification task. Due to the high number of features created by MiniRocket (10000 per raw sample), principal component analysis (PCA) was performed for reducing dimensionality to 1000 features per raw sample. Class imbalance during training was addressed by stochastic downsampling of the majority class. The final dataset was comprised of 6,613 program selections from 131 distinct users.

Next, we implemented two-level cross-validation to evaluate three binary classifiers: logistic regression (also serving as a baseline), gradient boosted trees, and a deep neural network. Due to the imbalanced nature of the data set (i.e., higher number of program selections in “General” over “Speech in Noise”), three metrics suited for imbalanced class distributions were used: balanced accuracy, F1-Score and Precision-Recall Area Under Curve (PR-AUC).

Training results on a balanced validation set showed the best-performing classifier – deep neural network – outperforms the other classifiers. Testing results on an imbalanced, realistic set revealed the model can make predictions given unseen data (balanced accuracy approx. 73%), but became less robust to imbalanced data (PR-AUC approx. 0.5). Finally, the deep neural network model was able to correctly label 69%
By rethinking contextual adaptation of HA settings as a time-series classification task, we validated the role of the sound environment in program selection and established a baseline approach for further investigating prediction models.

**SP505 Using Inverse Reinforcement Learning for Online Personalization of Hearing Aid Compression**

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In the currently practiced fitting of hearing aids, prescriptive compressions are used. These prescriptive compressions are normally defined by taking gain averages of a group of people having more or less the same hearing loss or audiogram. In real-world audio environments, people with the same hearing loss do not necessarily express the same preference. It is thus hypothesized that personalization of the hearing aid fitting process provides an improved hearing experience for hearing aid users. Our research group previously developed a human-in-the-loop machine learning approach to personalize the compression function of hearing aids. A deep neural network was trained based on a user’s preference to predict the reward as part of a reinforcement learning framework. This approach was not deployable online in the field as the training of the deep neural network needed to be conducted offline. In this work, an alternative human-in-the-loop machine learning approach is developed which is deployable online. In this approach, first a user’s preference model is set up via paired comparison feedbacks received from the user for audio signals compressed by different compression settings. Then, the maximum likelihood inverse reinforcement learning method is used to find the personalized compression settings based on the user’s preference model. A comparison between the outcome of the personalized settings and the prescriptive compression settings of DSLv5 was conducted on 10 hearing-impaired subjects. The results showed that the personalized settings were preferred about 10 times more than the prescriptive settings. A word recognition experiment was also carried out which showed that not only the personalization approach did not have any adverse impact on speech understanding in noisy conditions, but also it improved speech understanding. Conducting compression in a personalized way would lead to higher user satisfaction with their hearing aids.

**SP506 Optimizing Hearing Aids for Music Perception in Noise Using Subjective and Objective Measures**

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The perception of music is significantly compromised in the individuals using hearing aids (HA), which becomes further degraded in the presence of noise. Although HAs are heavily optimized for speech intelligibility in noise, they often are not well optimized for music perception, especially in the presence of noise. This study investigated the effect of signal-to-noise ratios (SNR) on music processed through HAs in two programs (1-Speech and 2-Music) using subjective and objective measures. The HA (fitted to NAL-NL2) processed music stimuli in the two programs at 5 SNRs (0, +5, +10, +15 and +20) were rated perceptually by 20 normal-hearing participants on a visual analogue scale of ‘Adjective descriptors of Music’ adapted from ‘Iowa Musical Background Questionnaire’, while the quality of music was objectively analyzed using a music quality index (HAAQI) using MATLAB code. Results showed that the music perception was better for higher SNRs (+15 and +20) than lower SNRs (0, +5 and +10) in both subjective and objective measures for both the programs. The effect of SNRs was more pronounced in the music program. These results emphasize the need for subjective and objective validation of HA programs for music listening in clinical setups.

**SP601 Efficacy of Individual Computer-Based Auditory Training for People with Hearing loss: An Updated Systematic Review and Meta-Analysis**

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Auditory training can be described as a process of training the brain to listen through active engagement with sounds. It has been shown to result in improved speech perception over the course of an adult’s lifespan (Wright and Zhang, 2009). As such, auditory training can be offered prior to, or alongside hearing devices to help improve listening for people with
hearing loss. Because auditory training interventions can be delivered via computers and the internet, they offer low-cost forms of self-management support that can be individually tailored and conveniently accessed by people with hearing loss. However, for auditory training to be considered effective, it should result in sustained generalised improvements that extend beyond trained tasks, to benefit everyday listening abilities.

In 2013, a systematic review of the efficacy of computer-based auditory training interventions for adults with hearing loss identified 13 eligible studies and concluded that the published evidence was not robust and therefore could not be reliably used to guide intervention decisions at that time. The authors identified a need for high-quality evidence to further examine the efficacy of computer-based auditory training in this population.

Here we report an in-progress update to the systematic review with the addition of meta-analyses. Registered via Prospero (CRD42017076817), this updated review addresses the primary research question “Does evidence exist to support improvements in trained and untrained measures of speech perception, cognition, and self-reported communication or quality of life, as a result of individual computer-based or internet-based auditory training in adults with hearing loss (with or without hearing aids or cochlear implants)?” Twelve electronic bibliometric databases and trial registries were searched during December 2021. Randomised controlled trials, non-randomised controlled trials, cohort studies, and repeated measures designs were all considered for inclusion.

To date, we have identified 51 studies eligible for inclusion in the updated review and data extraction is ongoing. Meta-analyses will examine the magnitude of improvement (learning) for trained tasks and transfer of learning to untrained outcomes for the domains of speech perception, cognition, and self-reported hearing across different training stimuli (speech/music) and participant device use (none/hearing aids/cochlear implants or bimodal). A narrative synthesis will describe key study characteristics and additional review findings. This review will provide a comprehensive synthesis of the robustness and estimates of efficacy of computer-based auditory training for adults with hearing loss, to help guide future evidence-based hearing healthcare and research.

**SP602 Hearing Aids Combined with Educational Counseling versus Educational Counseling Alone**

**Objective:** Hearing aids (HAs) and educational counseling (EC) are commonly recommended for tinnitus management in patients with hearing loss (HL) and persistent, bothersome tinnitus. However, the relevant studies are limited, and the effect of two treatments or the combination is ambiguous. This study aimed to investigate whether the combined use of HAs and EC is more effective than EC alone on tinnitus relief.

**Methods:** A total of 72 adults with chronic, bothersome tinnitus and co-existing sensorineural HL were included. After receiving EC and HAs prescription, 21 participants self-selected to purchase HAs and receive EC (i.e., HA+EC group), while the remaining 51 refused to use HAs and only received EC (i.e., EC group). They were asked to return for follow-up at 1 and 3 months after treatment initiation, and were encouraged to follow up frequently. At baseline, they all completed audiologic tests, tinnitus pitch matching, Tinnitus Handicap Inventory (THI), Tinnitus Evaluation Questionnaire (TEQ), Visual Analogue Scale (VAS) for loudness, Self-rating Depression Scale (SDS), and Self-rating Anxiety Scale (SAS). At each follow-up visit, both groups received EC sessions and completed THI, TEQ and VAS. Besides, International Outcome Inventory for Hearing Aids (IOI-HA) was administrated in the HA+EC group at 3-month follow-up. The primary outcome measure was THI, and tinnitus relief was defined as a 20-point or more reduction in THI. [This trial has been registered with Chinese Clinical Trial Registry (ChiCTR1900022624).]

**Results:** Demographic variables, status of insomnia, depression, and anxiety, degree of HL, tinnitus pitch, initial scores for THI, TEQ and VAS were homogenous across two groups. THI, TEQ and VAS scores decreased significantly after treatments (Wilcoxon signed rank test, all p < 0.05), and both groups yielded a similar trend of reduction. Differences in time-to-event curves (i.e., cumulative tinnitus relief) between groups were insignificant (Log-rank test, p = 0.63). Moreover, there was no significant difference in the incidence of tinnitus relief between the two groups (61.90% and 47.06% respectively, p = 0.25). Participants in HA+EC group were generally satisfied with HAs based on the mean score of 25.08 on IOI-HA.
**Conclusion:** Receiving HA+EC and EC were equally effective for tinnitus management. There was insufficient evidence to support the superiority of the combined use of HAs and EC for tinnitus over EC with no device.

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**Wednesday, August 10**

7:00 a.m. – 8:30 a.m.  Breakfast

8:30 a.m. – 9:10 a.m.

**IHAS ORAL SESSION 4**

**S4-1. A Neural-Network Framework for the Design of Individualised Hearing-Loss Compensation**  
*Fotios Drakopoulos*, Sarah Verhulst  
1Ghent University

Even though the human auditory system is known to be complex and highly non-linear, hearing aids (HAs) still rely on simplified descriptions of the auditory system or on sensorineural hearing loss (SNHL) estimations, such as hearing thresholds or perceived loudness, to yield optimal acoustic amplification to HA users. Standard amplification strategies succeed in restoring inaudibility of faint sounds, but sometimes fall short of providing precise treatment outcomes for complex sensorineural deficits such as presbycusis or cochlear synaptopathy (CS). To address this challenge, we adopt a deep-neural-network (DNN) version of a biophysically realistic model of human auditory processing (CoNNear). CoNNear accurately simulates individual SNHL and comprises a fully differentiable description that can be used to design individualised HA strategies from the ground up. DNN-based audio-processing models (DNN-HA) can be trained that can optimally process sound to restore hearing in impaired auditory peripheries. In this study, we evaluate the restoration capabilities of our framework using simulated auditory-nerve (AN) responses of normal and impaired auditory peripheries. We compare different loss functions designed to optimally compensate for a mixed outer-hair-cell loss and CS impairment, each time focussing on the enhancement of different auditory features. After evaluating which trained DNN-HA model yields the best restoration outcomes on simulated AN responses and speech intelligibility, we applied the same training procedure to two milder hearing loss profiles, separately for OHC loss and CS. Our results show that a simulated restoration of AN population responses was possible in all cases, with OHC loss proving easier to compensate than CS. Several objective metrics were considered from the literature to estimate perceptual benefits after processing, with the results holding promise for improved understanding of speech-in-noise processing for hearing-impaired listeners. Since our framework can be tuned to the hearing-loss profiles of individual listeners, we enter an era where truly individualised and DNN-based restoration strategies can be developed and be tested experimentally. [Work supported by European Research Council ERC-StG-678120 (RobSpear) and FWO grant G063821N Machine Hearing 2.0]

**S4-2. More is More: Physiological Markers of Successful Effort**  
*Andreea Micula*, Jerker Rönnberg, Elaine Hoi Ning Ng

IHAS August 9-10, 2022 32  
IHCON August 10-14, 2022
Working memory (WM) has been established as one of the most essential cognitive functions that support speech understanding, especially in sub-optimal listening conditions. Several studies have combined behavioral and physiological measures to explore WM resource allocation in challenging listening situations. The outcomes of two studies that have combined an auditory recall test, the Sentence-Final Word Identification and Recall (SWIR) test, and pupillometry in hearing aid users are summarized. The SWIR test has been used to quantify the WM resource allocation trade-off between speech processing and speech encoding for later recall. Pupillometry captures changes in pupillary dilation over the course of a task and several indices can be extracted. The overall aim of these studies was to investigate momentary WM resource allocation, as well as WM resource allocation over time using different pupillary responses. The baseline pupillary dilation (BPD) reflects the overall brain state and its effects on arousal or attention over relatively long time intervals. The task-evoked peak pupillary dilation (PPD) is a transient response triggered by specific stimuli and is commonly used to quantify the allocation of processing resources to a task.

Both studies included adults with mild to moderately severe symmetrical sensorineural hearing loss. The participants were provided with frequency-specific amplification based on the individual audiogram. The SWIR test was administered in competing speech at an individual signal-to-noise ratio estimated to result in 95% correct word recognition. The task was to listen to lists of sentences, repeat the last word immediately after each sentence and when the list is finished, recall as many of the repeated words as possible.

An increase in BPD over the course of a list was associated with better overall recall performance. Higher PPD measured while listening to a sentence and preparing to encode a target word was linked to significantly higher likelihood of subsequently recalling that word. The PPD is considered to reflect the momentary intensity of attention devoted during speech encoding, while the BPD is considered to reflect the accumulation of WM resources allocated over time to maintaining items for later recall. The concept of “successful effort” is proposed to describe the deliberate allocation of additional WM resources, indexed by larger pupillary responses, to achieve a better performance level. Consequently, allocating more resources or exerting more effort is considered successful, since it leads to a favorable outcome.

9:10 a.m. – 9:30 a.m. Coffee Break
9:30 a.m. – 11:10 a.m. IHAS Poster Session (Continued)
11:10 a.m. – 12:10 p.m. Mentoring Session 2A
   Industry: The job (job responsibilities, intellectual property, work/life balance, corporate culture)
11:10 a.m. – 12:10 p.m. Mentoring Session 2B
   Academia: The job (building collaborations, diversifying research, work/life balance, service)
12:10 p.m. – 1:30 p.m. Lunch
1:30 p.m. – 5:00 p.m. IHCON Arrivals / Leisure
IHCON PROGRAM

Wednesday, August 10

1:30 p.m. – 5:00 p.m.    IHCON Arrivals / Registration
5:00 p.m. – 6:00 p.m.    Welcome Reception
6:00 p.m. – 7:30 p.m.    Dinner
7:30 p.m. – 7:45 p.m.    Opening Remarks (Sunil Puria)

SESSION A1. Keynote Address
Session Chair: Louise Hickson

7:45 p.m.

A1-1. Avenues for Improvement in Hearing Aids
Brian Moore*
1University of Cambridge

Despite advances in signal processing in hearing aids over the past 20-30 years, hearing aids are still far from restoring “normal” hearing. This partly reflects limitations of impaired auditory systems, such as reduced frequency selectivity and reduced sensitivity to temporal fine structure, but also reflects limitations in the hearing aids.

One potential avenue for improvements is via “personalized medicine”. The idea is to characterize the hearing problems of a person in detail, using measures beyond the audiogram, and then to tailor the signal processing in hearing aids to compensate for the specific deficits of the individual. However, such approaches have achieved only limited success, except perhaps for specific types of hearing problems, such as “dead regions” in the cochlea. I will review some attempts to apply personalized medicine and consider why they have had only limited success.

Another avenue for improvements is via the application of machine learning, specifically via deep neural networks (DNNs). This area is starting to show promise, and DNNs are already used in some commercial hearing aids. The potentials and limitations of such approaches will be discussed.

A final avenue for improvement is to get the “basics” right, something that in principle should be readily achievable. Current common problems are:

1. The gains achieved on real ears are often substantially different from those programmed into the manufacturer’s software, even when averaged over many test ears. A common problem is a failure to meet target gains for frequencies above about 3 kHz.

2. The compression ratios obtained on real ears are often substantially different from (usually below) those programmed into the manufacturer’s software. As a result, soft sounds remain inaudible and strong sounds are too loud.

3. Despite claims of wide bandwidth, many hearing aids are unable to meet the fitting targets of common fitting methods for frequencies above about 4 kHz.
(4) The output of many hearing aids falls off markedly for frequencies below a few hundred Hz. This reduces the sound quality of music.

(5) Systems for acoustic feedback reduction/cancellation can have serious deleterious effects when listening to music.

More subtle problems arise as side effects of the signal processing in hearing aids. Processing such as multi-channel amplitude compression, noise reduction, and adaptive directionality changes the amplitude modulation patterns of the signal and this can have adverse effects on speech intelligibility and sound quality. Manufacturers need to consider how these forms of processing interact.

8:30 p.m. – 9:00 p.m. Discussion
9:00 p.m. – 10:00 p.m. IHAS Posters and Social

Thursday, August 11

7:00 a.m. – 8:00 a.m. Breakfast
8:00 a.m. – 10:00 a.m.

SESSION B1. Health Psychology Applications for Hearing Aids
Session Chair: Ben Hornsby

B1-1. Applying the Science of Health Behaviour Change to Hearing Health research: From Theory to Practice
Helen Henshaw*

*NIHR Nottingham Biomedical Research Centre, Hearing Sciences, University of Nottingham, UK

Theories and models to predict health-related behaviours are powerful tools in health research. This is particularly true for long-term conditions without cure (such as the vast majority of hearing loss cases), whereby methods to understand, predict and conceptualise patients’ attitudes and behaviours are crucial to inform interventions to support the management and self-management of health. New Medial Research Council (MRC) guidance emphasises that the development of complex interventions should be based on both research evidence and theory of the problem to be addressed. Several common models of health behaviour have been applied to hearing health research in recent years, including the health belief model (HBM), transtheoretical model (TTM), self-regulatory model (SRM) the self-determination model (SDM). More recently, the Behaviour Change Wheel (BCW) has gained significant momentum as a holistic framework for the development of behaviour change interventions. It features a ‘behaviour system’ at its hub (the ‘COM-B’ system), outlining three interacting factors that need to be present for any behaviour to occur: capability, opportunity, and motivation.

Funded by the National Institute for Health and Care Research (NIHR), a novel application of the BCW and COM-B system has been the development of a behaviour change intervention to improve the use of hearing aids by new adult audiology patients within the UK National Health Service (NHS). Semi-structured interviews with a diverse range of adults with hearing loss...
identified multiple barriers and facilitators to hearing aid use. A detailed behavioural analysis informed the selection of Behaviour Change Techniques (BCTs) to address barriers and promote facilitators. In-depth discussions with the target population were used alongside published evidence and expert stakeholder opinion, to generate ‘guiding principles’ for how intervention the components should be framed and delivered. A logic model was developed to explain how the intervention is expected to work, and how each intervention component was understood to contribute to the target outcome of improved hearing aid use.

Intervention development is currently ongoing. ‘Think Aloud’ interviews with patients will be used to iteratively review and refine the content throughout development, offering a deep understanding of the psychosocial context of the target population and their views of the intervention. The final stage of this research will involve examining the feasibility of the intervention itself, and the feasibility of evaluating the effectiveness and cost-effectiveness of the intervention for patients and the NHS within a multicentre randomised controlled trial (RCT).

B1-2. Effects of Patient Traits on the Relationship between Hearing Handicap and Readiness to Pursue Hearing Help
Lipika Sarangi*1, Jani Johnson1
1The University of Memphis

Hearing aid (HA) success is commonly assessed in the later stages of an individual’s hearing health journey in terms of adoption, use, benefit, and satisfaction in daily living. For those in earlier stages, readiness to pursue hearing healthcare could be an indicator of progress towards positive hearing health outcomes. We explored whether patient traits that are predictors of later indicators of HA success also predict readiness to pursue hearing intervention. We further explored the moderating/mediating effects of modifiable patient traits on the relationship between hearing difficulties and readiness to pursue intervention using moderation/mediation analyses.

Sixty-two adults with self-reported hearing difficulties and no prior experience with HAs participated in this descriptive study. Perceived hearing-related handicap was assessed as the primary predictor and readiness to change as the outcome variable. Hearing aid self-efficacy (HASE), personality traits, and affective states were assessed in general and in hearing-related situations. Moderation/mediation analyses were performed to identify significant predictors of readiness and to test the effects of HASE and affective states on the relationship between hearing handicap and readiness to pursue help.

When HASE was considered in the model, individuals with greater hearing handicap, high HASE, more agreeable personality, and those who had hearing loss for a shorter duration were more ready to pursue help (F(9,52)=7.83, p<.0001). Moderation analyses demonstrated that the relationship between hearing handicap and readiness didn’t change as a function of HASE, when controlling for personality and duration of hearing loss. When affective states were considered, individuals with greater hearing handicap, low Conscientiousness personality, and those who had hearing loss for a shorter duration were more ready to pursue help (F(14,48)=6.79, p<.0001). Mediation analyses demonstrated that the relationship between hearing handicap and readiness could not be explained by their relationships with affective states, when controlling for personality and duration of hearing loss.

This study confirms that HASE, certain personality traits, and duration of hearing loss are important variables that motivate people toward or away from becoming successful in their hearing health journey. Neither HASE and affective states significantly impact the relationship between perceived hearing handicap and readiness to pursue audiologic intervention. Future research should explore the associations among these patient traits and their impact on success at different stages of the hearing health journey to determine if an assessment of these factors is warranted.
B1-3. Co-Production of Text Message Content to Help Nhs Audiology Patients when They Are First Prescribed Hearing Aids

Emma Broome*, Katrina Copping, Sian Calvert, Helen Henshaw
1NIHR Nottingham Biomedical Research Centre, Hearing Sciences, University of Nottingham, UK

Introduction: In the United Kingdom, approximately 12 million people have permanent hearing loss, and 355,000 adults are fitted with hearing aids each year via the National Health Service (NHS). However, the non-use and infrequent use of NHS-prescribed hearing aids is high (18-31%). There can be many barriers to overcome when hearing aids are first prescribed and patients tell us that they would benefit from additional support.

The NHS-approved text-message service ‘Florence’ has been shown to help patients self-manage many different long-term conditions. However, it has not yet been used by patients with hearing loss. We are working in partnership with patients to coproduce a Florence intervention protocol for new hearing aid users, using qualitative participatory techniques. The intervention protocol, designed in accordance with the Medical Research Council guidance for the development and evaluation of complex interventions, will use health behaviour theory to improve patients’ capability, opportunity and motivation, when they are first prescribed hearing aid(s). Florence will deliver text-messages to individuals’ mobile telephones, with the aim of improving hearing aid use and benefit. Text-message content, drawing upon established behaviour change techniques, will be coproduced with patients in order to support the adaptation to and use of new hearing aids by addressing key barriers.

Objective: This study aims to create and refine theory-driven text-message content for a Florence intervention protocol.

Design: This work-in-progress qualitative study comprises a two-stage process to intervention development:
1. Maximum variation sampling will be used to recruit people with hearing loss and their communication partners for three workshops. Participants will be invited to coproduce and refine text-message content using PhotoVoice, a qualitative participatory method that documents and reflects reality. A further workshop will be held with audiologists to ensure that intervention content is aligned to clinical care.
2. To assess the acceptability of Florence to patients we will pilot a prototype Florence protocol with approximately five participants and gather in-depth feedback on the text-message content, language and framing, via semi-structured interviews. The protocol will be iteratively refined to ensure that it is accessible, engaging and aligned to patient needs.

Conclusion: This approach to intervention development will ensure that the Florence intervention addresses issues that are important to new hearing aid users. The findings of this study will ensure that the Florence protocol is fully functional and can be successfully implemented within future studies of intervention effectiveness and cost-effectiveness to improve outcomes for patients and the health service.

B1-4. To Tell or Not to Tell: The Stigma Experiences of Adults with Hearing Loss and Their Families

Louise Hickson*, Katie Ekberg, Barbra Timmer, Nerina Scarinci, Monique Waite, Carly Meyer, Mansoureh Nickbakht
1The University of Queensland, 2The University of Queensland and Sonova, 3University College London

In this multidisciplinary research project using a sequential, exploratory mixed methods design, we investigated how stigma is experienced by adults with hearing loss and their families, how they manage it in everyday life, and how these experiences relate to the decision to try hearing aids and to wear them in the long-term. In the first phase of the research, data from 20 dyads of adults with hearing loss (mean age 69.4 years, 17 female, 11 hearing aid wearers) and their
nominated family members (mean age 63.9 years, 16 female) was collected. Methods of assessment were 1) self-report questionnaires, 2) conversation analysis of video-recorded conversations in daily life, 3) Ecological Momentary Assessment, and 4) qualitative semi-structured interviews with adults with hearing loss and family members, individually and together. Additionally, in phase 1 we interviewed 25 hearing care professionals (mean clinical experience 10.7 years, 19 female) about their understanding of how stigma affects adult clients and their family members.

The findings from phase 1 were then mapped to the Major and O’Brien (2005) identity-threat model of stigma and revealed that stigma is a complex, social process experienced differently by adults with hearing loss and their family members. Stigma associated with hearing loss, hearing aids and ageing were evident and resulted in delayed hearing help-seeking, as well as reduced hearing aid uptake and use.

In the second phase, we explored the findings further with an online survey of 331 adults with hearing loss (mean age 66.2 years, 149 female) and 313 family members of adults with hearing loss (mean age 48.8 years, 247 female) in Australia, the UK and US. Stigma related to both hearing loss and hearing aids was evident and both hearing loss and hearing aids were strongly associated with ageing. While both groups were generally positive about modern hearing aids, family members of adults with hearing loss were more positive about hearing aids than adults with hearing loss. The topic of humour emerged as a theme in both phases of this study and should be explored further.

Reference:

B1-5. Low Levels of Hearing Stigma in Uk Older Adults Explain Null Effects of a Brief Psychological Intervention to Reduce Stigma in Hearing Health
Christopher Armitage1, Piers Dawes*2, Kevin Munro1
1University of Manchester, 2University of Queensland

Background: Although hearing loss is the second-most prevalent and fifth-most disabling health condition, only 9.25% of UK adults hearing loss own or use a hearing aid. Stigma is considered one of the greatest challenges facing people with hearing loss, undermining hearing health-seeking behaviours, including telling professionals about difficulties with hearing. The aim of the present research is, for the first time, to trial the effects of an intervention based on psychological theory to reduce stigma associated with hearing loss and/or use of hearing aids and improve hearing health-seeking.

Methods: Adults aged 60+ were invited to take part in an online questionnaire distributed by YouGov, a survey panel company in January 2021. Hearing health-seeking was assessed at baseline and 6-month follow-up using yes-no questions about whether people have hearing difficulties, have seen an audiologist and own/use a hearing aid. Participants were randomized to control or self-affirmation intervention. The intervention group were asked to complete a self-affirming implementation intention, for example “If I feel threatened or anxious, then I will think about things that are important to me”. [The trial was pre-registered with the US Clinical Trials Registry: https://ichgcp.net/clinical-trials-registry/NCT04680845.]

Results: The sample (n=3012) was representative of older adults in the UK; most were white (97.8%) and 53.3% were women. Mean age was 70.9 years. 43.3% reported hearing difficulties but just 12.5% reported wearing hearing aids most of the time. Public stigma of hearing loss was modest, with most people scoring at or below the midpoint of the 1-7—point scales. Self-stigma was lower still, with most people scoring below the midpoint of the scales related both to hearing loss and hearing aids. Of the people who reported hearing difficulties (n = 1304), 75.5% reported low levels of self-stigma regarding hearing loss and 72.6% reported low levels of self-stigma regarding hearing aids. There were no statistically significant condition x time interactions.
interactions between condition and time and no statistically significant main effects of condition or time on stigma or hearing health-seeking behaviours.

**Conclusion:** Low levels of stigma associated with hearing loss and/or hearing aids in the UK precludes the use of stigma-reducing interventions in the general population. Such stigma-reducing approaches might be more successfully targeted at clinical settings that might make feelings of stigma more salient whereas public health-level hearing health interventions might need to look beyond stigma as a driver of hearing health-seeking behaviours.

10:00 a.m. – 11:30 a.m. Coffee Break and…

**POSTER SESSION BP**

**BP101 A Protocol for Partner Engagement in Audioligic Care**

*Alix Klang*, Steven Gianakas, Timothy Beechey, Erin O'Neill, Peggy Nelson

1University of Minnesota, 2University of Nottingham, 3GN Advanced Science

Hearing loss is a shared experience for clients and families. In older adults it is often first identified by those closest to them, subsequent to communication challenges affecting the family. However, current audioligic service delivery models take a person-centered approach; here, professionals’ time is often spent focused on devices and audibility alone. We hypothesize that client outcomes can be improved through greater focus on collaborative partner engagement. A partnership-centered audiology care model could lead to improved outcomes by facilitating clients’ and partners’ mutual understanding of expectations, communication, and empathy.

Previously, we collected data from 14 people with hearing loss (ages 60 to 80 years) and their partners. Each couple completed questionnaires concerning the impact of the client’s hearing loss. Results were evaluated using an ordinal agreement statistic to measure the degree of client/partner mismatch. In addition, open-ended responses were evaluated using thematic analysis. Preliminary results revealed a complex mismatch between client and partner perceptions of the impact of hearing loss. Patterns of mismatch included: (1) clients indicated greater impact of hearing loss than acknowledged by partners; (2) partners revealed greater impact of hearing loss than indicated by clients; or (3) a mix of both. Open-ended responses were similarly variable across couples, and were important for revealing issues related to third-party disability, including the partners’ apprehension while communicating, their feeling responsible to assist their partner with hearing loss, and their lamenting the loss of social engagement. Clients and partners both provided important unique insights that could aid in device success.

Currently, we are collecting new data, administering client- and partner-based questionnaires prior to and following a laboratory-based shared communication experience. The laboratory experience includes recreations of realistic conversations, simulation of hearing loss, and ratings of listening effort. Results from the questionnaires pre- and post-session will again be evaluated through measures of agreement and thematic analysis, and will reveal initial and final mismatched perceptions. We will discuss results with the partners, and follow up with questionnaires in subsequent months.

Building on these findings we propose using areas of agreement and disagreement to facilitate discussions and allow for collaborative understanding of expectations, communication challenges, third-party disability, and empathy during hearing device fitting and follow-up. We further plan to investigate the most impactful ways to minimize client and partner mismatches, postulating that modifying expectations of both the client and partner could lead to improvements in hearing aid use, satisfaction, and quality of life.
BP102 Aging and Social Perceptions of Hearing Aid Users
Julie Beadle¹, Lorienne Jenstad¹, Diana Cochrane¹, Jeff Small¹
¹The University of British Columbia

Purpose: Previous research suggests that explicit (i.e., conscious) attitudes held towards hearing aid users have improved over the past 40 years. However, implicit (i.e., unconscious) attitudes held towards hearing aid users are not well understood. As implicit attitudes could influence perceptions, communication behaviour, and hearing aid uptake, this study investigated older and younger adults’ implicit and explicit attitudes towards older and younger adult hearing aid users. The study also evaluated whether implicit attitudes are related to participants’ self-reported levels of hearing disability.

Method: Thirty older adults (M age = 70 years, SD = 4.38) and 30 younger adults (M age = 23 years, SD = 3.01) who reported not having hearing aids or a diagnosis of hearing impairment participated in this online study. All participants passed a screening test for cognitive impairment. Implicit attitudes were measured using two Implicit Association Tests: one using images of older adults (OA-IAT) and one using images of younger adults (YA-IAT), either wearing or not wearing in-the-ear style hearing aids. To measure explicit attitudes, participants rated age, attractiveness, and intelligence of older and younger adults pictured with or without in-the-ear hearing aids. Self-reported hearing disability was measured using the Hearing Handicap Inventory for the Elderly/Adults (HHIE/A).

Results: Both older and younger participants showed negative implicit attitudes towards older and younger adults who wear hearing aids (compared to those who do not); however, older adult participants’ negative implicit attitudes were more strongly negative, and less variable, than younger adult participants’ implicit attitudes. Ratings of age, intelligence, or attractiveness did not vary in relation to the presence or absence of hearing aids for either age group. Images of older adults were rated overall as less attractive than images of younger adults. Spearman’s correlation coefficients indicated that older adults with higher HHIE scores (i.e., poorer hearing) showed weaker negative implicit attitudes towards older adults who wear hearing aids (r = -.41, p < .05).

Conclusions: Although explicit attitude measures suggest that attitudes towards individuals who wear hearing aids are neutral, older adults show strong negative implicit attitudes towards both older and younger hearing aid users. These negative implicit attitudes may contribute to the low hearing aid uptake rates currently observed in the senior population. Future research should investigate how implicit attitudes and hearing disability interact to predict older adults’ intentions to wear a hearing aid and actual hearing aid uptake.

BP103 Hearing Aid Users’ Emotional Responses to Sounds and Pictures
Erin Picou¹, Todd Ricketts¹, Benjamin Hornsby¹
¹Vanderbilt University Medical Center

Adults with hearing loss demonstrate a reduced range of emotional responses to non-speech sounds; their ratings of valence to pleasant and unpleasant sounds are less extreme than those of their peers with no hearing loss. Yet, it is unclear if these differences extend to visual stimuli. If the noted differences in emotion perception extend to pictures, it would suggest the previously observed group differences to non-speech sounds are not fully attributable to hearing loss. One purpose of this study was to evaluate whether the aforementioned group differences are evident in response to pictures, in addition to sounds. A second purpose was to evaluate whether personal hearing aids can expand the range of valence ratings in response to sounds. Previous work in this area has been limited to the fitting of research hearing aids; less is known about adults’ emotional responses to sounds when listening with and without their own hearing aids. In this study, ratings of valence and arousal in response to non-speech sounds or pictures were provided by two groups of adults: 1) adults no hearing loss (n=12, mean age = 65.8 years) and 2) adults with hearing loss (n = 15, mean age = 66.0 years). Adults with hearing loss rated sounds with and without their personal amplification. Analysis with linear mixed effects models revealed listeners with hearing loss demonstrate ratings of valence that are less extreme (less pleasant and less unpleasant) than their peers’ ratings, consistent with previous literature. In response to pictures, however, group differences were only evident for unpleasant stimuli. These data demonstrate that the group differences in ratings of pleasant sounds observed in prior literature can be attributable to hearing loss, whereas differences in ratings of unpleasant sounds might be attributable to factors unrelated to hearing loss, such as gender or personality. Finally, ratings were the same with and without use of their personal hearing aids, demonstrating that non-significant benefits of hearing aids...
In recent years the psychosocial aspects of living with hearing loss have gained increasing attention. Studied to a lesser extent is the influence of wearing hearing aids to counter these consequences. The aim of this study was to evaluate whether first-time hearing aid use affects self-reported social participation. We measured self-rated social participation in first-time and experienced hearing aid users with a repeated measure, longitudinal design using a Danish translation of the Social Participation Restrictions Questionnaire (SPaRQ) and a Danish translation of the Hearing Handicap Inventory for the Elderly/Adults (HHIE/HHIA). The HHIE/HHIA was included to assess the influence of the patient’s self-perceived handicap on changes in their social participation. Ninety-five hearing aid users were recruited, and approximately half were first-time hearing-aid users (intervention group), and half were experienced hearing-aid users (control group). Study data were collected and managed using REDCap electronic data tools hosted at Region Hovedstaden, wherein participants were asked to fill out the SPaRQ and HHIE/HHIA questionnaires five times over the course of 18 weeks. For the intervention group, there were two data collection points approximately one week apart directly before receiving a hearing aid, two data collections points approximately one week apart 6-8 weeks after receiving the hearing aid, and one follow-up data collection point three months after receiving the hearing aid. For the control group, a similar spacing between assessments was made. While data collection is still ongoing for the longitudinal aspect of the investigation, in this sub-study we assess both the test-retest reliability of the Danish translation of SPaRQ and HHIE/HHIA, as well as compare self-rated social participation and hearing handicap at baseline between the intervention and control groups. This assessment is an important component of the field testing we have conducted during the translation and adaptation of SPaRQ and HHIE/HHIA to Danish. Furthermore, it will inform the forthcoming investigation of the relationship between hearing aid use and self-rated social participation over time.

**BP104 Investigating the Effect of First-Time Hearing Aid Use on Self-Rated Social Participation**

Filip Roenne*, Signe Wischmann, Els Walravens, David Harbo Jordell, Sarah Gotowiec, Abigail Anne Kressner

WS Audiology, Rigshospitalet, Rigshospitalet and Technical University of Denmark

Hearing loss is a pervasive condition that can substantially affect many aspects of quality-of-life, including social participation, employment, and mental wellbeing. Hearing aids are the primary intervention for hearing loss, yet many adults who would benefit from hearing aids do not use them. This research aimed to explore the perceptions and experiences of adults with hearing loss regarding barriers and facilitators to successful hearing aid use.

Semi-structured interviews were conducted with adults with hearing loss who had been fitted with a hearing aid(s). The research team included a patient research partner (an adult with hearing loss), who was involved in the design and conduct of this study, including co-developing the interview schedule and contributing to the data analysis. Twenty-four participants were recruited via maximum variation sampling. There were 13 men and 11 women between 33-85 years of age. They varied in terms of educational attainment, audiometric profile, and hearing loss duration. The data were analysed inductively in accordance with an established reflexive thematic analysis procedure. Several techniques for enhancing qualitative rigour and trustworthiness were used, such as peer debriefing.

The analysis generated four main themes, each comprised of several sub-themes. Theme 1 (behaviour change and habit formation) described how perceived value, situational cues, and practical issues influence the initial behaviour change and formation of a hearing aid use habit. Theme 2 (stigma and social support) explored how perceptions about hearing aid users, peer support, and disclosure of hearing loss impact upon hearing aid use. Theme 3 (communication) reported that hearing aids and communication tactics
can facilitate active communication, though this depends on the listening context. Theme 4 (information required to succeed) described the importance of access to information and instruction, ongoing support, and an appropriate mode of information delivery. Furthermore, the four themes were underpinned by three key factors: (1) age (e.g., working age), (2) auditory lifestyle (e.g., living alone), and (3) confidence (e.g., willingness to disclose hearing loss). The results suggested that these factors influenced hearing aid experiences, especially the formation of a stable or automatic habit of hearing aid use.

This research provided an in-depth exploration of barriers and facilitators to successful hearing aid use, including the broader context and factors surrounding them. The findings will feed into the ongoing planning and development of a behavioural intervention to improve successful hearing aid use for adults with hearing loss that is grounded in health behaviour change theory.

**BP201 Evaluation of Speech Intelligibility during the Tracking of Noise Tolerance**

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The amount of background noise that an individual is willing to accept while listening to speech is predictive of his or her hearing aid satisfaction. When listeners can freely decide the reason why he or she is willing to accept the selected amount of background noise, different people are likely using different criteria to make this judgment (Mackersie et al 2020).

The Tracking of Noise Tolerance (TNT) test was designed to measure how much noise an individual can tolerate while still understanding 90% of speech. Results of a pilot study showed that the audibility-based estimates of speech intelligibility performance during the TNT tracking ranged from 40% to >90%. This suggests that the subjective intelligibility estimates that listeners use during the noise tracking task may vary across individuals. People who have a lower subjective intelligibility criterion would likely accept a higher noise level. Furthermore, it would also be reasonable to suspect that people who have a lower subjective speech intelligibility criterion will be more accepting of their hearing aids in noise than those who have a higher criterion.

In this study we investigated the speech intelligibility criterion used by different listeners during the noise tracking task. Seventeen listeners with hearing impairment participated. Noise tolerance was measured using the TNT unaided at 75 and 82 dB SPL speech levels with noise presented from 0° and 180°. We examined the estimated speech intelligibility during the noise tracking using objective P-I functions for TNT materials created for each listener individually, as well as with audibility-based estimates. Data collection is ongoing.

Listener’s subjective criterion for speech intelligibility in noise could be an important piece of profiling information to (1) identify challenging patients, (2) select the necessary hearing aid features to meet different criteria needs, and (3) provide internal criteria for hearing aid algorithms to ensure that intelligibility will not fall below the listeners individual criterion.

**BP202 More is More: Physiological Markers of Successful Effort**

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Working memory (WM) has been established as one of the most essential cognitive functions that support speech understanding, especially in sub-optimal listening conditions. Several studies have combined behavioral and physiological measures to explore WM resource allocation in challenging listening situations. The outcomes of two studies that have combined an auditory recall test, the Sentence-Final Word Identification and Recall (SWIR) test, and pupillometry in hearing aid users are summarized. The SWIR test has been used to quantify the WM resource allocation trade-off between speech processing and speech encoding for later recall. Pupillometry captures changes in pupillary dilation over the course of a task and several indices can be extracted. The overall aim of these studies was to investigate momentary WM resource allocation, as well as WM resource allocation over time using different pupillary responses. The baseline pupillary dilation (BPD) reflects the overall brain state and its effects on arousal or attention over relatively long time intervals. The task-evoked peak pupillary dilation (PPD) is a transient response triggered by specific stimuli and is commonly used to quantify the allocation of processing resources to a task.

Both studies included adults with mild to moderately severe symmetrical sensorineural hearing loss. The participants were provided with frequency-specific amplification based on the individual audiogram. The SWIR test was administered in competing speech at an individual signal-to-noise ratio estimated to result in 95% correct word recognition. The task was to lis-
ten to lists of sentences, repeat the last word immediately after each sentence and when the list is finished, recall as many of the repeated words as possible.

An increase in BPD over the course of a list was associated with better overall recall performance. Higher PPD measured while listening to a sentence and preparing to encode a target word was linked to significantly higher likelihood of subsequently recalling that word. The PPD is considered to reflect the momentary intensity of attention devoted during speech encoding, while the BPD is considered to reflect the accumulation of WM resources allocated over time to maintaining items for later recall. The concept of “successful effort” is proposed to describe the deliberate allocation of additional WM resources, indexed by larger pupillary responses, to achieve a better performance level. Consequently, allocating more resources or exerting more effort is considered successful, since it leads to a favorable outcome.

BP203 Relationship between Beamformer Patterns, Compression, and Working Memory for Different Noise Configurations

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Previous research has shown that speech recognition with wide dynamic range compression (WDRC) is associated with individual working memory ability, especially in adverse listening conditions. Our recent work has found that combining stronger directional processing (beamformers) with WDRC in hearing aids may reduce the role of working memory for speech recognition in ideal spatial conditions when the target signal is presented at 00 and the noise is at 1800. However, under realistic spatial conditions, the noise may arrive from multiple locations, rendering the beamformer less effective if the interfering noise is more diffuse and falls outside the directional null. We need to understand the impact of directional processing on the relationship between working memory and speech recognition in realistic spatial conditions. In this project, we extend our work to include different beamformer patterns and multiple noise locations.

Listeners with bilateral mild to moderately severe sensorineural hearing loss repeat low-context sentences mixed with multi-talker babble, presented at a realistic signal-to-noise ratio (SNR) in different spatial configurations. The spatial configurations include two (+90, -90) or three (+90, -90, and 180) noise locations around the listener. Wearable hearing aids, customized to the listener’s hearing level, are used to present four combinations of signal processing available in two different devices: a binaural-cardioid (commercial device) or a bi-directional beamformer (open-source device) combined with fast- or slow-acting WDRC (matched for both devices). Other advanced hearing aid features are turned off. Individual working memory ability is measured using the reading span test. To account for potential differences between devices, speech recognition with omnidirectional processing is measured in quiet. In addition, a signal fidelity metric is used to quantify envelope distortion in the processed signal across experimental conditions with respect to a linearly-processed signal in quiet.

Preliminary acoustic analyses show more change in signal fidelity (re: omnidirectional) between noise configurations with the binaural-cardioid beamformer compared to the bi-directional beamformer. Feasibility data show better speech recognition with the binaural-cardioid versus bi-directional beamformer for noise presented from three locations, but comparable performance with both beamformers for noise from two locations. Performance across conditions is related to individual working memory and signal fidelity using a linear mixed-effects statistical model. The results of this study will guide clinical decisions for fitting directional processing based on individual cognitive abilities and listener environments. [Supported by NIH-K01DC018324].

BP204 Binaural Beamforming: Exploring End-User Performance, Perception and Preference

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The purpose of this investigation is to better understand the benefits of a binaural beamformer for hearing aid users through a series of tasks that explored 1) speech-in-noise performance, 2) spatial perception, and 3) overall preference between monaural and binaural beamformers. Nineteen experienced hearing aid users with moderate to moderately severe sensorineural hearing loss were fit binaurally with RIC hearing aids. The participants were given an ecological momentary assessment (EMA) app that logged environmental factors from the hearing aids (sound classification, signal-to-noise ratio, noise floor estimate, sound pressure level) while simultaneously recording the user’s interaction with a manual control of a binaural beamformer, as well as ratings of perceived benefit and satisfaction.
with using it. The participants were tasked with using the hearing aids with the app for a four week home-trial. In addition to the home-trial, they also completed two lab tasks: 1) a MUSHRA that asked the participants to rank a monaural beamformer and a binaural beamformer of two different strengths based on scene broadness and overall preference and 2) the English Matrix task which measured the participants SRT50 across the three beamformer conditions. Both the MUSHRA and the Matrix test were assessed in a soft level of noise, defined as 55 dB noise floor estimate (NFE) as read by the hearing aids, and a loud level of noise, defined as 63 dB NFE.

The study found that there were no significant differences between the beamformer conditions for the Matrix test, but the participants perceived the binaural beamformer as narrower than the monaural beamformer in the MUSHRA task. Additionally, there was a trending association between perceived narrowness and preference in the MUSHRA task that indicated that narrower beamformers were more preferred. The home trial with the EMA app found that when activation of a binaural beamformer provides an audible difference to the user, it also improves their satisfaction. When the participants reported no or only slight satisfaction with using the manual beamformer control, the beamformer strength was at its highest setting, indicating that hearing aid users find themselves in speech-in-noise environments for which a binaural beamformer is not beneficial.

In conclusion, this investigation provides evidence that suggests a binaural beamformer may be subjectively beneficial in terms of preference and spatial perception to hearing aid users at all levels of noise, even soft levels, despite no additional benefit for speech-in-noise performance.

**BP205 Acoustic- And Intelligibility-Based Patient Assessment on an Integrative Speech-In-Noise Test**

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Listeners with a hearing loss generally find speech-in-noise comprehension more effortful than normal hearing listeners (Kramer et al., 2006), especially at poorer signal-to-noise ratios (SNRs). However, it can be difficult to ascertain how much the burden of listening reflects impaired intelligibility, alone, and how much reflects individual characteristics of the listener, such as their cognitive capacity (Peelle, 2018) or ability to deploy adaptive coping strategies (Picou and Ricketts, 2014). This difficulty then extends to evaluating the efficacy of assistive technologies meant to ease the burden of speech processing or tailoring counseling and rehabilitation strategies to help the patient succeed in challenging listening environments.

Previously, we developed the Repeat-Recall Test (RRT) as a validation tool that integrates speech-in-noise testing at realistic signal-to-noise ratios (SNRs) with estimation of working memory capacity and ratings of listening effort and willingness to engage with communication in noise (Kuk et al., 2020, 2021; Slugocki et al., 2018). The RRT has been so far useful in demonstrating the efficacy of a variable speed compressor (Kuk et al., 2019) as well as directional microphone and noise reduction algorithms (Kuk et al., 2020). We have now further measured the performance of normal hearing listeners under a variety of noise configurations (Kuk et al., 2021).

This presentation explores our use of data collected from 63 normal hearing adult listeners across 3084 RRT trials to establish norms of expected performance (i.e., intelligibility and memory) and subjective ratings (i.e., listening effort and tolerable time) at each of the acoustic SNRs assessed in the RRT. We contrast these acoustic-based norms with ones based on the actual intelligibility measured from patients. Whereas the former defines the expected range of normal hearing RRT outcomes at each SNR, intelligibility-based norms re-define this range for memory, listening effort, and tolerable time outcomes at different levels of intelligibility. In this way, intelligibility-based norms accommodate the patient’s speech-in-noise deficits. Judging patient performance against an intelligibility-based reference can inform the clinician whether the patient has further issues with memory, effort, or tolerance for communication in noise than would be expected for a normal hearing listener in an environment where intelligibility was similarly degraded. We highlight where each choice of reference (i.e., acoustic- and intelligibility-based) may be useful to clinicians who seek to demonstrate the efficacy of interventions, to counsel patients on realistic expectations for their hearing aids, or to better understand the nature of their patients’ speech-in-noise difficulties.

**BP206 Noise Reduction Affects Speech Intelligibility at Different degrees: A Comparison between Audio-Only and Audiovisual Stimuli**

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**Background:** Noise reduction (NR) in hearing aid (HA) can improve speech understanding for people with hearing impairment (HI). The size of effect is more pronounced in laboratory tests over field tests. The reasoning of such difference is multi-dimensional. For example, audio-only presentation has dominated in laboratory tests in hearing science until now. The visual component, however, is known to affect sound perception and human behavior.

**Method:** In this study, we compared the effect of NR, turned at maximum versus minimum level in HA, on speech intelligibility. A range of HA programs in which NR was manipulated were provided to test participants. Two speech-in-noise tests, using the same range of HA programs, were administrated to twenty-seven HI with mild-to-moderate hearing loss (HL). Test 1 used an audio-only presentation. Test 2 used a Virtual Reality (VR) and loudspeaker array-based audiovisual presentation, in which a canteen ambient was recorded and the target speech was added. In both tests, the target speech was presented via the front loudspeaker facing the participant. Moreover, in Test 1, the participants were instructed to fix the gaze to the front loudspeaker. While in Test 2, they could freely move their head while experiencing the 360-degree view in VR. Naturally, a binary head movement classification (HMC) was recorded in Test 2 (i.e., participants not moving at all vs moving the head during the test). Also, a Simulator Sickness Questionnaire (SSQ) was used to assess the cybersickness in Test 2.

**Results:** The result indicated that NR was a significant factor in both tests, where NR was perceived as useful and bringing the most contrast. However, the effect size of NR in Test 2 was smaller. When involving HMC, a statistically significant interaction between HMC and NR was found. Namely that for the participants classified as with head movement, NR did not provide a significant benefit. Moreover, HMC was found significant in explaining all SSQ components. That is the participants classified as with head movement experienced greater cybersickness.

**Conclusion:** This study reported that user behavior modifies the effect of NR on speech intelligibility. The uninstructed head movement furthermore contributed to cybersickness. This is important to consider when testing e.g., HA signal processing and VR experience. However, though Test 2 improved the ecological validity in laboratory tests, the user behavior in real-life might differ. It nevertheless provided possibly a perspective of explaining the limited effect of NR found in field studies.

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**BP207 Effects of Hearing Status on the Time Course of Vocal Emotion Recognition**

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Prior research has shown that adults with age-related hearing loss demonstrate deficits in auditory emotion recognition, and that the use of hearing aids does not mitigate these losses. However, there may be some nuance in the deficits that hard-of-hearing and hearing-aided individuals experience that is not captured by traditional paradigms where emotional stimuli are presented. The current study investigated the time course of vocal emotion recognition in order to assess differences in emotion recognition based on hearing status. In a remotely-conducted study, adults with normal hearing or mild-moderate unaided hearing loss completed an adaptation of the auditory gating paradigm used by Pell and Kotz (2011). Seven-syllable pseudo-utterances conveying different emotions (anger, fear, sadness, happiness, and neutral) were divided into gate intervals based on the number of syllables. Participants completed a forced-choice task identifying the emotion of each stimulus at each gate interval in a successive, blocked design, beginning with one syllable presented and ending with all seven syllables presented. The emotion identification point was defined as the gate in which responses were correct and remained correct for subsequent gates. Results indicated that participants with hearing loss identified each emotion at later gates than normal-hearing participants and were less accurate overall. At the emotion identification point, individuals with hearing loss also required greater variability in acoustic features (e.g., range of f0 and amplitude) compared to normal-hearing participants, suggesting that challenges with identifying vocal emotions are partially tied to the acoustic characteristics of different emotions. Data collection is currently underway for an in-lab follow-up study using the same paradigm with hearing aid users to assess the effects of hearing aids on the time course and overall accuracy of vocal emotion recognition.

**BP208 How Should We Define and Measure Hearing Aid Use success? Perspectives of Adults who Have Hearing Aids and Hearing Healthcare Professionals**
Introduction: Hearing loss, affecting one in five adults in the UK, can be managed using hearing aids; however, the number of adults utilising hearing aids is far lower than the number who could benefit from them. Previous measures of hearing aid use, such as the number of hours the hearing aid(s) are switched on, may not align with individuals’ perspectives of what ‘successful’ use means. Consequently, clinical trials focussed on improving use may not be person-centred. Defining ‘successful’ use is key to ensuring that future health research and policy reflects the needs and priorities of those who access and use healthcare.

Objectives: To define and rank the most important aspects of success when using hearing aids, by consensus, from the perspectives of a) hearing aid users and b) healthcare professionals.

Design: A three-stage priority setting process was undertaken separately with 113 hearing aid users and 51 healthcare professionals.

1. Responses to an open-ended question: “Describe what successful hearing aid use means to you” (hearing aid users) or “From your perspective, please describe what successful hearing aid use looks like for adults who have them” (healthcare professionals) were grouped according to meaning, and summarised as an ‘indicative’ statement that conveyed the key concept of the statements in that grouping.

2. All indicative statements were reflected back to respondents in a second survey, where they were asked to choose and rank their top 10 most important statements.

3. The 15 highest ranked indicative statements from the second survey were presented at two final consensus workshops (one for hearing aid users, one for healthcare professionals), where the top 5 indicative statements to describe hearing aid success were prioritised by consensus, using Nominal Group Technique.

Results: The top 5 indicative statements from each consultation workshop will be identified as priority factors indicating ‘successful’ hearing aid use.

Conclusions: This study will identify aspects of successful hearing aid use from the perspectives of those who use hearing aids to manage hearing loss and healthcare professionals. These are vital factors to consider in future decision-making in audiology services and clinical trials.

The outcomes from this study will be disseminated using professional illustrations, making them engaging and widely accessible to key stakeholders and the public. The impact of this research will include contributions to awareness raising and initiating discussions around what is important to hearing aid users and healthcare professionals and the outcomes they want to achieve.

BP209 Remote and Lab Measurement of Facial Expressions as a Measure of Emotional Responses
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Introduction: Emotional responses affect individuals’ attention, memory, behavior, overall quality of life, and likelihood of participation in communication situations. Moreover, negative emotional responses (confusion/frustration) in adverse listening situations may deter individuals from participating in social situations. Hence, it is important to measure emotional responses in individuals with hearing loss. A novel way to objectively measure emotional responses is using automatic facial expression recognition software. Using this measure in the real world can help determine emotional response to real-world situations. In the current study, we determine if remote measurement under relatively controlled environment provides similar responses as the strictly controlled lab condition. Our aim was to explore how emotional responses obtained remotely correlate with that obtained in the lab setting. We hypothesize that the emotional responses obtained in the lab will be able to predict emotional responses in remote settings.

Method: We measured facial expressions in 33 young adults with normal hearing in laboratory and remote settings at -1 dB signal to noise ratio and in quiet. Our outcome measures were speech recognition scores, listening effort rating, objective measures of emotional responses and subjective rating of emotions. We imported the remote recordings into the facial expression recognition software. We baseline corrected the emotional responses, plotted their time course, and determined the area under the curve (AUC) for the emotional response.

Results: We saw that the speech performance was worse in the remote condition for both the quiet and the (F(1,32)= 6.29, p= 0.017) and the -1 dB SNR
(F(1,32)= 28.76, p< 0.001) condition, the listening effort rating for the remote condition was greater than the lab condition for the -1 dB SNR (F(1,32)= 11.171, p= 0.0021). Further, the objective emotional response of confusion was greater for the remote condition than the lab (F(1,94.45)= 24.25, p< 0.001). The subjective rating of confusion was not different between the remote and the lab conditions. Multilevel correlation showed a positive correlation between the lab and remote measures (r = 0.27 to 0.71, p< 0.05).

Discussion and Conclusion: Poorer performance, greater listening effort and greater negative emotional response in the remote condition were contrary to our predictions. This could be due to unstandardized transducers used, due to instability of the internet or Zoom connectivity/ audio transmission issues. Nevertheless, these results and the positive correlation of lab and remote measures shows us that facial expressions could be recorded in remote settings with some stimulus and environmental considerations.

BP210 Impact of Musical Aptitude on Subjective and Objective Correlates of Hearing Aid Processed Musical Stimuli
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Background: Despite the advancements in digital processing, the ability of the HAs to restore music appreciation in individuals with hearing impairment is reported to produce variable results and appears to be significantly compromised. Music appreciation, in turn, is linked to innateness (nature) and training (nurture). Due to the virtue of training, musicians exhibit better perceptual musical abilities than non-musicians, whose abilities, in turn, vary with their musical aptitude. Hence, there is a strong need to explore whether the variability in the perception of HA processed music has any links with the musical aptitude of the listener.

Methods: The study included a total of 30 participants consisting of three groups (musicians, non-musicians with good musical aptitude and non-musicians with low musical aptitude). The musical aptitude was measured based on the mini-Profile of Music Perception Skills (mini-PROMS) cut-off criteria. Musical stimuli having equal contributions of pitch, rhythm and melody were processed through a 16-channel HA programmed for a moderate hearing loss fit. The HA processed stimuli were recorded a SLM connected to a microphone placed on a ‘Manikin’. The processed stimuli were recorded in two HA settings: dedicated music program ‘on’ and ‘off’ and were subjected to perceptual and objective analyses. Subjective ratings were obtained from all three groups of participants using a visual analogue scale of ‘Adjective descriptors of Music’, a part of the Iowa Musical Background Questionnaire. The quality of HA processed music was quantified using HA music quality index (HAAQI) version 1, a function running on a MATLAB environment for the unprocessed and processed stimuli.

Results and Discussion: It was found that the non-musicians with poor musical aptitude rated the quality of HA processed musical stimuli lower than non-musicians with good musical aptitude (music program on and off), who in turn rated the HA processed music stimuli similar to trained musicians in both conditions. This finding suggests the positive influence of innate musical abilities on perceptual music quality in the former group, whose ratings matched the musicians. Also, comparisons of objective analyses between the two conditions (music program ‘on’ and ‘off’) in terms of the objective quality index (HAAQI) showed significantly higher music quality in the music program ‘on’ condition compared to ‘off’, indicative of the need for dedicated settings in the HA for music perception, where musical attributes such as pitch, rhythm and timbre are preserved.

BP301 A Questionnaire Study on the Impact of Face Mask on Individuals Suffering from Hearing Loss during the Pandemic
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Objectives: During the COVID-19 pandemic, mask wearing has become a major part of our daily lives. Despite its benefit of limiting transmission of respiratory droplets, face mask can introduce a barrier against communication and can be a source of inconvenience for those wearing hearing aids. We aim to investigate the impact of face mask on hearing aid usage and selection for individuals suffering from hearing loss.

Method: 72 patients aged 54 – 93 completed a questionnaire regarding the impact of face mask on communication, hearing aid usage, selection, and the decisions to pursue hearing aids at a busy Ear, Nose and Throat practice in San Francisco. Questionnaires were answered either at the first hearing evaluation encounter or in subsequent hearing aid follow up appointments.
**Results:** The questionnaire showed that face mask caused a substantial impact on communication in more than 70% of the participants. In particular, those with severe hearing loss or greater reliance on lip reading were more severely affected by face mask use. The impact on communication also drove more individuals to seek medical advice on hearing loss and to obtain hearing aids of their choices.

**Conclusions:** The widespread use of face mask during the pandemic contributed to communication problems among individuals with hearing loss and may explain the rebound growth on the demands of hearing aid products in the post lock-down era. Hearing care providers must take into account the multitude of physiologic and psychological effects of face mask wearing when counseling patients about hearing aid selection.

**BP302 New Research is Needed for Hearing Aids to Treat Severe and Profound Hearing Loss.**

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The term “Other” is fundamental to current philosophical discourse. As recently as March 18 2022 Professor Banu Subramaniam invoked Edward Said’s terms “in-place” and “out of place” to differentiate insiders from those outside or “Other”, during a keynote lecture at Duke University.

At the 2022 International Hearing Aid Research Conference, where all presentations “should report the results of original research”, this paper may be “out of place” and the hearing aid practitioner presenting may be “Other”. Nevertheless this paper is important because it is a call-out to hearing aid researchers from around the world who are “in-place” here at IHCON.

Around 87 million people worldwide have a severe and profound degree of hearing loss, but only 5 to 7% of those who meet criteria for a cochlear implant will go on to be implanted. The vast majority will continue to rely on hearing aids.

To improve outcomes with hearing aids, audiologists must deliver quality evidence-based care for adults with severe to profound hearing loss. To support this care a group of international experts have recently published practical guidelines for the audiological management of severe and profound hearing loss (Turton et al 2020).

The guidelines provide 153 recommendations for best practice, based on current published evidence and expert opinion, addressing a range of topics including selecting and fitting hearing aids. It is not a systematic review, however the development process required an intensive review of the existing evidence.

This paper will highlight the findings in the area of hearing aids for adults with severe and profound hearing loss, where too much of the evidence was weak (experience and expert opinion) or even out of date (20 year-old research about analog versus digital hearing aids). Up-to-date research about how improve outcomes with hearing aids for severe and profound hearing loss is badly needed. Because the underlying auditory abilities accompanying this degree of hearing loss vary hugely, we don't just need new technologies, but also diagnostic tools and accompanying outcome measures to target new technologies to the individuals who will benefit most.

As an appeal to hearing aid researchers, this paper is “in place”. It will highlight the gaps in the evidence and inspire new research which is needed for hearing aids in the management of severe and profound hearing loss.

**Reference:**


**BP303 Improved Accuracy and Efficiency in Virtual Acoustic Rendering for Hearing Aids Using Principal Components-Based Amplitude Panning**

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Virtual loudspeaker arrays can be rendered over headphones using a head-related impulse response (HRIR) dataset, but these techniques often introduce undesirable spatial and timbral artifacts. Similarly, virtual loudspeaker arrays can be rendered for aided listening conditions using a hearing aid-related impulse response (HARIR) dataset. Existing virtual acoustic rendering algorithms suffer from significant high-frequency errors, and these errors occur when the array’s sweet spot is smaller than the head and when the HARIR contains a high degree of spatial and spectral complexity. Minimizing such errors is even more important for hearing aid applications, which requires magnitude and phase accuracy broadband, including high frequencies. This accuracy is critical since a hearing aid’s benefit is often most significant at high...
frequencies. A new and efficient virtual acoustic rendering algorithm called principal components-based amplitude panning (PCBAP) has been developed which greatly minimizes magnitude and phase errors in reproducing aided and unaided listening conditions. The algorithm uses principal components analysis (PCA) performed on a time-aligned set of directionally dependent HARIRs. The resulting principal components (PCs) act as virtual acoustic filters, and the PCA scores are implemented as linear, time-domain weights. The PC filters serve the same role as virtual loudspeaker HRIRs, and the PCA scores serve the same role as panning gains, when compared to a vector-based amplitude (VBAP) or higher-order Ambisonics (HOA) processing algorithm. A spatial and spectral error analysis of magnitude and phase errors for each hearing aid microphone showed similar HARIR magnitude reconstruction accuracy using 4-9 filters, compared to a 121-loudspeaker array. Phase reconstruction accuracy showed even greater improvement, having accuracy with only 4 PC filters that exceeded that of a binaurally optimized 121-loudspeaker array. The high frequency phase accuracy of the PCBAP algorithm allows for phase sensitive aspects of a hearing aid to be accurately rendered, such as the beamformer or combination of the aided and unaided pathways. Another benefit of PCBAP is the inherent scalable nature of the algorithm. With the same fixed set of filters, PCBAP is capable of accurately rendering a source from any direction, whether the sources are single, free-field sources or loudspeakers in a virtual array, rendering a complex background scene. An overview of the algorithm will be discussed and results for magnitude and phase errors will be presented. Also, the algorithm has been implemented into a demonstration for interactive fitting in an audiology clinic, and an example of this virtual reality demonstration will be presented.

BP304 Language-Independent Hearing Screening – Increasing the Feasibility and Validity of a Hearing Screening Self-Test at School-Entry

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About one per 1000 children are born with hearing loss. Therefore, many developed countries have implemented neonatal hearing screening programs. However, reports indicate that the prevalence of hearing loss doubles before the age of nine and for that reason, the World Health Organization (WHO) and the European Federation of Audiology Sciences (EFAS) strongly recommend screening all children for hearing loss and ear diseases at school entry as a bare minimum. Ideally, every child with hearing thresholds of 20 dB HL or higher should be identified, as even mild hearing loss affects educational attainment. However, hearing screening in young children requires an innovative approach as practical test conditions, acoustics, and stimuli, and the applicability in young children are challenging. Not all countries have resources for the development, validation and implementation of school-age hearing screening. Therefore, the key to success is choosing an appropriate screening method that is cost-effective, quick and inexpensive, highly reliable and valid, and can be used internationally without adaptations.

At this moment, research is being conducted to develop a language-independent sound-in-noise test, the Sound Ear Check (SEC). This is an automated adaptive self-test on a tablet based on recognizing masked ecological sounds. The SEC has already been evaluated in adults and children and has shown promising results. A reference curve with a steep slope of 18%/dB was obtained, resulting in a test with a high measurement precision of 1 dB. Significant correlations with both pure tone thresholds ($r = 0.70$) and the Digit Triplet Test ($r = 0.79$) speech-in-noise test were found in adults. Sensitivity and specificity values of about 80% were obtained. However, the feasibility of the SEC in young children was questionable, and 20% of the children tested were not able to obtain reliable results.

The current study is an international collaboration between multiple European countries. It aims to find the optimal test procedure and features for a self-test for young children. Three different test versions were evaluated regarding feasibility and validity in detecting conductive and sensorineural hearing loss in young children. A large group of young children at the age of school entry (5-6 years) is currently being tested in different countries. In-depth results of this study will be presented. This research will not only bring us closer to an international applicable screening test but will also help as a guideline for the development of other self-tests for young children.
**BP305 Investigation of hearing device technologies for individuals with hearing impairment in noisy workplaces**

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In the United-States, 12% of the working population suffers from hearing loss and 25% is exposed to hazardous noise. For many workers operating in high-level noise environments, wearing hearing protectors is often necessary to reduce sounds to safe levels. Because hearing loss is characterized by higher hearing thresholds, these protectors often worsen a hearing-impaired user’s ability to perceive and localize sounds. Thus, such users are more likely to suffer from difficulties in communication and performing tasks efficiently and safely. Moreover, they may wear hearing aids at work, with or without hearing protectors, even though consequences of this practice have not been extensively researched. There is a need to develop a protective hearing aid able to amplify sounds of interest thus preserving speech perception yet reducing noise exposure when necessary. This work explores three axes: enhancing communication, managing noise exposure, and maintaining safety.

Communicating in noise is a common struggle, especially for people with hearing-impairment. To enhance communication, the hearing technology should be individualized by adapting amplification parameters according to the user’s hearing loss while preserving speech intelligibility. Multichannel wide dynamic range compression (WDRC) is widely recognized in current hearing aid technologies to personalize the device depending on the user’s hearing profile. Protection is allowed with both amplification of soft sounds and reduction of loud sound levels. However, some studies have highlighted that, depending on the selected parameters and the noise environment, speech intelligibility may be compromised after WDRC processing. Speech intelligibility relies greatly on acoustical cues, i.e. frequency and time components of the speech signal. This research will explore how WDRC parameters affect speech intelligibility in the context of noisy occupational environments. After objective evaluation and optimization, the designed algorithms will be validated on participants with or without hearing loss.

Noise hazard depends both on the level and the duration of noise exposure. The implementation of real-time in-ear noise dosimetry will be combined with adaptive amplification/compression to prevent the worker from being overexposed to excessive noise. This will also pave the way for research on a better understanding of the risks faced by workers with hearing-impairment operating in high-level noise environments.

Attenuating loud sounds combats over-amplification and should prevent further hearing loss but can also lead to safety issues. Alarm sounds and other key sounds are loud by nature but should not be missed by the user. Attenuation/compression algorithms should then be adjusted such that the workers’ safety is guaranteed.

**BP306 What a difference an aid makes – explaining speech intelligibility differences during streaming**

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**Rationale:** Audio and phone streaming functionality has become an expected and useful feature for many hearing aid (HA) users. Direct drive technology, providing extended bandwidth of audible amplification with an open fit, delivers better streaming sound quality for speech and music compared to traditional acoustic hearing aids (AHAs; Mandikal Vasuki et al., 2020; Levy et al., 2021). We set out to investigate whether there would also be a benefit to streamed speech intelligibility, specifically in the presence of environmental noise. In a pilot study with HA users, we found an average 26% improvement when HA users switched from listening to streamed speech through a traditional AHA to a direct drive HA while listening to environmental noise at a constant level. In this study, we seek to dissociate factors contributing to this surprisingly large effect. We hypothesize that the benefit may be due improved overall SNR at the eardrum, due to the extended bandwidth of amplification that is provided by directly driving the eardrum.

**Design:** For the pilot study, listeners with mild to moderately-severe hearing impairment were bilaterally fitted with both traditional vented AHAs, and a direct drive aids. Connected Speech sentences were streamed to each device, in separate trials, at equivalent input levels with hearing aid microphone disabled. Participants adjusted the level of environmental noise presented acoustically through loudspeakers until they could not hear the streamed speech anymore. Environmental noise level was reduced by 5 dB.
and speech intelligibility was measured with each device. Ratings for listening effort and perceived speech intelligibility were also obtained for the two conditions. In order to disambiguate the contribution of various frequency bands towards speech intelligibility for streaming, we will be repeating the testing by simulating various narrowband fittings within the direct drive HA.

**Results:** Pilot results from six participants showed superior streamed speech understanding in the direct drive condition than in the AHA condition. Participants also reported lower listening effort and higher perceived speech intelligibility with the direct drive HA than with ACHA. We will be presenting additional data on the contributions of various frequency bands of amplification to explain the effect seen in the pilot study.

**Conclusions:** Participants performed better with direct drive technology while listening to streamed speech in noisy situations in part due to improved SNR at the eardrum. Other potential reasons for improved speech recognition and superior perceived performance with the direct drive hearing aids will be discussed.

**BP307 Factors Influencing Hearing Help-Seeking and Hearing Device Uptake in Adults with Hearing Difficulties: A Systematic Review of the Past Decade**

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**Objective:** The aim of this systematic review was to examine the audiological and non-audiological factors that influence hearing help-seeking and hearing device uptake in adults with hearing difficulties based on the literature published during the last decade.

**Design:** Peer-reviewed articles published between January 2011 and February 2022 were identified through systematic searches in electronic databases CINAHL, PsycINFO and MEDLINE.

**Results:** Forty-two articles were included in the review. These studies investigated 72 and 161 factors respectively in relation to help-seeking and hearing device uptake. Significant non-audiological factors that influence both hearing help-seeking and hearing device uptake include the following categories: demographics (e.g., age, sex), health/cognition, and behaviour. Furthermore, factors in the social category affect hearing help-seeking alone whereas factors in the categories of finances/work, funding/insurance, audiology appointment, motivation, and technology affect hearing device uptake alone. Significant audiological factors that influence both hearing help-seeking and hearing device uptake emerged in the following categories: pure tone audiometry, self-reported hearing difficulties and beliefs, communication, and hearing aids. Additionally, factors in the categories of speech testing, hearing healthcare consultation, and readiness for change influenced hearing device uptake alone. Of the included studies, 28 were classified as level 4 evidence, 12 as level 3 evidence and 2 as level 2 evidence. In terms of quality, 37 studies were rated fair, 1 good and 4 poor quality.

**Conclusions:** Various factors relating to help-seeking and hearing device uptake have been investigated although limited studies examine factors like the influence of the hearing device cost. Some factors have conflicting findings, like the influence of self-reported health, requiring further exploration. Our findings inform clinical audiological practice for help-seeking and hearing device uptake by patients.

**BP308 What Are the Experiences of Adult clients, Significant Others and Clinicians with Using Wireless Microphone Systems to Manage Hearing impairment?**

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This study explored the perceptions and experiences of adults with hearing impairment, their significant others, and clinicians on the use and provision of wireless microphone systems. A qualitative descriptive methodology was used, with a total of 43 participants across three groups: (1) 23 adults with hearing impairment who used wireless microphone systems; (2) 7 significant others of adults who used wireless microphone systems; and (3) 13 clinicians who provided wireless microphone systems to adults with hearing impairment. Participants completed an individual semi-structured in-depth interview to explore their experiences, with data analysed using thematic analysis. Analysis revealed five themes encompassing the perceptions and experiences of wireless microphone systems: (1) with experience and clear expectations, users believe in wireless microphone systems and how they can make a difference; (2) the trial and decision-making process; (3) what happens when clients use wireless microphone systems; (4) issues with wireless microphone systems and technology;
and (5) users require ongoing training and support to use wireless microphone systems.

These findings highlight the complexities of providing and using wireless microphone systems with adults with hearing impairment. However, clients, significant others and clinicians all reported that with appropriate experience, expectations, training, and support, wireless microphone systems can make a real difference to listening and communicating in different situations. There is also opportunity to involve significant others more throughout the rehabilitation process.

BP309 Influence of Hearing Aid Experience on Electrophysiological Measures of Speech detection, discrimination, and Comprehension
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It is well known that sensory deprivation and stimulation can induce changes in brain structure and function. However, literature findings regarding the effects of sensorineural hearing loss (SNHL) and hearing aid (HA) treatment on the cortical processing of speech sounds are mixed. The current study therefore investigated the influence of HA experience on cortical speech processing. Three age-matched groups of older participants (N = 3 × 15) were tested: (1) participants with a pure-tone average hearing loss of <25 dB HL from 500 to 4000 Hz, (2) participants with mild-to-moderately-severe SNHL but no prior HA experience, and (3) participants with mild-to-moderately-severe SNHL and ≥2 years of HA experience. In terms of behavioral measurements, speech detection thresholds (SDTs) and speech recognition thresholds (SRTs) were measured. In terms of electrophysiological measurements, speech evoked N100, P300, N400 and Late Positive Complex (LPC) responses were measured using multi-channel electroencephalography (EEG). The N100 and P300 responses were evoked using an active auditory oddball paradigm. The N400 and LPC responses were evoked using an arithmetic paradigm with either audio-only or audio-visual stimulus presentation. The EEG measurements were performed at 10 dB above the individual SRTs. All measurements were performed in the free field in the presence of stationary speech-shaped noise. The participants in groups 2 and 3 were tested with HA fittings that provided good audibility. The analyses revealed no group differences in any of the EEG components. Regarding the arithmetic paradigm, audio-only presentation led to larger N400 amplitudes than audio-visual presentation. Overall, these results suggest that, when audibility is ensured, cortical responses reflecting the detection, discrimination, and semantic processing of speech remain ‘intact’ in individuals with SNHL regardless of HA experience.

BP310 Development of a new German speech test using synthetic speech
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Existing German speech-recognition tests have disadvantages like training effects, outdated words, or a limited number of sentence lists. The goal of this project is to develop a new speech-recognition test using a Text-To-Speech (TTS) system to generate the speech material. First, the quality of different TTS systems was evaluated. A TTS system based on deep neural networks was rated best in the quality dimensions prosody, speech flow, and naturalness. Using the preferred TTS system, sentences from an existing German speech-recognition test were synthesized and compared to the original regarding listening effort and speech recognition. The synthetic speech resulted in better speech-recognition scores and listening effort was comparable to that for natural speech. In conclusion, the TTS system is suitable for the generation of speech-recognition tests.

In a next step, the requirements for the new speech-recognition tests were compiled. These included that the speech material should be composed of meaningful and nowadays well-known words covering different parts of speech. Three to four words should each be combined in the same natural syntax to form sentences or phrases. A large amount of those combinations and test lists should be available to allow test repetitions and avoid training effects. It was decided that the new speech material is composed of phrases of the structure "article-adjective-noun-infinitive" (German: “Den netten Mann grüßen”; English with different word order: “to greet the nice man”) and is therefore called Oldenburger Phrasen (OLPhra).

To compose the speech material, several annotated German corpora were filtered by word categories to obtain as many different words as possible. Since the words should be well-known, their frequency was analyzed with a database. With the selected words, thematic noun categories (e.g., food, people, ...) were
formed and adjective and infinitives which are related to the categories were added. Subsequently, each noun per category was combined with all adjectives in the same category and meaningless, discriminatory, or too negative combinations were dismissed. This process was repeated for the nouns and infinitives. The remaining adjective-noun and noun-infinite combinations were merged and the selection process was applied again. Test lists formed of 20 phrases each should be balanced in terms of number of syllables, phoneme frequency, noun categories, articles (German: den, die, and das) and infinitives without prefix (e.g., to inform) and with prefix (e.g., to misinform). This contribution presents the procedure for developing speech-recognition tests and discusses its potential for continuous update of the speech material.

**BP311 The Impact of Changes in Hearing Thresholds on Hearing Aid Fitting in Children who are Hard of Hearing**

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Children with hearing loss may show increases or decreases in their hearing thresholds as they age. With these changes, children who use hearing aids may need adjustments to their hearing aid programming to accommodate changes in unaided hearing thresholds. However, previous research suggests that hearing aids are often underfit relative to their prescriptive targets, as approximately 50% of pediatric hearing aid fittings deviate from targets by 5 dB or greater. In this study, we examined how changes in hearing thresholds impact pediatric hearing aid fittings and aided audibility in children who use hearing aids. Participants included 190 children with hearing loss who use hearing aids (age 8 months – 11.5 years old) with mild-to-severe hearing loss at baseline measurement. Pure-tone hearing thresholds were assessed and hearing aid verification was performed at annual intervals. We examined relationships between magnitude of change in pure-tone thresholds and: 1) change in deviation from prescriptive targets (i.e., RMS error) and 2) change in aided ability (i.e., aided Speech Intelligibility Index). We found that deviations from target were independent of changes in unaided hearing thresholds. For instance, some children improved their fitting (i.e., smaller RMS error) after a decrease in their hearing and others had poorer fittings after this decrease. Several children saw changes in RMS error despite no changes in hearing thresholds. However, changes in hearing thresholds (specifically higher frequency thresholds) were associated with changes in aided audibility, such that as hearing worsened, aided audibility decreased. These findings suggest that progressive hearing loss, particularly in the higher frequencies, may result in a loss of audibility even for children who use hearing aids. This may be because audiologists are not uniformly making the necessary adjustments to hearing aid programming to compensate for shifts in hearing, or because audibility cannot be fully restored via hearing aids due to the severity of the hearing loss. Clinically, these findings suggest that regular verification and programming of hearing aids is key to supporting a child’s access to audible speech. Ongoing research examines how ear canal acoustics impact longitudinal changes in hearing and verification of hearing aids.

**BP312 Introducing the Audible Contrast Threshold Test – a Clinically Viable Measure of Spectro-Temporal Modulation Sensitivity**

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Measures of spectro-temporal modulation sensitivity have shown significant correlation to speech-in-noise performance in several studies with aided hearing-impaired test subjects. Results have been particularly promising when speech-in-noise testing was done in an ecologically valid test condition (e.g., with everyday target sentences presented against a background of running speech maskers coming from separate loudspeakers set up in a room with average living-room acoustics including reverberation). Such measures are clinically interesting, as upfront knowledge about a prospective hearing-aid user’s realistic speech-in-noise performance might be useful for individualization of hearing aid fitting and can contribute to more targeted patient counseling. Notably, compared with ecologically valid speech-in-noise testing, measuring spectro-temporal modulation sensitivity requires limited equipment and avoids language-specific test material. Therefore, a clinically viable implementation of spectro-temporal modulation sensitivity testing is proposed and termed the Audible Contrast Threshold test. The test builds on previous research versions of spectro-temporal modulation sensitivity testing but solves earlier issues with
long and tiring test runs, and the need for non-standard equipment for the patient response. The development rationale was to make testing as similar as possible to the pure-tone audiogram procedure: audiologist-operated threshold tracking according to a Hughson-Westlake 2-down-1-up rule, patient responses recorded with the audiometry push-button, and stimuli delivered via standard clinically used headphones. The stimuli are presented in 1-second “waves” as a train of unmodulated-noise reference waves alternating with modulated-noise target waves activated by the audiologist. The result is given in dB normalized Contrast Level (nCL), with 0 dB nCL corresponding to the average performance of young normally-hearing adults and 16 dB nCL corresponding to maximum possible modulation. Thus, positive dB nCL values indicates a “contrast loss” while negative dB nCL indicates better-than-normal performance, akin to the pure-tone audiogram and the dB HL scale. This contribution will present the normative data used to determine the dB nCL scale, as well as test-retest reliability. In addition, results will be shown from several experiments where Speech Reception Thresholds (SRT) were measured with HINT in an ecologically valid (aided) condition and compared with Audible Contrast Threshold results, corroborating the relationship between the two measures and adding German to the pool of languages previously investigated (US English, Swedish, Danish).

BP313 Hearing Safety of Transcranial Ultrasound Modalities with Emphasis on Parameters for a Novel Hearing Aid Device
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Ultrasound (US) research has grown rapidly in the past decade for noninvasively modulating brain regions with high spatial resolution and as a tool for creating transient openings in the blood-brain-barrier for drug delivery, showing exciting applications of this new technology. While investigating the neuromodulation technique, our lab discovered that US applied to the head readily activates the auditory system through vibrations of cerebrospinal fluid that then directly vibrate fluids within the cochlea (Guo et al., Neuron, 2018). Due to the potential applications of US induced auditory activation, our group is interested in characterizing safe levels of US pressure for the hearing system, particularly when used for neuro-modulation, blood-brain-barrier, or hearing applications. To characterize this, we collected auditory brainstem responses (ABRs) and electrocochleography (ECochG) in response to air-conducted acoustic pure tones (2, 4, 8, 12, 20, and 30 kHz) and broadband noise at varying levels (10-70 dB SPL) before and after US stimulation in anesthetized guinea pigs. Both acute and chronic preparations were performed to fully characterize the potential for a parameter to be damaging. Control data was also collected for characterizing the stability of the recording protocol and for standard noise-induced hearing loss (NIHL). We assessed ABR and ECochG thresholds, amplitudes, and latencies over time to identify changes that are associated with hearing damage. Some tested US parameters showed neurophysiological changes associated with hearing loss with most tested parameters outside of the hearing aid modality showing some form of loss. Parameter settings used to effectively send complex information to the auditory system don’t show any changes associated with hearing loss at lower pressures. When changes were seen, threshold shifts were most prevalent in the high frequencies with some more severe cases of US stimuli causing threshold shifts in the middle frequencies as well with a similar pattern to NIHL. A reduction in ABR wave amplitudes can be seen also and including some parameters without significant shifts. Choice of center frequency may have an impact on safe level ranges. Future studies will include an in-depth characterization of the US parameters of interest also in large animal models that better mimic the head size in humans.

BP314 Effects of Mandibular Motion for In-Ear Hearing Devices
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Design of a one-size-fits-all hearing device such as a hearing aid, in-ear monitor, or earbud is difficult due to individual variability in pinna size and shape, ear canal volume and geometry, and cerumen production. Further increasing design difficulty is the poorly understood interaction between mandibular movement, device placement, and its impact on device performance. Mandibular motion (occurs during talking, singing, chewing, breathing, head movement) provides a nearly continuous source of external ear canal movement resulting from forces applied by the condyle of the mandible against the walls of the ear canal. Here, the effects of mandibular motion on the attenuation provided by a closed ear piece (earplug) were investigated by testing anatomically correct models
of human ears using a custom acoustic test fixture developed for this study. Anatomical ear models simulating the entire pinna and canal up to the second bend were obtained from impressions of human ears (N=60 ears) with the jaw open, jaw closed, and jaw open with head turned toward one side. An acoustic test fixture was created that housed a standardized ear simulator with ear canal extension (GRAS RA0045), microphone (GRAS 40AH ¼”), loudspeaker (JBL D220TI), and a single-board microcontroller with stepper motor (Arduino). Mandibular motion was simulated in repeated cycles using a computer-controlled drive-pin with a silicone tip that articulated with the portion of the model ear canal that contacted the condyle of the mandible (i.e., the mandibular bump). The test procedure simulated various levels of mandibular motion by measuring insertion loss in the presence of 130 dB pink noise (attenuation with no earplug minus attenuation before and after various number of mandibular motion cycles) with and without the presence of artificial cerumen. Results indicate that as few as 50 cycles of mandibular motion could significantly compromise hearing device coupling and compromised insertion loss was highly subject-dependent. The methods developed for this study represent a novel approach to simulate the interaction between the physical design and real-life usability of hearing devices. The apparatus and methods used here could be extremely valuable in future investigations of the effects of mandibular motion on the fit and acoustic seal of custom and non-custom ear pieces.

BP315 UltraHearing: Complex Activation of the Auditory System Using Body-Coupled Ultrasound

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Ultrasound stimulation (US) is an exciting new technique to non-invasively modulate neural activity with high spatial precision. Recently, studies have demonstrated that US, when coupled to the animal, activates the auditory system through an indirect, peripheral pathway (Guo et al., Neuron 2018). The current hypothesis posited by these studies suggests that the method of activation is via a fluid pathway, in which the ultrasound waves vibrate the cerebrospinal fluid and travel into the cochlea via the cochlear and vestibular aqueducts. Due to the differing mechanism and pathway of activation, US induced activity of the auditory system will have different characteristics from traditional air stimuli. Our lab has previously reported on the comparisons between air-evoked and US-evoked neural activity, and how US can effectively evoke frequency specific activity when modulated with pure tones. However, these experiments have not explored the complexity of information which can be encoded.

In this study, we investigated the neural activity in response to complex signals transmitted via air-, bone-, and fluid-conduction (ultrasound). Neural activity was recorded using a two-shank 32-channel NeuroNexus electrode array placed in the central nucleus of the inferior colliculus (ICC) of anesthetized guinea pigs. Air-conducted stimuli consisted of guinea pig vocalizations presented via a speaker. For fluid-conducted stimuli, we extracted the envelope of the vocalizations and then amplitude-modulated a 220 kHz sinusoid. This signal was presented via an ultrasound transducer coupled directly over the brain of the guinea pig with agarose and a focusing cone. Bone-conducted stimuli were presented with a B-81 bone conduction device coupled to the skull via a skull-nut system (Curthoys et al., Exp Brain Res, 2006).

Our results demonstrate that complex auditory information can be encoded in ultrasound, and the ICC reliably responds to vocalizations when presented via air-, bone-, or fluid-pathways. Our analysis demonstrates similarities between the air-, bone-, and fluid-driven stimulation approaches while also highlighting that these paths are different. Further research will investigate how the fluid pathway differs from the air-conduction pathway. Better understanding of how ultrasound stimuli act on the auditory system can help guide the development of next-generation hearing devices, possibly even a combination of technologies leveraging the different conduction pathways.

BP316 Microphone Directionality in Bimodal Listening

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Bimodal hearing is defined as using a cochlear implant (CI) on one ear and a hearing aid (HA) on the other ear. This study investigated the effect of microphone directionality on speech understanding of bimodal listeners under several spatial configurations of speech and noise.

Firstly, the relative hearing performance of the CI and HA ears was characterized by scores for CNC words in quiet. Secondly, to validate a model of bimodal hearing proposed by Dieudonné and Francart (2020), Speech Reception Thresholds (SRTs) were measured.
for CI alone and bimodal listening, with speech (sentences) from the front, and speech-weighted noise either from the front or the HA side. Thirdly, SRTs were measured with three combinations of fixed directionality (dir) and omni-directional (omni) microphone configurations: 1) CI dir and HA dir, 2) CI omni and HA dir, 3) CI dir and HA omni. Speech was presented from either the front or the side, with noise from the other three cardinal directions.

Preliminary results are consistent with the Dieudonné and Francart (2020) hypothesis that bimodal listeners do not use binaural cues. Dual fixed directionality was better than asymmetric directionality for speech from the front, while asymmetric directionality was better in some conditions with speech from the side. Results will be presented for 24 bimodal listeners, encompassing a wide range of relative CI and HA hearing performance.

BP317 Withdrawn by author

BP318 Speech recognition in the presence of speech maskers in children
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Listening to a target talker in the presence of other talkers follows a protracted developmental period and is particularly challenging for children. Such listening skills are critical for speech-language development. Evaluating speech-in-speech recognition has substantial translational value, e.g., for hearing aid and cochlear implant fittings. In addition, a speech-in-speech test would help detect supra threshold hearing problems that are functionally important. However, such needs are currently unmet. The purpose of the present study was to examine speech-in-speech recognition ability using digits in children. The major advantages of using digits are that it overcomes the biggest challenge of test administration, and digits are among the few first words a child learns. A matrix style speech recognition using digit triplets and coordinate response format was used. The test had four conditions similar to the Listening in Spatial Noise test. (1) Low-cues: The target (male talker) and two male maskers are presented from the front. (2) Talker advantage: The target and two female maskers are presented from

the front. (3) Spatial advantage: The target is presented from the front, and two male maskers are presented, one each from + and - 90 degrees. (4) The target speech is presented from the front, and two female maskers are presented, one each from + and - 90 degrees. This test was also implemented into HDA 200 headphones using a generic head-related transfer function. Data were collected from 30 children (4-12 years) using full bandwidth loudspeakers (up to 20 kHz). The initial results demonstrate the feasibility of the computerized version of the test for examining masked speech perception in children. Further, results will be discussed in the context of age (developmental effects), various conditions, test-retest reliability, demographic variables, and potential clinical applications.

BP319 Frequency Importance Function in Simulated Bimodal and Electric Acoustic Stimulation Hearing
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Background: Bimodal and electric acoustic stimulation (EAS) users differ in degrees of residual hearing and insertion depths of an electrode array. These differences affect spectral processing significantly, which is critical for speech perception. In this study, using simulation of bimodal and EAS hearing, we derived a frequency importance function (FIF) in bimodal and EAS hearing.

Methods: Two groups of ten normal hearing adults were recruited. Acoustic hearing was simulated using low-pass filters with a cutoff frequency of 500 Hz. For electric simulation, a 6-channel sinewave vocoder was used with an output frequency range (1000-7938 Hz) with three input frequency ranges to create frequency maps found in bimodal and EAS patients: overlap (188-7938 Hz), meet (500-7938 Hz), and gap (750-7938 Hz), relative to the cutoff frequency of acoustic hearing. To determine FIF, six single-spectral hole conditions were created by setting the amplitude of each channel to zero. Fifteen two-spectral hole conditions were created by setting the amplitudes of two adjacent or distant channels to zero, along with all 6 channels intact as a control. For bimodal hearing, the acoustic and electric stimulations were delivered to opposite ears in noise, whereas for EAS hearing, both stimulations were presented to the same ear in quiet. Sentence perception was measured with acoustic stimulation alone, electric stimulation alone, and
combined stimulations as a function of spectral hole and frequency maps. We derived a FIF from speech perception scores using information theory.

**Results:** The FIF differs between the two hearing technologies and across frequency maps. In bimodal hearing, the lowest band contributed the least in overlap and meet maps, while the second lowest band contributed the least in the gap map. The higher two bands contributed the most across all frequency maps. In EAS hearing, the FIF is relatively flat compared to the bimodal hearing FIF. The three higher bands contributed the most and the three lower bands contributed the least for the overlap and meet maps. Meanwhile, the opposite was true for the gap map.

**Conclusions:** Different FIFs between hearing technologies suggest different spectral processing mechanisms. This could also be a result of different testing conditions (bimodal in noise while EAS in quiet). Further testing of EAS in noise is being conducted to confirm this possibility. Meanwhile, different FIFs between different frequency maps suggest some spectral integration and interference occurring, but further study is warranted for affirmative conclusions.

**BP320 Reverberation impairs speech intelligibility in noise. Is it even worse for hearing-impaired listeners?**

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Hearing-impaired (HI) listeners often complain that they have a hard time understanding speech in noisy environments, in particular in reverberant rooms. Several effects of reverberation can impair speech intelligibility in noise. They have been extensively studied for normal-hearing (NH) listeners: temporal smearing of the target speech that becomes intrinsically less intelligible, reduction of the dip listening advantage associated with the temporally smeared modulations in the envelope of the masker that becomes more masking, reduction of the spatial release from masking relying on binaural hearing. The present study focused on the two first monaural effects to understand whether HI listeners were relatively more affected by reverberation than NH listeners, and if so, why. The two effects were studied separately and in combination, by applying reverberation either only on the target speech, only on the noise masker, or simultaneously on both sources. The intelligibility scores of 32 NH and 32 HI listeners were measured at several signal-to-noise ratios (SNRs) and several reverberation levels, for both stationary or speech-modulated noise maskers. The HI listeners were tested without their hearing aids, but received linear amplification that partly compensated for their hearing loss. The intelligibility scores of both groups were analysed using Bayesian statistics to detect for the presence and absence of significant differences between groups. The detrimental effect of the speech temporal smearing was found similar in both groups. The noise temporal smearing was less detrimental for the HI than for the NH listeners, because the HI listeners experienced less dip listening advantage to start with. Note that the comparison of dip listening advantage was greatly affected by the SNR at which it was evaluated. Because the speech temporal smearing had the strongest influence on the intelligibility scores, overall, reverberation was not more detrimental for the HI listeners.

**BP321 Interactions between Slopes of Residual Hearing and Frequency Maps in Simulated Bimodal Hearing and Electric Acoustic Stimulation**

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**Background:** Numerous bimodal and electric acoustic stimulation (EAS) studies attempted to relate bimodal/EAS benefit in speech perception with the degree of residual hearing of the hearing aid ear, producing mixed results. This inconsistent correlation may be due to different slopes of residual hearing across patients, which is not carefully considered in the studies. Our pilot data suggests that slopes of residual hearing affects bimodal and EAS benefit significantly. In this study, using simulation of bimodal and EAS hearing, we determined the effects of the slopes on bimodal benefit in speech perception and interactions with different frequency maps in bimodal and EAS hearing.

**Methods:** Two groups of ten adults with normal hearing were recruited for simulated bimodal and EAS hearing. Sentence perception was measured in quiet and noise with acoustic stimulation alone, electric stimulation alone, and combined stimulations. For the acoustic stimulation, three slopes of high-frequency hearing loss typical in bimodal patients were created using low-pass filters with a cutoff frequency of 500 Hz: steep (96 dB/octave), medium (48 dB/octave), and shallow (24 dB/octave). For the electric stimulation, 8-channel sinewave vocoder was used with a
fixed output frequency range (1000-7938 Hz) with three input frequency ranges to create typical frequency maps for both bimodal and EAS hearing: overlap (188-7938 Hz), meet (500-7938 Hz), and gap (750-7938 Hz), relative to the cutoff frequency in the acoustic ear. Sentence perception was measured as a function of filter slope and frequency map.

**Results:** Correlation between bimodal/EAS benefit and residual hearing significantly improved when slopes were carefully considered. Bimodal/EAS benefit was significantly improved as slopes became shallower and SNRs improved. The effects of the slopes on bimodal/EAS benefit were greatest with the meet map, followed by the gap map, and the least with the overlap map. With overlap, EAS benefit was marginally greater than bimodal benefit. With meet and gap, EAS benefit was greater at two higher SNRs while bimodal benefit was greater at two lower SNRs.

**Conclusions:** The results indicate that spectral information, residing under the slope of residual hearing, plays a significant role in bimodal and EAS hearing. The optimal frequency map differed with different slopes, suggesting that the slopes of residual hearing in the acoustic ear should be carefully considered in fitting bimodal and EAS hearing. EAS hearing provided greater benefit over bimodal hearing, suggesting that spectro-temporal integration effectively occurred within ear (i.e., EAS) better than across ears (i.e., bimodal).

**BP322 Comparing Speech Intelligibility Improvements in Noise with Technical SNR-Improvements of Hearing Aids in a Complex Listening Scenario**

**Tim Jürgens***, Peter Ihly*, Jürgen Tchorz, Sébastien Santurette*, Johannes Zaaar*, Sören Laugesen*, Gary Jones*, Thomas Behrens*

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**Introduction:** Besides amplification, most commercially available hearing aids use spectral and spatial noise reduction algorithms to improve the signal-to-noise ratio (SNR) within an acoustical scene. However, not all improvements in SNR can be exploited by hearing aid users for improvement of speech intelligibility, due to possible introduction of artifacts. The goal of this study was to assess speech reception thresholds (SRTs) in noise with and without noise reduction and compare SRTs to technical SNR-improvements within the same scenario.

**Methods:** SRTs of 59 experienced hearing aid users with a large variety of hearing losses from mild to severe were measured. High-end hearing aids were binaurally and fitted according to NAL-NL2 with verification of target gains using real-ear measurements (REM). After an accommodation period of at least two weeks, the subjects’ SRTs were obtained in a loudspeaker setup with frontal target speech and two symmetrically placed interfering talkers from 100° and 260° azimuth. Four conditions were tested, (1) noise reduction off, (2) very mild noise reduction, (3) very strong noise reduction, and (4) an unaided condition as reference. Noise reduction involved processing using different strengths of a minimum variance distortionless response beamformer and a deep-neural-network processing postfilter.

**Results:** SRTs in noise varied considerably across subjects and were on average 1.4 dB better with very mild and 3.9 dB better with very strong noise reduction than with noise reduction off. Average unaided SRTs were not measurable in nine participants due to insufficient speech intelligibility when not using hearing aids. For those measurable, the unaided SRTs were 2.9 dB poorer than with noise reduction off. Technical measurements in the same setup with the phase inversion method showed 0 - 3 dB SNR-improvement for very mild, and 4 - 9 dB for very strong noise reduction over the ranges of SNRs tested with participants, see also poster by Santurette et al.

**Discussion and Conclusion:** With mild noise reduction the output SNR improvements were in line with the average SRT-improvements, whereas with strong noise reduction settings, the SRT-improvements were generally higher, but did not exploit the full output SNR-enhancement potential that was seen in the technical measurements. Possible explanations for this limit may be direct-sound interference, or that the subject-inherent acoustic scene decomposition may be disturbed by too strong noise reduction. [This work was funded by William Demant Foundation.]

**BP323 Identifying the Cues that Support Intelligibility of Reverberant Speech**

**Ramesh Kumar Muralimanohar***, James Kates*, Kathryn Arehart*

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People typically communicate in indoor environments where speech is degraded by both noise and reverberation. Particularly, reverberation-related distortions alter the spectral and temporal envelope cues that support speech perception. The effect of these perturbations is more intrusive for people with hearing impairments and hearing aids. Identifying the cues that help such people successfully understand speech in different environments might not only help characterize the impact of environmental factors on their ability to understand speech in different contexts but also help us test the effectiveness of any signal processing that is intended to aid listeners in these environments.

The goal of this experiment is to identify the relative importance of envelope modulation-rate cues in determining the intelligibility of reverberant speech. In order to analyze the relative importance of different modulation-spectral cues, we first obtained sentence-level intelligibility scores from a total of 22 adults. Ten participants (range: 20-32 years; mean: 24.5 years) formed the normal hearing group (NH group) as defined by air conduction thresholds of 20 dB HL or better at octave frequencies 250 Hz through 8 kHz. Twelve participants (range: 53-77 years; mean: 65.3 years) with mild to moderate cochlear hearing loss formed the group with hearing loss (HL). These participants listened to monaural presentations of IEEE sentences in their better ear (or right ear for participants in the NH group) processed through the impulse responses of 4 different rooms. In total, each participant listened to 528 sentences, including envelope expanded versions of these reverberant conditions. Sounds were presented at 70 dB SPL, with listeners with hearing loss receiving NAL-R-based amplification. These sounds were also analyzed to extract the quantitative measures of changes to modulation spectra. The relative importance of different acoustic cues were examined by fitting the different measures to participant intelligibility scores. The effectiveness of the reverberation-specific weighting of cues was also compared to the Hearing-Aid Speech Perception Index (HASPI) version 2 predictions of speech intelligibility. This analysis provides insights into the relative importance of different modulation rate filters and envelope vs fine structure changes on intelligibility of monaural reverberant speech subjected to hearing aid signal processing. [Work supported by a research grant from GN ReSound to the University of Colorado at Boulder.]

**BP324 Does the Type of Unilateral Beamforming Processing Affect Sound Location identification?**

Todd Ricketts*, Monica Folkerts1, Erin Picou1, Christopher Stecker2

1Vanderbilt University Medical Center, 2Boystown National Research Hospital

The ability to localize sounds in noise is affected by reverberation, separation between sound sources, and presence of visual information. In addition, hearing loss and hearing aids can both affect localization performance. For example, unilateral beamforming microphones typically affect the magnitude of interaural level differences (ILDs) in comparison to those that occur naturally at the unaided ear. Additionally, adaptive beamforming processing can also provide a variable ILD through adaptation to different listening environments (i.e. different noise locations in the rear hemisphere). While the effects of adaptive and fixed unilateral beamforming processing on ILDs are predictable, the relative effects of advanced hearing aid microphone processing on auditory localization in complex environments are less well understood. The purpose of the current study was to examine performance on a complex sound position identification task in listeners with hearing impairment when fitted bilaterally with commercially available unilateral beamforming hearing aids that were programmed for fixed asymmetric or adaptive processing.

Participants included adults with mild-moderate to moderately severe hearing loss who had at least 6 months of experience with one of the two types of microphone processing under investigation. For the purpose of the study, all participants were fitted bilaterally with two different brands of commercially available hearing aids using a common prescriptive gain procedure. In an anechoic chamber based virtual-reality environment, the participants identified the position of target talkers which was differentiated from a single distractor talker in the presence of a noise surrounding the listener. The noise consisted of cafeteria noise surrounding the listener and a high-level “jammer” noise (vacuum cleaner). The target and the distractor consisted of male and female talkers and the participant’s task was to click on the position of the targets virtual avatar. Talker and jammer separation and presentation angles were variable. All signals and noise were generated by a 64 loudspeakers array surrounding listeners in an anechoic chamber. The first 13 orders of reflection were used to simulate reverberation in a 10m X 10m room. Scoring was based on accuracy and latency when identifying the target on each trial. Preliminary results suggest both individual...
listener and processing-based differences in performance. Results and implications for these findings will be discussed.

**BP325 Multi-Dimensional Evaluation of User-Operated Audible Contrast Threshold (UACT) Tests by a Diverse Group of Participants**

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The user-operated audiometry (UAud) project aims at introducing an automated system for user-operated audiometric testing into everyday clinical practice. In that context, the Audible Contrast Threshold (ACT™) test is proposed as a test of supra-threshold hearing ability and as a language-independent alternative to speech-in-noise tests. Here, five user-operated ACT™ (UACT) test-paradigm candidates were evaluated in terms of performance and usability by 28 participants with diverse hearing and cognitive abilities. The five test candidates differed in the task and the procedure. UACT-0 and UACT-1 use a train of consecutive stimuli and the patient must indicate “when” the target was presented. UACT-2 through UACT-4 employ a sequential presentation of three separate intervals; UACT-2 uses a 3IFC task, where the patient must indicate whether the second interval contains target or reference (“which”), while UACT-3 and UACT-4 apply a 3AFC task, where the target is presented in one of the three intervals and the patient must indicate “where” the target stimulus was. The study was divided in two sessions conducted at least one week apart. To investigate the efficacy of non-verbal instructions (using pictograms), the participants underwent a block with the five tests in random order with exclusively non-verbal instructions. Three additional blocks were carried out for the purpose of evaluating test-retest reliability and training effects. Furthermore, the clinical audiologist-operated ACT™ and a 3AFC spectro-temporal modulation (STM, research baseline) test were performed for comparison in each session. The results showed that only 50% of the participants with lower cognitive abilities were able to provide reliable thresholds in the first block. The main factor affecting the thresholds estimated with the different tests was the task. All tests provided a reasonably good test-retest reliability, while the most reliable one was UACT-2 (3IFC task). UACT-2 also showed an excellent agreement with both the clinical ACT™ and the STM baseline tests. In terms of usability, the only significant aspect was the self-perceived duration. All participants reported a good experience with all test candidates with no significant differences among their judgements. Overall, UACT-2, the test based on a 3IFC with a custom adaptive procedure that includes catch trials, was chosen for being included in the UAud protocol as a user-operated test of supra-threshold hearing abilities. UACT will be part of a planned clinical trial investigating the effects of using audiologist-operated vs. user-operated audiometric tests on hearing rehabilitation.

**BP326 Sound Quality Assessment in Bimodal Cochlear Implant Users – Mechanisms of Hearing Aid Fitting**

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**Keywords:** Bimodal stimulation; sound quality assessment; MUSHRA; eGauge

**Objectives:** Many cochlear implant (CI) recipients use a hearing aid (HA) on the non-implanted ear. Current fitting rules of HAs are optimized on speech rather than music as input. However, a potential improvement might be the transmission of low-frequency acoustic sounds particularly targeting the requirements of music signals. In this project bimodally fitted CI-users are presented with music stimuli that are modified in a standardized and controlled manner and their individual sound quality judgements are collected. The goal is to find possible mechanisms that may be used for HA optimization based on either an individual or generic level.

**Design:** Non-linear signal processing of the acoustic path is simulated based on the individual pure tone audiogram of the participant using a generic fitting rule. Stimulation is provided by a HA receiver and the CI path transmits the unmodified stimulus via bluetooth to the subject's speech processor. The stimuli consist of acoustic only, electric only, and combined transmission considering several modifications of the acoustic path. Modifications are based on discrimination experiments conducted with the listeners. The MUIti Stimulus test with Hidden Reference and Anchor (MUSHRA) is used to assess sound quality for different stimuli such as classic and pop music with
or without lyrics. Outcome of the assessments is determined regarding test-retest reliability via the expertise Gauge (eGauge).

**Results:** Here we illustrate the methodology and present our first outcomes of the study.

**Conclusions:** Based on preliminary results the method seems generally suited for assessment of sound quality in bimodally fitted CI users and for providing information of optimized HA fitting. Increased low-frequency gain may be beneficial for some but not all music signals.

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**BP327 Hearing Aid use, Benefit and Satisfaction in adults: A Systematic Review**

Bopane Mothemela*, De Wet Swanepoel†, Vinaya Manchaiah‡, Megan Knoetze†

1University of Pretoria, 2University of Colorado School of Medicine

**Importance:** Understanding the factors influencing hearing aid outcomes can inform treatment approaches and patient support towards optimal benefit and satisfaction. There has been a growing body of evidence accumulating across various studies over the past decade but with the last review of literature done in 2010.

**Objectives:** This study systematically review peer-reviewed studies (published between January 2010 and January 2021) on factors influencing hearing aid use, benefit and satisfaction in adults.

**Methods:** This systematic review was conducted in accordance with the Preferred Reporting Items for Systematic Reviews and Meta-Analyses (PRISMA) guidelines 2020. A systematic literature search was conducted from the following databases (i) Web of Science (ii) Scopus, (iii) PubMed, (iv) EBSCOhost including CINAHL, and Academic Search Complete. All relevant articles were identified, exported to the Ryann Systematic review software and screened by two researchers independently. Full text copies of articles identified during the search and considered to meet the inclusion criteria were obtained for data synthesis. Periodical searches were conducted prior to the completion of the systematic review to assess for any further studies. The snowballing of the reference list method was also used to identify related articles that may have not been found during the initial search. Articles found through this exercise were included except for unpublished and non-peer reviewed publications.

**Results:** A total of 1022 peer-reviewed articles were identified from the following databases (i) CINAHL, (ii) PubMed, (iii) EBSCOhost including Web of Science, and (iv) Academic Search Complete. After removing 346 articles duplicates, the remaining 676 articles were further reduced based on outcomes, Implantable device, population, timeframe, publication, study design. A total of 33 articles were included for data extraction and analysis. From the articles reviewed, audiological factors that had significant influence on hearing aid outcomes included hearing sensitivity, speech perception, self-reported hearing disability, ear, tinnitus and balance, hearing aid acoustics and features, hearing aid candidate factors, hearing aid fitting and follow up. Non-audiological factors that showed a significant influence on hearing aid outcomes includes demographics, social networks, mental health, psychosocial, health and socio-economic factors.

**Conclusion:** Factors other than hearing loss and hearing technology are important for hearing aid use, benefit and satisfaction. The result of this study places an emphasis on patient-centered care, where non-audiological factors such as social networks, mental health, health, socio-economic should be considered to optimize hearing aid outcomes.

**BP328 Follow-Up study: Evaluation of a Non-Diagnostic Web-Based Hearing Screening Tool**

Joannes de Laat*, Louisa Enthoven†, Sjors van de Weijer‡

1Leiden University Medical Center, 2Eargo, Inc. USA

**Rationale:** Tele-audiology is a healthcare service delivery model that allows remote access to audiologists. The latest category of hearing aids, named over-the-counter (OTC) hearing aids, may soon make hearing aids directly available to consumers, without a visit to hearing health professionals. However, lack of awareness about hearing loss is potentially a key factor in the relatively low number of hearing aid users (e.g. ~3% of people worldwide with hearing loss, ~20% in developed countries). The Eargo Web Screener (powered by Clementine) is a non-diagnostic hearing screening tool specifically designed to give users general information about their hearing, remotely, from the comfort of their homes. The screener can be completed on mobile devices (e.g. mobile phone, tablet, laptop, and personal computer) using a variety of hardware (e.g. generic headsets, headphones, and/or earbuds). Initially, there is no need to travel to a clinic or interact with an audiologist, making it a low threshold tool to raise awareness about
hearing health amongst consumers and patients, consequently lowering the barrier to contact hearing health professionals.

**Objective:** The objective of this study is to investigate the feasibility and usability of the Eargo Web Screener in an adult population and to assess the hearing health screening capabilities of the Eargo Web Screener.

**Study Design:** Assessing usability of Eargo Web, and comparing the non-diagnostic Eargo Web hearing screening results to the audiograms of ISO 8253-1 compliant audiometric hearing assessments (gold standard hearing assessment). A hearing health questionnaire is administered to assess subjective audiometric profiles.

**Study Population:** A group of healthy adults with perceived light or mild to moderate hearing loss (target group), and a group of healthy young adults without hearing loss (control group).

**Main Study Parameters/Endpoints:** The system usability score for Eargo Web Screener and an evaluation of Web Screener performance.

**Preliminary Results:** Eargo Web Screener showed good usability scores and, although non-diagnostic, the Eargo Web Screener, showed adequate performance, when compared to golden standard audiometry.

**Conclusion:** Eargo Web Screener can be considered as a usable tool for raising awareness for hearing health in people with light or mild to moderate hearing loss, without the initial intervention of a hearing health professional. It is easy to complete, and thereby potentially increases awareness for hearing health in a larger group. Ultimately the Web screener may lower the barrier to contact a hearing health professional.

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**BP329 Experiences of Adult Hearing Aid owners: A Systematic Review of Qualitative Studies**

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Ilze Oosthuizen, University of Pretoria, South Africa

There has been an increasing number of qualitative studies exploring the experiences and perceptions of adult hearing aid owners throughout their hearing aid journey. As these studies and reported experiences vary greatly, a systematic review was conducted to identify and synthesize the key dimensions in adult hearing aid owners’ experiences. A systematic search of three electronic databases was conducted, yielding 1,130 results. Articles were evaluated for inclusion based on pre-determined eligibility criteria. After removing duplicates and screening for eligibility, data were extracted from 25 included articles. The quality of included articles was evaluated using the Rating of Qualitative Research scale. Guidelines of the Preferred Reporting Items for Systematic Reviews and Meta-analyses (PRISMA) and the Synthesis Without Meta-Analysis (SWiM) were followed. A narrative synthesis was conducted and studies were grouped into three main domains, namely experiences of owners related to: 1) hearing aid adoption fitting, 2) hearing aid use, and 3) hearing aid non-use or non-optimal use. The majority of owners’ descriptions of their lived hearing aid experiences contained dimensions related to facilitators for hearing aid use and to non-device related challenges that hinder hearing aid use. Specific service delivery related, hearing device related, and hearing aid owner related dimensions featured across all three main domains with relevant clinical implications. The synthesized dimensions that emerged from this review emphasize important aspects for hearing aid manufacturers and hearing care professionals to support optimal hearing aid intervention outcomes.
SESSION BX. Sig Soli Memorial Session

Session Chair: Sunil Puria

BX-1. In Memoriam: Dr. Sigfrid D. Soli

Susan Egerman

1Wife of Sig Soli

Sig was a member of the IHCON Steering Committee and an advisor. He served as organizational co-chair from 2000 to 2018 and was responsible for initiating the conference, which has been continually supported by NIDCD conference grants since 2000. At the time of his death, Sig was a Clinical Research Scientist at the House Institute Foundation in Los Angeles, California and an Adjunct Professor at the University of British Columbia in the department of Audiology and Speech Sciences in Vancouver, BC where he collaborated with School colleagues and students on several projects related to speech perception in noise.

In recent years, a major focus of his work involved the development of multilingual versions of the Hearing in Noise Test (HINT) to provide clinicians and their clients with a standardized tool that could be used for clinical assessment in a variety of languages.

Sig was born in Granite Falls, MN on May 15, 1946, the eldest son of Jane and Dan Soli, and died peacefully on April 11, 2022 after a short illness. Sig graduated from St. Olaf College in 1968, served as an officer in the US Air Force to work on early GPS technology, and then received his PhD in Experimental Psychology from the University of Minnesota (1979). Early in his career he became a distinguished professor in the Psychology Department at the University of Maryland.

He returned to Minnesota in 1984 to accept a position as a research scientist at 3M where he worked on the early development of cochlear implants and hearing aid technologies. In 1989, his research took him to Los Angeles where he joined a team of global experts at the House Ear Institute. As a principal contributor to the development of cochlear implants, he played a significant role in the creation of tools and methodologies that enhanced the lives of individuals with hearing loss. During his 23 years at the House Ear Institute, Sig traveled extensively in Europe, South Korea, Japan, and China to share his expertise with other professionals in his field. His trips with Sunil Puria to Daegu and connections with Prof Jin-Ho Cho at Kyungpook National University in South Korea, in addition to significant advances in hearing research, led to the transfer of the conference to Massachusetts Eye and Ear, and Dr. Puria becoming its chair.

Sig and his wife, Susan, raised two sons, Andrew and Daniel, in Sierra Madre, California. While living there, Sig became an avid fan of the Los Angeles Dodgers, and he regularly attended baseball games with his friends and family. Sig also enjoyed taking his family on camping trips and to international destinations to expose his loved ones to the outdoors and to the rich cultures around the world. Sig had enthusiasm for both research and teaching. He authored over 65 scientific publications and holds over 30 US and international patents. HINT has been developed in more than 20 languages.

Later in his life, Sig spent much of his time in British Columbia, Canada with annual travel to Jacksonville Beach, Florida and Kauai, Hawaii. While living in British Columbia, Sig enjoyed the natural beauty of the surrounding area and visits from his family, friends, and colleagues. He would happily point out the resident bald eagles and harbour seals between heartwarming stories of his latest research collaborations.
**BX-2. Sig Soli’s Personal Story (Live via Zoom)**
*Susan Egerman*\(^1\)
\(^1\)Wife of Sig Soli

**BX-3. Sig Soli and Speech Recognition in Noise Research (Live via Zoom)**
*Andrew J. Vermiglio*\(^1\)
\(^1\)East Carolina University

**BX-4. Discovery and venture into multilingual HINT (Prerecorded)**
*Lena Wong*\(^1\)
\(^1\)University of Hong Kong

**BX-5. Sigfrid Soli’s Pioneering Contributions to Hearing Screening for Hearing-Critical Jobs: A Canadian Perspective (Prerecorded)**
*Chantal Laroche*, *Veronique Vaillancourt*, *Christian Giguere*\(^1\)
\(^1\)University of Ottawa

**BX-6. The South Korean connection and how I came to be the chair of IHCON (In-person)**
*Sunil Puria*\(^1,2\)
\(^1\)Mass Eye and Ear, \(^2\)Harvard University

**BX-7. Open mic (In-person)**

12:30 p.m. – 1:30 p.m. Lunch

1:30 p.m. – 5:00 p.m. Leisure

5:00 p.m. – 7:00 p.m.

**SESSION B2. New Approaches to Aided Outcome Assessment**
Session Chair: Jörg Buchholz

**B2-1. Hearing aid outcomes beyond speech intelligibility**
*Frank Russo*\(^1\)
\(^1\)Toronto Metropolitan University

The central outcome of most hearing aid studies tends to be speech intelligibility in quiet or in noise. Increasingly, however, clinicians and hearing aid users are interested in realizing improvements in other areas of auditory perception. In this presentation, I will focus on hearing aid outcomes concerning the perception of music and vocal emotion. After setting the scene by reviewing some fundamentals from the perspective of acoustics, psychoacoustics, and the brain, I will consider the extant research that has considered music and vocal emotion in hearing aided listeners. Relative to speech, live music challenges the limits of hearing aid processing with a higher input level, larger frequency bandwidth, and larger dynamic range. Not surprisingly, deficits have been observed in hearing aided listeners relative to normal hearing insofar as pitch, melody, and timbre perception are concerned, though aided conditions have been shown to lead to improvements over unaided conditions. While vocal emotion does not present the same challenge as music at the signal level, there is clearly a deficit in hearing aided listeners relative to normal hearing listeners, and several studies have found that aided conditions do not yield improvements over unaided conditions. This apparent failure of amplification may arise from shortcomings in hearing aid processing, but it may also be due to limitations in the
methods used to study emotion, and/or compensatory neuroplastic changes in the brain that are not easily undone following amplification. I will sum up with an agenda for future research and considerations for future development of hearing aids.

**B2-2. Impact of Amplification on Intra-Individual Emotion Processing**

Jani Johnson\*1, Lipika Sarangi1

1The University of Memphis

Although acquired hearing loss can negatively impact emotional processing, there is limited research exploring how amplification through hearing aids (HAs) might compensate for these deficits. Available evidence on emotional processing is mostly restricted to measures obtained under controlled laboratory conditions; however, social-emotional factors that modulate this outcome are difficult to replicate in contrived environments. The current study aimed to clarify the real-world effects of premium-feature HAs on inter- and intra-individual emotion processing in older adults.

Thirty individuals aged 50-78 years with bilateral, uncomplicated, mild-to-moderate hearing loss, and no experience with HAs participated in this ABA repeated reversal trial. Participants completed an unaided baseline trial, an aided trial, and an unaided withdrawal trial. Speech communication outcomes and self-reported and physiologic measures of intra-individual emotion perception were assessed for each arm of the study. Repeated measures with corrections for pairwise comparisons and multilevel linear mixed model analyses were performed to explore differences with and without HAs.

In addition to improved speech communication outcomes, participants reported greater arousal with hearing aids especially in quieter environments both in daily listening and in the lab. Increased heart rate and respiration activity supported self-report measures when assessed in daily listening. However, physiologic changes were not reflected in laboratory conditions. Participants reported feeling more pleasantness when wearing hearing aids in daily listening environments when speech was present, but less pleasantness when listening to nonspeech sounds or in background noise. The impact of amplification on this dimension of emotion processing was not clearly observed in the lab. Finally, and possibly most importantly, peripheral physiologic measures indicated a healthier autonomic nervous system in daily listening when wearing hearing aids, suggesting better physical recovery from stress.

This study confirms the notion that premium level HAs can improve emotion processing in the real world and suggests that these and other positive impacts of wearing hearing aids might result in physiologic improvements for those with hearing difficulties. Further, these results suggest that emotional processing might best be reflected using measures in daily listening, including in-situ questionnaires and wearable sensors to capture the impact of salient situational, social-emotional, and motivational factors on experiences that are difficult to replicate in a laboratory environment.

**B2-3. Recognition of Vocal Emotions in Children with Hearing Aids**

Laura Rachman\*1, Gizem Babaoglu2, Başak Özkişi Yazgan3, Pınar Ertürk2, Etienne Gaudrain1, Leanne Nagels1, Stefan Launer4, Gurjit Singh5, Hannes Wüthrich6, Peter Derleth4, Frédérick Jehle4, Monita Chatterjee6, Esra Yücel2, Gonca Sennaroglu2, Deniz Başkent1

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Correctly interpreting the emotions of another person is an important part of daily communication, and especially for children, it is also crucial for their social development. Despite this
importance, previous research on vocal emotion perception in children with hearing aids (HAs) has been limited, mostly due to difficulties of having adequate test materials and test methods, but also because of the assumption that these children should be able to hear vocal emotions once the sounds are amplified with their devices. In children with normal hearing (NH), vocal emotion recognition continues to develop throughout childhood. In children with hearing loss, decreased audibility and potential loss of sensitivity to relevant acoustic cues may affect vocal emotion perception, but it is not clear if, and to what degree, children with HAs may have difficulties in perceiving vocal emotions, compared to their peers without hearing loss. Additionally, although HAs are typically optimized for speech perception, it has not previously been studied whether HA settings may affect acoustic cues that are relevant for vocal emotion perception.

In the current study, we used the EmoHI test, previously developed in our lab (Nagels et al., 2020, doi: 10.5281/zenodo.3689710), to assess emotion recognition abilities of native-Turkish children with HAs and children with NH between 5 and 18 years of age. Furthermore, we collected data from NH adults between 18 and 35 years of age. The EmoHI test materials consist of non-language specific pseudospeech sentences, produced by various talkers and expressing three core emotions: happiness, anger, and sadness (Nagels et al., 2020, doi: 10.7717/peerj.8773).

In addition to behavioral data, we collected data about individual HA settings from a subset of the participants to investigate whether emotion recognition abilities may have been affected by specific HA settings. We will present exploratory acoustic analyses from HA simulations of the EmoHI test materials. Finally, the current behavioral data will be compared to previously collected data using the same paradigm in children from the Netherlands and the United Kingdom for cross-language comparisons. [Funding: Phonak AG, Stäfa, Switzerland; VICI grant 918-17-603 from the Netherlands Organization for Scientific Research (NWO) and the Netherlands Organization for Health Research and Development (ZonMw); the Heinsius Houbolt Foundation; and a Rosalind Franklin Fellowship.]

B2-4. Cortical Sensory Gating for (non)speech as a Predictor of Noise Acceptance in Older Normal Hearing and Hearing-Impaired Listeners Tested in the Aided Mode.

Christopher Slugocki*, Francis Kuk†, Petri Korhonen‡

1WS Audiology - ORCA-USA, 2ORCA US, WS Audiology

Listening to speech in noise is often problematic for hearing aid wearers. However, at present, we do not understand the source(s) of a considerable variability in listeners’ noise acceptance thresholds. Such knowledge might otherwise help to refine the development/deployment of noise management systems.

Sensory gating refers to a phenomenon that occurs when continuous, habitual, or redundant sensory information is blocked such as to result in reduced conscious processing (Korzyukov et al., 2007). Auditory cortical gating has been measured with a paired-stimulus paradigm, wherein cortical auditory-evoked potentials (CAEPs; e.g., P1-N1-P2) are measured in response to two brief sounds (tone pips, speech syllables, etc.) separated by a short inter-stimulus interval (< 1 s). Pairs themselves are separated by a longer period (8–10 s). The degree of cortical gating is then quantified by the reduction in component morphology evoked by the second sound relative to the first (Korzyukov et al., 2007). Using this paradigm, Miller et al. (2018) showed that cortical sensory gating accounted for a significant portion of variability in Acceptable Noise Level scores (ANL; Nabelek et al., 2004)—the current de facto method of assessing noise acceptance—among normal hearing listeners.
Previously, we introduced the Tracking of Noise Tolerance (TNT) test (Seper et al., 2020) as a clinically oriented alternative to the ANL for measuring acceptance of noise during speech listening. The TNT uniquely measures how noise acceptance changes over time due to the action of hearing aid features/systems. Moreover, we have shown the TNT test to have good test-retest reliability and predictive power for hearing aids satisfaction in loud noisy environments (Seper et al., 2020).

Here, we present work (in progress) exploring the power of the TNT as a listener profiling tool by measuring whether TNT scores can be predicted from the strength of auditory sensory gating in a group of older normal hearing adults (n = 24) and listeners with a hearing loss (n = 16) tested in the aided mode. We expect listeners with lower TNT scores (i.e., less tolerant of noise) to exhibit less auditory sensory gating compared to listeners with higher TNT scores (i.e., more tolerant of noise) and for this relationship to be possibly mediated by hearing status. Gating measured with tone pips versus speech sounds will also be compared in each group of listeners and evaluated separately for predicting noise acceptance.

**B2-5. Effect of Hearing Aid Directional Processing on the Neural Encoding of Spatial Changes in a Speech Stream**

_Erol Ozmeral*, Dana Cherri*

*University of South Florida*

**Background:** Selective attention preferentially highlights auditory objects of interest by applying sensory gain control that modulates the representation of relevant and/or irrelevant features. It is known that hearing-impaired listeners localize speech poorly in a mixture but not in quiet, leading to the possibility that spatial tuning is disrupted by the presence of background signals. It is unclear, however, whether improved SNR (via hearing aid directional processing) improves spatial tuning to auditory objects, and if so, whether this facilitates sensory gain control.

**Method:** In this study, hearing-impaired listeners (n = 20) were tested with and without aids on a spatial localization task. Cortical event-related potentials were simultaneously measured using EEG to investigate spatial tuning of speech in quiet or babble. Further, active attention to a location was tested to determine whether spatial encoding is enhanced with fixed directional processing relative to omnidirectional processing.

**Hypotheses:** Spatial tuning is broader for hearing-impaired listeners because of poor object formation when maskers are present, and directional processing can improve object formation and subsequent spatial encoding of speech. Second, whereas poor object formation impairs modulatory effects of selective attention, directional processing can restore this sensory gain control mechanism.

**Results:** There were poorer overall responses to speech changing location when background babble was present; however, we observed significantly larger P3 amplitudes with a beamformer fixed to the front compared to omnidirectional mode when listeners were aided.

**Conclusions:** Likely due to improvements in SNR, beamforming shows a neural advantage over omnidirectional microphones when listening to speech in background babble.
Friday, August 12

7:00 a.m. – 8:00 a.m.    Breakfast

8:00 a.m. – 9:20 a.m.

SESSION C1. Part I: New Approaches to Hearing-Aid Fitting and Services Delivery

Session Chair: Peggy Nelson

C1-1. Toward a Self-Empowered Wellness Model of Hearing Healthcare in Adults

_Larry Humes*_1  
1_Indiana University_

The 2021 World Report on Hearing from the World Health Organization (WHO) estimates that more than 1.5 billion people, 20% of the global population, have hearing loss. Of these, most have either mild (1.15 billion) or moderate (266 million) hearing loss. The WHO report also notes that the prevalence of hearing loss worldwide increases with age with nearly 50% of people 60 years of age and older having mild hearing loss and 25% having hearing loss that is moderate or greater. Clearly, mild-to-moderate hearing loss in adults is a highly prevalent health concern. This is true, moreover, not just because of the direct negative consequences of the hearing loss on everyday communication but because of the well-established impact of that hearing loss on psychosocial well-being and cognition.

For adults with mild-to-moderate hearing loss, the most common treatment is the provision of devices, such as hearing aids, designed primarily to restore sound that has been rendered inaudible by the hearing loss. Contemporary devices are of high quality and can restore audibility over a broad frequency range maintaining a comfortable loudness in the process. Positive outcomes have been frequently documented for well-fit hearing aids in adults with mild-to-moderate hearing loss.

Despite the widespread presence of mild-to-moderate hearing loss in adults and the positive outcomes that have been demonstrated for intervention with hearing aids in this population evidence suggests that less than 20% of those who could benefit from hearing aids seek them out and use them. The problem of limited uptake and use of hearing aids has been largely attributed to shortcomings of the prevailing service-delivery model, primarily poor accessibility and affordability. The prevailing service-delivery model is based on the identification and treatment of hearing loss by a hearing healthcare professional. This presentation will describe a self-driven auditory-wellness model and will present evidence in support of its viability for adults with mild-to-moderate hearing loss.

C1-2. Comparing Hearing Aid Outcomes for Emerging Direct-To-Consumer and Hearing Care Professional Service Delivery Models

_De Wet Swanepoel_1, Ilze Oosthuizen*_1, Marien Graham_1, Vinaya Manchaiah_2  
1_University of Pretoria, South Africa, 2_University of Colorado School of Medicine, Aurora, Colorado, USA_
More accessible and affordable hearing aids are in view with the pending over-the-counter hearing aid regulations in the United States. While laboratory studies have validated many over-the-counter hearing technologies there is limited real-world benefit studies of these technologies and different service-delivery models. The aim of this study was to compare hearing aid outcomes reported by clients receiving hearing aids through an emerging direct-to-consumer and conventional hearing care professional service-delivery models.

A prospective cross-sectional survey design was followed and an online survey was sent during October and November 2021 to Hearing Tracker user database and to the direct-to-consumer Lexie hearing aid user database. 656 hearing aid users completed the survey; 406 through conventional hearing care professional services and 250 through the direct-to-consumer model.

Self-reported hearing aid benefit and satisfaction was measured with the 7-item International Outcome Inventory – Hearing Aids using a 5-point Likert scale. No significant difference for overall hearing aid service-delivery outcomes between hearing care professional and direct-to-consumer users were evident using regression analyses controlling for age, gender, duration of hearing loss, duration before hearing aid purchase, self-reported hearing difficulty and unilateral versus bilateral hearing aid fitting. For individual questions there were no significant differences (p > .05) on items of Benefit, Satisfaction, Residual participation restriction, Impact on others and Quality of life. For Daily use hearing care professional clients reported significantly longer hours of daily hearing aid use (OR = 0.50; 95% CI, 0.33 to 0.75; p < .001). For residual activity limitation direct-to-consumer hearing aid users reported significantly less difficulty hearing in situations where they most wanted to hear better (OR = 2.50; 95% CI, 1.80 to 3.45; p < .001).

Direct-to-consumer hearing aid outcomes could complement and provide similar satisfaction and benefit to hearing care professional models. Self-fitting, acclimatization programs, remote support, behavioral incentivization and payment options should be investigated for their potential role in direct-to-consumer/over-the-counter hearing aid outcomes.

C1-3. Insights into Fitting Minimal Hearing Losses

Brent Edwards*1, Jorge Mejia2, Joaquin Valderrama-Valenzuela1, Nicky Chong-White1

1National Acoustic Laboratories, 2National Acoustic Laboratories, Sydney Australia

A challenge that hearing healthcare professionals (HHPs) face every day is what hearing health recommendation to make with each patient. Whether to recommend a hearing aid is often based primarily on the level of hearing loss as measured by the audiogram, possibly in combination with a speech test and needs consultation. Recent research, however, suggests that audiograms are insufficient indicators of need, leaving HHPs with the challenge of determining who to make device recommendations to. This is particularly challenging for people who present with self-reported hearing difficulty but very little measurable hearing loss (PTA less than 25 dB HL).

This talk will answer two questions on this topic: Should HHPs recommend hearing devices to people with no-to-mild earing loss, and how well can hearing devices benefit people with no-to-mild hearing losses? This discussion will include the context of the current hearing healthcare environment, including emerging technology innovations. Results from several research studies conducted at the National Acoustic Laboratories will be presented.

In the first study, participants with no measurable hearing loss but with speech in noise complaints were fit with hearing aids in two groups: one with the hearing aid fully featured and one with placebo devices. Participants wore devices for six weeks and data was gathered through laboratory measures, questionnaires and an environmental momentary assessment app. In the second study, a similar cohort of participants were fit with Apple AirPods Pro with a similar protocol and set of outcome measures. In a third study, electroacoustic measures were obtained.
with AirPods Pro in Conversation Boost mode, a feature that provides similar functionality to hearing aids, and the results compared to traditional hearing aid performance and the NAL-NL2 fitting algorithm.

Implications from all of these investigations to the provision of hearing healthcare to people with minimal hearing loss will be discussed.


CP101 Community-Based Adult Hearing Care provided by Community Healthcare Workers Using mHealth Technologies

Caitlin Frisy*, Robert H. Eikelboom2, Faheema Mahomed-Asmail3, Hannah Kuper4, Tersia de Kock4, Vinaya Manchaiah4, De Wet Swanepoel1
1University of Pretoria, 2Ear Science Institute Australia, 3London School of Hygiene and Tropical Medicine, 4hearX Foundation, 5University of Colorado School of Medicine

Background: Rising prevalence rates of hearing loss is of concern due to the global shortage of hearing healthcare services. Task-shifting to community healthcare workers (CHWs) supported by mHealth technologies has been recommended to overcome the lack of services.

Objective: This study evaluated the feasibility of a community-based rehabilitation model for providing hearing aids to adults in low-income communities by using CHWs supported by mHealth technologies.

Method: An implementation approach evaluated the hearing assessment, hearing aid fitting, and support process for adults with hearing loss in two low-income communities in the Western Cape, South Africa. CHWs facilitated hearing assessments and hearing aid fittings, after training by audiologists, over a period of 13 months from September 2020 to October 2021. Data was gathered using qualitative and quantitative measures and analysed using a mixed methods approach. Hearing aid outcomes were measured using the International Outcome Inventory – Hearing Aids.

Results: 148 of 152 adults in the community who self-reported hearing difficulties were successfully tested by CHWs during home visits. Most had normal hearing (39.9%) with 24.3% having sensorineural hearing loss bilaterally, 20.9% with suspected conductive hearing loss and 14.9% with unilateral hearing loss of which 5.4% was a suspected conductive loss. 40 adults met the inclusion criteria to be fitted with hearing aids of whom 19 were fitted with hearing aids bilaterally. Positive hearing aid outcomes and minimal device handling challenges were reported at 45 days post-fitting and was maintained at six months with no significant changes. CHWs were able to support and resolve minimal challenges.

Conclusions: Implementing a hearing healthcare service-delivery model in the community facilitated by CHWs is feasible. mHealth technologies used by CHWs enables a scalable service-delivery model for improved access and affordability in low-income settings. Future studies should compare this innovative model to conventional hearing care professional service-delivery models.

CP102 Hearing Aid Self-Adjustment methods: Adjusting Hearing Aid Gain and Noise Reduction Algorithms

Jonathan Goesswein*, Jan Rennies1, Birger Kollmeier2
1Fraunhofer IDMT, Oldenburg Branch HSA, Oldenburg, Germany, 2Department of Medical Physics and Acoustics, Carl von Ossietzky University of Oldenburg, Oldenburg, Germany

Self-adjustment of hearing aids (HAs) is subject of recent research, it should make adjustments more user-friendly and faster. In the current study we review a sequence of experiments with the aim to optimize the methods and user interfaces (UIs).

The fitting process of HAs is very time consuming and does not always lead to a satisfactory result. Typically, the process starts with an audiogram measurement, on which basis a prescription gain rule is calculated. This prescription is then further fine-tuned by the audiologist based on the patient’s reported perception. One possible solution of improvement is to involve the HA user more in the fitting processes by means of self-adjustment. In one study a two-dimensional (2D) graphical UI was utilized to reasonably reduce the available parameter space for the naïve HA user to self-adjust. The results of this study prove to be reliable and fast, while big interindividual variabilities were observed. This variability raises the question if a different starting point than the audiogram-based prescription could lead to similar preferred self-adjustment results. In the current study we aim to replace the audiogram-based prescription with an iteratively designed self-adjustment method using the
evaluated 2D graphical UI. The initial parameter space is designed around an estimated prescription based on the sex and age of the HA user. With every iteration the size of the parameter space is reduced according to the HA user’s chosen preferences until—in the smallest and final parameter space—the final HA gain setting is determined.

Another aspect of HA fitting is the adjustment of noise reduction algorithms. Noise reduction algorithms are a double-edged sword: the more noise they suppress, the more distortions they create. Individually fitting these algorithms is crucial because HA users are known to differ strongly in their noise and distortion tolerance. Regarding this tolerance, two personal traits are described in the literature: noise haters who consistently prefer a rather strong noise reduction despite the distortions and distortion haters who consistently prefer a very moderate noise reduction to avoid these distortions. In the current study we aim to distinguish between these two personal traits and predict a preferred noise reduction setting based on a limited set of self-adjusted preference settings.

Overall, we demonstrate feasibility and the relevant factors and interindividual variabilities to be observed when designing self-adjustment strategies for HAs.

**CP103 Remote Audiological assessment: The Hearing-Aid Listening Test**

*Gaston Hilkhuysen*, Tim Green, Stuart Rosen, Mark Huckvale

1UCL

The COVID19 pandemic forced audiology to reshape its hearing assessment procedures. The elderly, the age group with the highest incidence of hearing problems, were particularly prone to develop complicated Covid. Lock downs brought clinical and scientific evaluation of their hearing almost to a stand-still. The Hearing-Aid Listening Test (HALT), an android application running on a tablet computer with specific headphones, circumvented these problems while providing audiological assessment in the home environment.

HALT’s calibrated test equipment consisted of a pair of circumaural headphones (Superlux HD572) and a tablet (Samsung SM-T500) capable of reproducing the high sounds levels generated by hearing aids. A customized app on the tablet provided an audiometer and procedures for testing the intelligibility of speech processed by hearing aids.

Test kits were sent out to participants. A participant would power up the tablet at pre-arranged time. At that moment, the experimenter took remote control of the tablet and established a video-conferencing connection. The participant was informed, instructed, and tested as in the typical laboratory setting. Even participants with few computer skills could contribute. They only needed to connect the headphones to the tablet and have the tablet connected to their WiFi. The latter was sometimes accomplished by someone else in their household.

Although the pandemic appears to be receding, HALT remains an attractive platform for hearing assessment. When applied in research it provides access to populations from much larger geographic areas than the traditional ring around the lab. Participants no longer need to travel to the laboratory, facilitating extensive data collection across multiple sessions spanning several days. These benefits also hold for its potential clinical application. Once extended with self-administered test modules, HALT can give hearing-care providers valuable supplemental information, presently ignored due to time constraints on consultations.

**CP104 A Touchscreen-Based Self-Fitting Procedure for Hearing Aids and Initial Evaluations**

*Bertan Kursun*, Chemay Shola, Lauren Langley, Yi Shen

1University of Washington, 2Technical University of Denmark

Self-directed gain adjustments may present a plausible solution for fitting a hearing-aid (HA) without the requirement of audiology test and real-ear verifications. The current study evaluates the feasibility of a self-fitting procedure, in which the user interacts with the HA using a touchscreen-enabled mobile device, while a continuous speech stimulus with a background noise is played in the sound field. The user explores the locations on a 2D surface by dragging a point on the touch screen while hearing its effect on the hearing-aid processed audio in real-time and identifies a preferred setting. This adjustment process is repeated over 30 trials, with the mathematical map between the touch-screen location and the amplification profile of the HA updated from trial to trial. The final estimate is the mean of the user identified gain settings of the last 25 trials. Using this procedure, self-directed fitting was performed by six older adults with mild-to-moderate hearing loss. Following self-fitting, the participants were also fitted with the same hearing aids using the best clinical practice and a
standardized gain prescription method (NAL-NL2). The gain profiles obtained from self-fitting and traditional fitting resembled each other, though considerable and inhomogeneous deviations between the self-adjusted gains and the NAL-NL2 prescription were observed. The Speech Intelligibility Index was not found to be significantly different between self-fit and NAL-NL2. In addition, a paired comparison test did not show a consistent preference of the participants toward either of the two fits.

**CP105 Real-World Evaluation of a Self-Fitted Hearing Aids**

*Jorge Mejía*, Alex Meera, Alexandre Thompson, Arun Sebastian, Jessica Cooper, Catherine Morgan

*National Acoustic Laboratories, Sydney Australia*

Self-fitting hearing aids allow users to perform a hearing threshold assessment to produce a prescribed amplification setting, without audiology support. The primary goal of these devices is to expand and diversify hearing intervention options available to adults with hearing loss and increase the uptake of hearing aids. Increasingly, self-fitting hearing aids are made available directly to consumers. However, there is limited evidence to support the efficacy of self-fitted hearing aids in meeting the communication needs of users. We conducted a clinical study to examine the real-world benefit of a self-fitted hearing aid. Forty-one participants were recruited to attend two appointments. At the first appointment, participants were tasked to self-fit the Nuheara IQbuds™ 2 PRO hearing aids using the propriety Ear ID™ software, which provided NAL NL2 amplification targets. Comparative assessments were conducted at the same appointment, a qualified audiologist administered a standard hearing threshold test and real-ear measures of insertion gain. Finally at the first and second appointments, participants also completed questionnaires, Abbreviated Profile for Hearing Aid Benefit (APHAB), and the short form of the Speech Sound Quality (SSQ12) and performed hearing in noise and sound quality rating tests of a target talker in diffuse listening conditions. In between appointments, lasting between 1 to 4 weeks, participants were tasked to evaluate the hearing aids, and perform daily surveys of performance using an Ecologically Momentary Assessment (EMA) app developed by the National Acoustic Laboratories. The EMA app also captured acoustic scene information at the time the survey was conducted. Herein we report our laboratory and real-world outcomes and discuss clinical implications.

**CP106 Over-The-Counter Hearing Aids Challenge the Core Values of Hearing Healthcare**

*Katherine Menon*, Michelle Hoon-Starr, Katie Shilton, Eric Hoover

*University of Maryland - College Park*

Regulatory changes in the United States introduced over-the-counter (OTC) hearing aids with the goal of increasing the accessibility and affordability of hearing healthcare. These changes represent a shift away from the provision of hearing aids by licensed audiologists and hearing instrument specialists. If the proposed solution does not share the same values as the current model, then it may fail according to existing metrics (e.g., poor fit-to-target) and still be highly successful by its own metrics (e.g., devices on more people). Our recent work identified the values of hearing healthcare service delivery. In this study, we evaluated the relative importance of these values across service delivery models, and the extent to which regulatory changes represent a coherent re prioritization of values.

We performed a qualitative content analysis of two document categories: critique documents representing the motivation to create an OTC model, and regulatory documents governing the implementation of OTC. Team members coded portions of text for the values they expressed. In total, 29,235 words were coded across 72 pages in four documents. Rank-order analyses were performed to determine the relative importance of values within each category of documents, between document categories, and in comparison to the existing model.

We observed a strong association between the rank order of values within each category, indicating that values are internally consistent in both critique and regulatory documents. Comparing between categories, the rank order of values in the regulatory documents was largely inconsistent with the critique documents, suggesting that the OTC model does not address the barriers that motivated its creation. Differences in the rank order of values in the regulatory documents compared to the existing model showed that the OTC model represents a values shift, but it remains unclear what values are prioritized by the OTC model. In order to evaluate the extent to which regulatory changes improve hearing healthcare, we need to establish the values of the new model through a consensus of stakeholders, including underserved individuals from diverse backgrounds.
**CP107 Self-adjustment versus prescriptive fitting: How much of a difference really makes a difference?**

*Dana Urbanski*1*, Peggy Nelson1, Sophia Donato1, Joyce Rosenthal2, Jayaganesh Swaminathan3*

1University of Minnesota, 2Eargo Inc.

Many contemporary hearing aids have self-adjustment algorithms as an integral part of their technology. Questions remain, however, about end users’ ability to fit gain profiles appropriately to achieve satisfactory outcomes and clinical efficacy.

One such device is Eargo’s hearing aid that allows end users to program devices by performing in-situ hearing screening and self-adjustments using a smartphone application. In the current study, we are recruiting adults aged 18 to 85 with perceived mild-to-moderate sensorineural hearing loss to test Eargo’s hearing aids with research software that allows an audiologist to adjust gain parameters. About 30 participants are recruited for a single-blind, crossover study testing two fitting methods in randomized sequence: A) prescriptive fit based on the participant’s clinical audiogram and fit to NAL-NL2 real-ear aided response targets; B) self-adjusted by participants using Eargo’s Sound Match in-situ hearing screening and self-adjustment controls. Participants are enrolled and randomly assigned to group 1 or 2, which determines the sequence of the two methods. Users wear the study devices programmed to the first method for 2-3 weeks and return for outcome testing (including real-ear measures, AzBio sentence recognition in noise, and APHAB questionnaire). Participants then wear the devices programmed to the second method, followed by 2-3 additional weeks of in-field use. At the third and final laboratory visit, participants complete outcome measures and sound quality ratings.

Preliminary data on the Eargo Sound Match system indicate good usability and selection of audiometrically appropriate gain profiles. Data from the first several participants indicate that self-selected gain profiles are within 5 dB of prescriptive fittings from 500 to 4000 Hz. When the study is complete, data should reveal results relevant to current clinical questions, including:

- How large are the differences between self-selected and prescriptive-fit gain profiles in this group using this device and algorithm?
- How much difference in audibility for realistic situations would result from these differences?
- How do these differences translate into meaningful differences in outcomes such as sound quality, speech understanding in noise, and perceived benefit?

Ultimately, we aim to learn more about the amount of the ‘difference-from-target’ that can result in positive outcomes for adults with mild-to-moderate hearing loss. The results of this study may inform design and development of self-adjusting hearing aids grounded in hearing science and clinical audiology. [This work is supported by a grant from Eargo to the University of Minnesota Center for Applied and Translational Sensory Science (CATSS).]

**CP108 What is audiologic counseling and what could it be?: A mixed-methods study of adult hearing aid counseling and possible adaptations for remote service models**

*Dana Urbanski*1*, Erin O’Neill2, Randi Rankl1, John Ellison2, Peggy Nelson1*

1University of Minnesota, 2GN Advanced Science

According to a concept promoted by Freston, innovation is taking two things that exist and putting them together in a new way. The emergence of over-the-counter (OTC) amplification underscores this idea, as OTC manufacturers pair technological advances with elements of traditional, brick-and-mortar clinical audiology to create new solutions. Specifically, hearing scientists and professionals have shown interest in leveraging telehealth and mobile applications for delivery of audiologic counseling outside of formal clinic visits. To achieve this aim, we must define the core elements and desired outcomes of audiologic counseling in everyday clinical settings.

Toward this objective, we designed a mixed-methods study to examine audiologists’ implementation of adult hearing aid counseling in dispensing clinics. We are recruiting 20-25 audiologists from a variety of practice settings to complete semi-structured interviews and ecological momentary assessment (EMA) surveys. Interview questions explore how audiologists define, implement, prioritize, evaluate, and individualize counseling activities, along with audiologists’ visions for change/innovation in counseling service delivery. Interviews are conducted using video conferencing and transcription, quality-checked for accuracy. After establishing interrater reliability, two research team members will code interview transcripts using inductive thematic analysis. EMA surveys collect quantitative data categorizing the type and frequency of audiologists’ daily counseling activities and barriers/facilitators to counseling.
Participants complete daily EMA surveys using ExpiWell, an EMA smartphone application. EMA data will be summarized in descriptive statistics and interpreted relative to identified qualitative themes.

Early qualitative results reveal several key themes: a) counseling is time intensive; b) counseling is a process that takes place over several interactions; c) counseling is most effective when tailored and sequenced to patient characteristics including experience and attitudes; and d) counseling has an observable positive impact on adult hearing aid outcomes. Early EMA data confirm the time demands of effective counseling, with several participants citing appointment length as a daily barrier to counseling. EMA results reveal that audiologists spend substantial time reviewing basic device use, care, and maintenance, including smartphone applications/streaming, sometimes to the exclusion of other counseling activities. Audiologists highlighted ways in which mobile wireless technologies might improve and augment in-person counseling, including reminders/notifications for cleaning, care and maintenance, user-friendly availability of instructional videos, and delivery of listening situation-specific communication and hearing aid strategies.

This poster will present results of our full participant sample with data collection and analysis slated for completion in early-mid Summer 2022.

CP109 How’s what sound: Challenging misassumptions in the personalisation of hearing aids

William Whitmer1*, David McShefferty1, Benjamin Caswell-Midwinter2, Graham Naylor2

1Hearing Sciences - Scottish Section, University of Nottingham, Glasgow, UK, 2Otolaryngology - Head and Neck Surgery, Massachusetts Eye and Ear, Harvard Medical School, Boston, MA

In the personalisation of hearing aids, it has long been common practice for the clinician to adjust gain away from prescription based on feedback from the patient. Underlying this personalisation process, though, are several assumptions about the perception of those adjustments, from the adjustments being adequately large enough and the stimuli adequately long enough to elicit a reliable preference to the feedback being reliable within and across patients. Through a series of psychophysical studies, we have explored the viability of those assumptions.

In the first study, we examined the role of duration in the scale of gain adjustments necessary to elicit reliable preferences from patients. Twenty-nine hearing-aid users judged whether varied adjustments in broad frequency bands away from their individual settings were better, worse or no different when listening to speech segments of varying duration. The necessary gain adjustment to elicit a preference decreased and became more reliable with increasing stimulus duration, but the effect was limited, and the scale of the adjustment at longer durations was still larger than suggested by current guidelines. That is, speaking longer can lead to better fits, but only by so much.

In the second study, we examined the assumption of reliable patient feedback in fine-tuning, and across patients. Twenty-eight online participants with minimal-to-mild hearing loss described (open-response) the difference of gain adjustments away from median prescription settings. While participants generally used some terms associated with hearing-aid gain, their usage, with “quiet” and “loud,” lacked reliability within or across participants. That is, particular descriptors were not associated with any particular adjustment.

These findings highlight some of our misassumptions in the personalisation of hearing aids, casting doubt on the efficacy of current fine-tuning practice and indicating that alternative approaches to personalising hearing-aid settings may be a more viable use of valuable clinical time.

CP110 Verification and Validation of a Self-Fitting Hearing Device

Jiong Hu1*, Jayaganesh Swaminathan2, Jade Kwan1, Mayra Rodriguez1, Alexis Dalager1, Anna Walters1

1University of the Pacific, 2Eargo Inc.

Current developments in advanced consumer electronics have allowed manufacturers to develop hearing-aid self-adjustment algorithms as an integral part of their technology. Such devices may change how the patients with hearing loss, clinicians, and the industry operate and interact with each other in the future. However, questions remain about the clinical efficacy and effectiveness of such devices/approaches compared to current standard of care. One such device is Eargo’s hearing aid that allows users to perform app-based in-situ hearing screening and self-select gain parameters based on the results of the hearing screening. The primary goals of this study were twofold: 1) To validate the accuracy of hearing thresholds measured with Eargo’s in-situ hearing screener against the hearing thresholds measured following Audiology best practice methods (i.e., in-booth audiometric testing with a clinical audiometer)
and 2) To establish comparability of Eargo hearing aids’ output with NAL-NL2 prescription targets through real ear measurements. As an extended goal, objective and subjective sound quality comparisons will be made between Eargo devices self-adjusted by the user compared with a traditional hearing aid fit-to-target by a clinician following Audiology best practice (ABP) methods.

127 subjects with normal-hearing and varying degrees of hearing loss were recruited. 100 subjects completed the hearing screener study. The results showed that the audiometric thresholds measured using Eargo’s in-situ hearing screener were comparable to the thresholds measured using ABP methods. Furthermore, the results showed excellent accuracy (93%), sensitivity (97%) and specificity (100%) for the thresholds measured with Eargo’s in-situ hearing screener compared to those measured with a clinical audiometer. Overall, the results indicated that, compared to ABP methods, Eargo’s in-situ hearing screener provides accurate and reliable inference about the hearing status of individuals with and without hearing loss.

For the second part of the study, 20 subjects completed the real ear measurements. The real ear aided response showed that the average deviation from NAL-NL2 targets was less than 5 dB SPL for audiograms representing mild-to-moderate hearing loss profiles. Results demonstrated that Eargo’s gain selection approach delivers hearing aid output comparable to a hearing aid fit to NAL-NL2 targets.

Taken together, the results validated the efficacy and effectiveness of a user directed self-adjusting hearing aid in providing adequate amplification for those with mild-to-moderate hearing loss. The results of this study may inform future design and development of self-adjusting hearing aid strategies grounded in principles of hearing science and clinical audiology.

CP301 Predicting real-life listening outcome based on in-situ acoustic and heart-rate measurements for listeners with normal and impaired hearing
Klaudia Andersson1, Jeppe Christensen2, Rasmus Skipper, Tobias Neher4
1Institute of Clinical Research, University of Southern Denmark, Odense, Denmark, 2Eriks Holm Research Centre, Oticon A/S, Snekersten, Denmark, 3Department of Audiology, Odense University Hospital, Odense, Denmark, 4University of Southern Denmark

Ecological momentary assessment (EMA) can provide insights into the real-life auditory ecology of hearing aid (HA) users. In a previous study, we combined EMA with acoustic data-logging and found that this can reveal situation-specific benefits of HA noise management (Andersson et al., 2021, Am J Audiol). Given that physiological measures such as heart rate (HR) are known to be sensitive to acoustic influences (e.g., sound pressure level and signal-to-noise ratio), we hypothesized that the inclusion of such data could provide even more detailed insights into daily-life listening outcome.

In the current study, participants with normal hearing (NH; N = 10) or mild-to-severe sensorineural hearing impairments (HI; N = 16) completed smartphone-based EMAs during a 2-week period. The HI participants were fitted bilaterally with behind-the-ear HAs with a single automatic program. The NH participants received a single HA each that they fastened to their collars. The HAs were used to collect continuous sound pressure level and signal-to-noise ratio data in the participants’ daily surroundings. Wristbands worn by the participants were used to collect complementary HR data.

Linear mixed-effects models with participant and time of day as random intercepts showed that the acoustic data could predict self-reported listening outcome for both participant groups (likelihood ratio tests against NULL models; both chi-squared, i.e., χ²(2) > 42.2, both p < 0.001). Including time-synchronized HR estimates in the models improved the predictions further. In addition, the models revealed negative associations between self-reported listening outcome and both sound pressure level and HR. Signal-to-noise ratio was positively associated with self-reported listening outcome for the HI group.

Overall, these findings imply that combining EMA with in-situ physiological and acoustic data can increase the predictive value of real-life listening evaluations.

CP302 More Robust Self-Assessment of Hearing Ability Using Retrospective Questionnaires
Joerg Buchholz1, Andrew Myles1, Yvonne Tran1, Kelly Miles1, Kiri Mealings1, Peter Derleth2
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Retrospective questionnaires are widely used to assess an individual’s hearing ability in the real world. However, it is widely acknowledged that their resulting data are prone to significant variance that is difficult (and sometimes impossible) to interpret. As such, the effectiveness of these tools to assess hearing abil-
ity is limited. The present study evaluates a new retrospective questionnaire called ECO-5, that was designed to provide a more robust measure of a participant’s ability to understand speech in noise. The ECO-5 assesses speech understanding in five relevant real-world scenarios that vary significantly in their expected difficulty for people to communicate. These scenarios are described in detail and are accompanied by informative photos to simplify their interpretation and reduce recall bias. A big-5 personality questionnaire as well as a socially desirable response set were additionally administered to control for subject-related biases. For reference purposes, we included the speech component of the Speech, Spatial, and Qualities of Hearing Scale (SSQ). A total of 134 adults varying broadly in their age and degree of hearing loss completed the questionnaires online. Sixty-two participants were identified as hearing aid wearers. Linear mixed-effects models were developed and controlled for age, sex, and “subject bias”. The SSQ revealed a significant effect of hearing loss on speech understanding, but no difference between the unaided and aided group. In contrast, the ECO-5 not only showed an increased effect of hearing loss, but also a significant effect of aiding which significantly covaried with test scenario. Moreover, using a congeneric confirmatory factor analysis applied to the five test scenarios (or items) of the ECO-5 we demonstrated strong construct validity. This suggests that the ECO-5 measures what it was designed for, i.e., to measure the ability of a participant to understand speech in noise. The ECO-5 therefore provides a valid retrospective hearing self-assessment instrument that, when compared to the SSQ, provides significantly increased sensitivity. Future studies will need to further evaluate its properties and compare individual outcomes to those derived from more direct (or objective) real-world measures, such as those obtained from ecological momentary assessments and realistic speech-in-noise tests.

**CP303 The Behavioral Dynamics of Speech Intelligibility Optimization during Conversations in Noise**

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People effortlessly coordinate simultaneous channels of verbal and non-verbal information while engaging in behavior to optimize speech intelligibility. When intelligibility is sufficiently jeopardized and people are not able to overcome the signal-to-noise challenge, successful communication is at risk. Here, we use speech and motion tracking sensors to uncover the behavioral dynamics that pairs of talkers use to optimize speech intelligibility during unstructured conversations. Twenty-two pairs of young, normal-hearing adults were tested under different conditions of simulated realistic background noise while standing freely inside a room or sitting around a table. The results demonstrate how different adaptive phases of behavior to optimize speech intelligibility unfold over time. With the presence or onset of background noise, pairs promptly negotiate a comfortable interpersonal distance and speech level which is dependent on background noise level and talking configuration. Following this enabling phase, pairs transition to a sustaining mode, where more subtle interpersonal coordination processes serve to facilitate communication by covarying with background noise and talking configuration. When conversations are sufficiently challenged by high levels of background noise, the number of communication breakdowns increase and pairs exhibit maintenance behaviors to restore communication. This transpires as decreased interpersonal distance and increased speech level adjustments in response to the signalling of a communication breakdown. We observe that a noise level of 78 dB SPL defines a critical threshold where optimization tactics to aid speech intelligibility begin to fail and communication breakdowns soar. Understanding human behavior when people interactively communicate in adverse listening conditions is crucial for innovating next-generation hearing devices, technologies and signal processing algorithms.

**CP304 Aiding in small group conversations – effects on behaviour of both the wearer and other group members**

Bryony Buck¹, Graham Naylor¹
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People with hearing loss exhibit coping strategies to help navigate challenging social situations. It is plausible that wearing hearing aids may not eliminate the additional perceptual and cognitive challenges people with hearing loss may face. Furthermore, behavioural effects are likely to become more diverse and dynamic when there are more than two interlocutors, and group context is the conversation scenario where people with hearing loss are most likely to experience activity limitations and participation restrictions. Interlocutors with normal hearing may also adapt their behaviour, be that consciously or unconsciously, in
response to the difficulty being experienced by interlocutors with hearing loss. Therefore, the impact of hearing loss on conversation behaviour and the benefit of hearing aids should be considered in terms of their effects on all interlocutors in a group conversation setting.

This research explores to what extent the wearing of hearing aids by people with hearing loss alters communication behaviour of all interlocutors (both with and without hearing loss) in complex conversation settings.

Groups of four participants (two with mild-to-moderate symmetrical hearing loss (HL), two with normal hearing (NH)) were recorded having conversations in quiet, moderate, and noisy environments. Environments were presented contrasting bilaterally aided (own devices) and unaided conditions for HL participants. 3D motion capture, 2D video, and audio recordings were combined with self-report measures of conversation fluency, effort and success. Metrics of conversation behaviour were examined with respect to contrasting hearing ability group (NH vs. HL), noise level, and aiding in the HL participants.

In this poster we will present preliminary results, with behavioural metrics grouped not according to their modality, but according to the coping strategy they may be presumed to reflect. This promotes meaningful interpretation of ‘selfish’ and ‘altruistic’ behaviours (examining ‘me’ versus ‘us’ motivation), and of behaviours which superficially provide no perceptual benefit. This research provides a novel exploration of the effects of hearing aid use in small group conversations comparing NH and HL communication behaviours.

**CP305 Prior knowledge alleviates listening effort during speech perception**

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Speech recognition tests almost always use stimuli where each sentence is completely unrelated to the sentence that came before it, which puts the listener into an unrealistic situation. The current study examines the impact of having some prior awareness of the topic of an upcoming sentence, especially as it relates to the effort of processing that sentence when missing words need to be mentally repaired.

Individuals with normal hearing and hearing aids (using their everyday programs) heard sentences with a masked target word which required them to retroactively fill in the missing word using later occurring context. All sentences had sufficient context for accurate disambiguation of the missing word. Sentences were preceded by visual display of text that either directly or indirectly cued the missing word, or was a neutral ‘XXXXX’. Pupillometry was used to track changes in effort over time as a function of prime type. We hypothesized that the effortful cost of retroactive repair observed in unrelated sentences would be reduced when individuals were provided prior knowledge of the topic of an upcoming sentence.

Preliminary results show the effort of listening to sentences was reduced when listeners were primed for the target word, indicating quicker resolution of the ambiguity. This study represents a transition from describing listening effort to measuring a factor that could help alleviate it, especially for individuals with hearing loss. In the clinic, patients with hearing loss are often given advice to aid in everyday communication like requesting an agenda prior to a meeting. The current study provides evidence to support that claim, along with detailed data on the timing of perceived ambiguity and repair. Prior knowledge can aid in an individual’s ability to follow along in conversations more accurately and quickly while exerting less effort. [This work is supported by NIH NIDCD F32DC109301.]

**CP306 Hearing Aid Fitting and Listening Performance in Spatially Complex Scenes**

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The gap between real-world speech reception performance of hearing aid users and results of lab assessments is still large. Furthermore, people with similar hearing thresholds may experience very different benefit and satisfaction from hearing aids. Evaluation and deeper understanding of listening performance in realistic situations remains a crucial step to individualize hearing solutions.

**Methods:** We used the Portable Hearing Laboratory (PHL) as a research hearing aid. It combines ear-level hearing aid hardware with the open Master Hearing Aid (openMHA) software.

Virtual listening situations with varying acoustic complexity in a realistic, reverberant cafeteria environment were created with a multi-loudspeaker setup and higher-order ambisonics using the Toolbox for
acoustic scene creation and rendering (TASCAR). Diffuse background noise and competing talkers were varied in number and level. The generated conditions ranged from a target speech source with two softer interfering sources (+6.3 dB SNR) to a target in loud background noise with two interfering sources (-7.5 dB SNR).

Based on a questionnaire completed in advance, 20 test subjects were classified into self-reported “high performer” and “low performer” groups. The subjects did a word scoring test and a subjective listening effort screening in each acoustic condition. We compared the performance between the unaided case, the subjects’ own hearing aids and the PHL. On the PHL hearing loss compensation was provided according to the trueLOUDNESS fitting method to restore binaural broadband loudness perception. In addition, three different signal enhancement algorithms were implemented: binaural coherence filtering, bilateral adaptive differential microphones (ADM), binaural minimum variance distortionless response (MVDR) beamforming.

**Results:** Overall, lower word scoring performance was found for the “low performer” group, which also seems to drop faster in more complex conditions than for the “high performer” group. This appears to be independent of the device (own hearing aid/PHL). “Low performers” also report higher listening effort in most of the conditions. Word scoring results indicate that spatial filtering provides a higher benefit for the “low performers”. For this group, the optimal result was achieved on average with the research hearing aid.

**Conclusions:** Speech reception results measured in simulated spatially complex listening scenarios reveal self-reported differences in performance with hearing aids. The benefit from spatial filtering is prevalently higher for self-reported “low performers”. However, large inter-individual differences were observed and suggest that further investigation into individual effects is needed to enable individualized selection of speech enhancement methods in the hearing aid fitting process.

**CP307 Evaluating the Intelligibility of Hearing-Aids in Realistic scenes: The Hearing-Aid Listening Test**

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¹UCL

Evaluating the intelligibility that one obtains with commercial hearing aids (HAs) is complex. Each HA employs a combination of signal processing strategies with implementations differing across manufactures and models. Measurements of the aids’ physical properties can be helpful to quantify performance but are difficult to relate to realistic listening scenes. Classic intelligibility metrics only consider audition hence falsely assume that intelligibility is fully defined by the listener’s audiogram. Consumers, on the other hand, demand the HA that is best fit to their individual needs. This raises the following question: given a hearing-impaired listener and a listening scene, which HA provides optimal intelligibility?

To answer this question, we equipped a manikin with six pairs of commercial HAs marketed as ‘binaurally connected’. They were fitted to a mild-to-moderate sloping standard audiogram, and we recorded the audio that the HAs generated in four auditory scenes: an anechoic room, a kitchen, a car cabin and a restaurant. Background noises natural to each scene masked target speech that was presented from two directions. We subsequently recruited hearing-aid users with hearing losses similar to the standard audiogram. Their keyword-in-sentence recognition was tested with the six HA recordings and in two additional conditions: unaided and with NAL-R1 amplification.

The HA providing the highest intelligibility varied across listeners and within listeners across scenes. The brand names of the listeners’ clinical devices mostly differed from their best performing HA. Simple NAL-R1 amplification more than occasionally outperformed the commercial HAs.

Inter-individual intelligibility differences in HA performance among hearing-impaired listeners are poorly understood. Rather than searching for the underlying origins, we took a utilitarian approach. It allows the identification of the HA that works best for a particular listener in a particular realistic scene. Tests like these could be designed to be self-administered and support consumers in the selection of their HAs. Binaural room impulse responses as obtained from HA microphones in each of the four scenes are publicly available and allow the evaluation of novel hearing-aid strategies. The current data provide a baseline for such strategies. Aggregated across listeners, the Hearing-Aid Listening Test permits the evaluation of binaural intelligibility metrics and potentially the application of machine-learning approaches to predict the intelligibility of speech processed by HAs.
CP308 Relation between Listening effort, demand, and Motivation Based on Ecological Momentary Assessments
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For adverse listening situations with background noise or reverberation, people with hearing impairment report difficulties and efforts to understand speech. McGarrigle et al. (2014) defined the term “listening effort” as “the mental exertion required to attend to, and understand, an auditory message.” A framework for Understanding Effortful Listening (FUEL) was proposed by Pichora-Fuller et al. (2016). According to FUEL, measures of listening effort are expected to vary according to two dimensions: demands and motivation. Furthermore, Pichora-Fuller et al. (2016) identified the question whether listening effort is related to psychological, social or health factors. This contribution explores the factors contributing to listening effort experienced in natural environments measured with self-reports in an Ecological Momentary Assessment (EMA) study. In total, 47 elderly participants with age-typical or mild-to-moderate hearing loss were equipped with a smartphone system for several days. They completed prompted and self-initiated surveys over the course of each day. These surveys included context items like situational settings and sound sources as well as perception items like loudness, listening effort, pleasantness, importance of hearing well, and speech understanding. Responses were given on 7-point categorical scales. On average, 17 surveys were collected per participant per day resulting in 2781 total surveys, including 2002 surveys of speech listening events. The context items revealed that the participants spent most of the time at home. The main sound sources observed were 1-on-1 conversations and radio program. Responses to the perception items indicated that listening was most effortful in activities outside home. Linear mixed model (LMM) analyses with listening effort as outcome were performed using the complete EMA data set and the speech listening events only. The model showed significant effects of demand (construed as loudness, hearing loss, location, and voice familiarity), and motivation (construed as importance to hear well) on listening effort. Data on personal characteristics collected in an interview (e.g., physical and mental health) did not contribute to the model.

References:

CP309 Less Effort and Less Fatigue: Long-Term Psychosocial Changes among New Cochlear Implant Users
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Recent research suggests that adults with hearing loss (AHL) are more likely to experience severe listening-related fatigue. The reason for increased fatigue in AHL is complex but the high levels of sustained effort required by AHL during listening tasks likely plays a role. To date, most research in this area has focused on AHL who are hearing aid (HA) candidates/users. Relatively little work has examined fatigue in cochlear implant (CI) candidates/users. Although some evidence suggests that HA can reduce fatigue in AHL, similar work in CI candidates/users is lacking. While CI candidates/users may experience increased effort/fatigue in everyday settings, the natural history of postoperative listening-related effort and fatigue remains unknown. This study addresses this critical gap in the literature.

Participants were adult CI candidates/users. Control group (N=101) participants included experienced unilateral/bilateral (n= 55/46) CI users. Experimental group (N=36) participants were first-time unilateral or bilateral (simultaneous-implantation) CI recipients (n=34/2).

Subjective measures of listening effort in everyday settings (Effort Assessment Scale) and fatigue (Vanderbilt Fatigue Scale for Adults) were collected via online surveys pre-activation and 0.5, 1, 2, 3, 6, and 12-months post-activation (Experimental group) or at baseline and 3 months later (Control group). Measures of hearing handicap (Hearing Handicap Inventory for the Elderly/Adults) and perceived isolation (subscale of the Social Disconnectedness Scale) were collected pre-activation/baseline and at 3, 6 and 12 months.

Linear mixed effects models were used to examine changes in scores over time. Time point was included as a within-subject factor, group (control, experimental) as a between-subject factor, and participant as a random slope. Results revealed a significant decrease in listening-related effort, fatigue, hearing

References:
handicap and isolation, over time for the experimental group. Significant reductions were observed as soon as 2 weeks post-implantation (effort and fatigue) or 3 months (handicap and isolation). Effort and fatigue ratings 12 months post-activation were significantly lower than those obtained 2 weeks post-activation. There were no significant changes in any of the control group ratings of effort, fatigue, handicap, or isolation.

Compared to control group baseline scores, experimental group participants reported significantly higher levels of effort, fatigue, hearing handicap, and isolation. No between-group differences were apparent 3-months post-implantation.

In conclusion, adult CI candidates are at increased risk for high levels of listening-related effort and fatigue, hearing handicap, and perceived isolation. Receipt of an initial CI can significantly reduce their effort, fatigue, handicap, and isolation. However, substantial individual variability in psychosocial benefit was observed.

**CP310 Adaptive Noise Reduction Preferences of Bilateral Hearing Aid Users in Different Daily-Life Acoustic Situations**

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This study examines the impact of daily-life situations on noise reduction preferences of hearing aid users. Therefore, subjective outcome data from 59 experienced hearing aid users with mild to severe hearing loss, who were equipped bilaterally with the same high-end hearing aids, was analysed. Hearing aids were fitted according to NAL-NL2 and target gain settings were verified using real-ear measurements. Hearing aids were provided with two listening programs in randomized order using the same gain settings but different noise reduction strength: very mild and very strong noise reduction.

After an accommodation period of two to four weeks, subjects started into a four- to six-week field period. During this field period, subjects were asked to compare both listening programs in daily life for better hearing support (Q1), better sound quality (Q2) and, when speech was present, easier speech understanding (Q3) and report on their experiences using field-report questionnaires. At the end of the field period subjects were asked for their overall program preference (either P1, P2 or no preference).

In total, 959 field-report questionnaires were completed by the 59 subjects, with conversations with one or several people (341) being the most frequently reported situation and music listening (44) being the least frequently reported situation. 9 subjects stated to have no overall noise reduction preference, while 26 subjects voted for very mild and 24 subjects for very strong noise reduction. Analyses of all field-report questionnaires across all situations showed significant differences between the three preference groups for Q1, Q2 and Q3 (p < 0.01) with a mean score according to the overall preference of each group. Situation-specific analyses showed that field-report ratings of speech situations for Q1, Q2 and Q3 were significantly different between the three preference groups (p < 0.05). In contrast, field reports on music and everyday sounds showed no significant difference between the three preference groups (p > 0.05). The results suggest that the overall help level preference is mainly driven by situations including speech while listening to music and everyday sounds seem to have only a small impact. The overall noise reduction preference ratings in this study support the hypothesis that experienced hearing aid users relatively clearly fall into two groups: those who prefer the increased contrast between background and speech from strong noise reduction, and those who prefer having access to all sounds in an acoustic scene. [This work was funded by William Demant Foundation.]

**CP311 The Effect of a Remote Microphone Technology on the Group Listening Experiences of Older Adults**

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**Objectives:** Conversing in noise may result in mental fatigue, decreased well-being, and communicative disengagement.

A remote microphone system can connect wirelessly to hearing aids (HAs) and increase SNR. It assists with listening in challenging situations where there is background noise and/or extended distance from the sound source.

This study evaluated the effects of using a remote microphone technology (RMT) on older listeners’ experiences during group conversations in noise.
**Design:** Four groups of 4 older adults (65-80 (average 71.5) years old; 11 male), with mild to severe hearing loss and difficulties hearing in noise, engaged in casual conversations carried out in 4 different conditions (RMT on/off x facial mask on/off). Participants sat in a circle and maintained COVID protocols distancing. They were fitted with hearing aids connected to RMT table microphones. They completed a conversation experience questionnaire after each condition and a final focus group discussion and questionnaire about listening condition preferences and expectations.

**Results:** Five separate ANOVAs were conducted with “RMT” and “Mask” as within-subject variables. The five dependent variables were questionnaire scores (total score, sub-scores for cognitive, psychological and inter-personal aspects of listening, and listening condition preferences). A significant main effect on total score (F(1,15)=5.94, p = .028) revealed better performance when the RMT technology was used compared to using HAs alone. Scores tended to be lower when masks were used than when they were not used, but the effect was not significant (with or without the RMT). There were significant main effects of RMT on the cognitive (p = .028), and psychological reaction (p = .036) subscales, but not on the inter-personal subscale. The final preference questionnaire confirmed that 93% of the participants expect a positive effect of the RMT in a wide variety of listening situations.

**Conclusions:** Using the RMT yielded significant benefits compared to using HAs alone during group conversation in background noise, either with and without facial masks. RMT can provide benefits in various challenging listening situations such as in noise and when communication redundancy is reduced (i.e., with the use of facial masks or when distancing). These benefits include positive effects on multiple (cognitive and psychological) aspects of listening; however, inter-personal reactions involving motivation to engage in conversation were unaffected by technology use. The results demonstrate the potential value of RMT in promoting hearing accessibility in group conversation-based activities for older adults such as may be offered in seniors’ community centres.

**CP312 Examining Changes in Subjective, Behavioral, and Physiological Indices of Listening Effort with Access to Noise Reduction Features**

Brittany Jaekel*1

1Starkey

Understanding speech in noise is a difficult task and may require significant listening effort, especially among listeners with hearing loss. Experiencing high levels of listening effort reduces the cognitive resources available for other mental processes; thus, determining effective methods for reducing listening effort could lead to a more efficient allocation of cognitive resources and improved speech understanding experiences, even in difficult listening scenes. We measured the extent to which noise reduction (NR) algorithms and directional microphones (DM) could reduce hearing aid users’ listening effort in noisy environments.

Participants were experienced hearing aid users, and the primary experimental task was speech-in-noise recognition, with or without NR and DM enabled. The secondary experimental task involved responding to visual stimuli using a computer keyboard. In addition to completing this dual-task paradigm, which served as a behavioral measure of listening effort (in which changes in reaction times to the visual stimuli across conditions indicate changes in cognitive resource availability), participants also reported their level of perceived mental workload for understanding speech (a subjective measure of listening effort). Finally, throughout the experiment, participants wore a fingertip heart rate monitor; data gathered by the monitor, which included calculations of heart rate variability, were interpreted as a physiological measure of listening effort and stress.

Through these three measurements – behavioral, subjective, and physiological – we achieved a more comprehensive understanding of how noise reduction features in hearing aids may impact listening effort in people with hearing loss. In addition, we evaluated the extent to which these different listening effort measures correlated with one another, and whether noise reduction features appeared to be particularly useful in reducing effort in certain profiles of listeners. This work will inform future investigations into further improving the noise reduction features available to hearing aid users, not only to increase speech understanding in noisy scenarios but to also reduce the effort exerted in the pursuit of understanding that speech.
**Objective:** The purpose of this study was to investigate the feasibility of using a portable, open-source hearing aid paired with ecological momentary assessment and soundscape recording in the real world. Open-source hearing aids and ecological momentary assessment will enable the transparent and collaborative development and real-world testing of hearing aid processing algorithms. This study describes platform development, compliance, feasibility, participant experiences, and potential challenges and use cases.

**Design:** This study used the Portable Hearing Aid Lab (PHL), a BeagleBone computer with wired receiver-in-the-canal earpieces, running the Open Master Hearing Aid software. Participants (with and without hearing loss) were asked to collect data in 5–10 complex listening environments over a 7–10-day period. In each environment, participants wore the PHL and completed two ecological momentary assessments on a smartphone over the course of 30 minutes. Between each momentary assessment for participants with hearing loss, the smartphone triggered the PHL to switch gain settings from an aided (fit to NAL-NL2 targets) to unaided (acoustically transparent) condition, or vice versa, testing the possibility of within-environment A/B comparisons of hearing aid algorithms in the real world. The PHL also recorded the environment.

**Results:** 12 participants with hearing loss and 10 with normal hearing completed the study. Most participants were able to operate the devices, but not universally. 266 EMA surveys were completed, but only 127 had associated recordings. Absent recordings were determined to typically be the result of user error. Only 1 participant (hearing loss group) declined to complete the study because the devices were too complicated. Participants with normal hearing completed EMAs in, on average, 7 complex listening environments and participants with hearing loss completed EMAs in, on average, 5 environments. Compliance did not differ significantly between groups. Data were collected at home (46%), in restaurants and bars (14%), transportation (10%), shops (8%), work (6%), and other environments, suggesting participants could use the device successfully in different places. Participants also wore the PHL for a variety of listening activities, including passive speech listening (26%), one-on-one conversation (22%), music listening (17%), and group conversation (9%). Compliance tended to increase over the course of the study as training was adapted.

**Conclusions:** Using an open-source hearing aid platform paired with EMA and sound recording for within-environment A/B hearing aid algorithm comparison is feasible. However, pitfalls were noted, particularly user error, user frustration, and device limitations. Training and streamlined interfaces and procedures are critical for real-world use.

**CP314 Spatial Release from Listening effort: Effects of Noise Direction and Age**

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Listening to speech in noisy environments is challenging and effortful. Our ability to understand and follow speech is not only affected by background noise, even when understanding perfectly well noise interferers increase the perceived listening effort (LE). In this contribution we analyze how spatial separation between target speech and noise interferer(s) influences subjective LE, and how this varies depending on age. The latter point is especially relevant in the context of age-related hearing loss.

**Method:** In this study we measured the effect of the spatial separation of target speech and interfering noise as well as the influence of the listener’s age on the perceived LE. Using the “Adaptive Categorical Listening Effort Scaling” method (ACALES, Krueger et al., 2017) the individual perceived LE function was rated in different spatial conditions with a speech source in the front and noise arriving from 0°, 90°, 135°, or 180° azimuth angle by 70 normal hearing listeners divided into three age groups (Group A: 18 and 30 years; group B: 41 and 59 years; group C: 60 and 76 years). Based on these data, the spatial release from LE was calculated and modeled using the binaural speech intelligibility model (BSIM2020, Hauth et al., 2020), which was originally developed for prediction of speech intelligibility.

**Results:** The effect of spatial release from LE was measurable using the subjective scaling method ACALES. Spatial release from LE increased with increasing spatial separation, up to a noise azimuth of 135°. This effect occurs for positive as well as negative SNRs, but is more pronounced in the case of negative SNRs. No significant effect of age was found for LE. Modeling the spatial release from LE using the binaural speech intelligibility model BSIM2020, was accurate in the negative SNR range. For positive SNRs, the predicted effect was larger than the measured spatial release from LE.
Conclusion: Spatial release from LE is not an age-related effect. BSIM2020 can model release from LE.

Outlook: Ongoing research investigates the spatial release from LE for hearing impaired listeners and the influence of the hearing aid provision.

References:

CP315 Fatigued at the end of the day? Hearing Aids help!
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Introduction: Increased listening effort, equivalent to increased cognitive load, builds up and can accumulate to higher feelings of fatigue. In this sense, listening effort can be interpreted as a “fatigue dose”. Several studies could prove that hearing aids help to reduce listening effort. A study comparing impaired listeners with and without hearing aids could show increased listening effort lead to more fatigue. This poster will describe a study with modern hearing aids with the results of the earlier study being an anchor.

Method: 20 experienced hearing aid users with a mild to moderate hearing loss were fitted with modern hearing aids. Participants underwent two “Time-Compressed Acoustic Days (TCAD)”; one day unaided, and another day aided. During each TCAD, they undertook various listening tasks over a 2.5 hour time period, simulating a typical day with common acoustical challenges. Performance on each task was measured behaviorally. Between each task, participants were asked to make a rating of the level of concentration required, as well as mental fatigue required. The study also utilized a non-auditory measure of concentration and attention ability. This was undertaken before and after the TCAD.

Results: Over the course of the TCAD, participant ratings of concentration and subsequent fatigue increased in both the aided and unaided conditions. However, concentration and fatigue ratings were significantly lower in the aided compared to the unaided condition. The impact of technology on both behavioral performance and self-reported ratings of listening effort were well demonstrated by one of the tasks, of which results revealed significantly better performance when using hearing aids in combination with an accessory versus the unaided condition. These findings suggest that hearing aids reduced the listening effort required for the task, which, in turn, is also reflected in lower concentration and subsequent fatigue ratings after the task has been conducted. Results of the non-auditory measures of concentration and attention ability showed that amplification leads to significantly faster higher processing speed at the end of the day compared to an unaided situation.

Conclusion: Results of this study suggest that at the end of the TCAD, hearing aid use reduced fatigue, allowing greater cognitive resources to be available for the concentration test. The results of the present study demonstrate that subjects have poorer performance both behaviorally and on self-report ratings when unaided. These findings suggest that when there is less ‘effortful’ listening, there is less opportunity cost and consequently less reported fatigue.

CP316 Head Orienting Behavior When Following a Multi-Talker Conversation
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Introduction: During speech communication and conversational turn-taking, listeners direct their head and eyes to receive meaningful auditory and visual cues and to relay nonverbal cues back to talkers. There is scant understanding of how listeners interact with multiple talkers, how this interaction may convey listener intent, and how hearing aids might take such movement or intention into account. This study investigated head orienting behaviors when participants were in an engaging audiovisual environment: a conversation among four talkers with live-action audiovisual (AV) information. Methods. Ten younger adults with normal hearing and twenty older adults with mild-to-moderate sensorineural hearing loss completed the study. Participants wore a headset snugly fit to the head upon which were small sensors. Movement of the sensors was tracked in real time by an infrared camera system (OptiTrack V120-Trio). Head movements were monitored during blocks of 90-second conversations, in which participants were tasked with following the content of the discussion (American football analysis). Blocks were designed to test head orienting behaviors for different AV source locations in broadband or babble background.
at +6 dB or +12 dB SNR. Hearing-impaired listeners were tested unaided and with hearing aids using omnidirectional microphones. We investigated the effect of these parameters on self-rated listening effort, which may mediate head movement activity levels, speech comprehension, or both. Results. Movement trajectories showed that hearing-impaired listeners turned much closer to the target location with an increase in latency compared to normal-hearing listeners. However, there was no significant hearing aid effect. Listening effort was significantly lower for normal-hearing listeners compared to hearing-impaired listeners. For hearing-impaired listeners, the listening effort decreased with aided listening. Conclusions. Head movement behaviors differ between hearing-impaired and normal hearing populations when following a conversation. It may be important for hearing aid spatial processing to consider the expectation that listeners are orienting towards target talkers when possible.

**CP317 Using Functional Near-Infrared Spectroscopy (fNIRS) to Evaluate the Effectiveness of Hearing Instruments**

Colette McKay*, Julia Wunderlich, Darren Mao, Onn Wah Lee, Gautam Balasubramanian, Mica Haneman

Newborn hearing screening has made a massive difference to early intervention for infants with hearing loss, leading to greatly improved language development for most of these infants. However, many infants currently suffer a delay between diagnosis and the establishment of the optimum hearing instrument and/or its accurate programming: a delay that would ideally be reduced to provide optimum language development. These latter infants include those with auditory neuropathy (how much can they hear, and what is their speech sound discrimination like?), those with severe/profound loss (do they need a cochlear implant?) and those with mild or unilateral loss (would they benefit from aiding at all?). In our lab, we are developing EarGenie, a complete assessment system using fNIRS that addresses the above challenges. EarGenie’s “Speech Module” evaluates speech sound detection and discrimination in sleeping infants to establish the adequacy of a hearing aid or need for a cochlear implant. We have developed the test and analysis techniques for this module to the level where we are confident that the results are more than 95% accurate in individual infants. EarGenie’s “Hearing Module” (currently under development) establishes thresholds and comfort levels in sleeping infants (especially needed for infants with auditory neuropathy or to objectively program cochlear implants).

**CP318 Real-World Evaluation of a New Active Vent Receiver for Hearing Aids**

Jorge Mejia*, Taegan Young, Matthew Crouteau, Catherine Morgan

The abnormal perception of one’s own voice as too loud or “boomy” in occluded ears remains a primary reason that most hearing aids fitted today use relatively large vents in earmoulds or open fittings. However, the disadvantage of open-fitted hearing aids is acoustic leakage, limiting the bandwidth of amplified sounds arriving at the user’s ears and impacting the potential gain that the hearing aids can provide without causing feedback.

The newest advancement in hearing aid occlusion management was recently introduced as an electronically controlled acoustic vent (ActiveVent) integrated into the receiver as part of a receiver-in-the canal fitting. The ActiveVent opens in situations where own-voice sounds may be problematic, such as while talking in quiet listening situations, and closes in situations where directional processing and noise reduction systems are most beneficial, such as when listening in a noisy environment, and also when having a larger bandwidth is important, such as when listening to audio streaming devices.

In an exploratory study, we examined the real-world benefits that a hearing aid with the ActiveVent feature may offer. We used an ecologically momentary assessment (EMA) method, deployed in smartphones, to carry out real-world data collections. Twenty-two recently fitted (with up to one year of hearing aid experience) hearing aids users were invited to attend two appointments, with a 3 to 4 weeks period between appointments while they performed field evaluation of the devices. At the first appointment, participants were fitted with a hearing aid with the ActiveVent, and at the second appointment, the EMA field data was collected. The EMA data captured the everyday benefit the hearing aid provided through daily check surveys. The EMA also collected real-time environmental data from the participant’s descriptions of their listening situation and interaction with others, as well as the acoustic features, e.g., sound level, extracted from smartphone microphone signals. Participants also responded to pre-and post-trial questionnaires examining their perceived hearing.
aid benefit, overall satisfaction, and listening effort. Preliminary results are presented herein, and the clinical significance of the outcomes is discussed.

**CP319 Dynamics of Pupil Response during the Sentence-Final Word Identification and Recall Test**

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There is a growing interest in understanding how listening- and memory effort are invested during speech recognition and recall. One way to investigate the effort investment is to compare the pupil responses during a speech recognition task that requires recall. Previous studies have demonstrated that baseline pupil dilation increases when speech recall is required in the task, suggesting higher memory effort due to speech encoding and memory processing. In addition, recent findings suggest better overall recall performance is associated with increased baseline pupil dilation (see presentation More is more: Physiological markers of successful effort by Micula et al).

Noise attenuation technology in hearing solutions is designed to improve speech intelligibility in noise. Studies have shown that even when speech intelligibility is high, noise attenuation enhances recall performance by freeing up cognitive resources for speech recall. Such benefit has been consistently found for people with hearing loss. Recently, there are similar investigations done among people with normal hearing and hearing loss using the combination of pupillometry and a speech recall task. However, these studies reported mixed results, in terms of recall performance, as well as changes in baseline and task-evoked pupil responses.

To better understand the effect of noise attenuating technology on the dynamics of pupil responses during a speech recall task, we took one step back and examined the effect of a preprocessed noise attenuation technology on recall performance and pupil responses. Participants with normal hearing were recruited. Pupil responses were obtained while the Sentence-final Word Identification and Recall test (SWIR) was administrated. Expectedly, speech recall performance was better in the condition when noise attenuation was applied. In the same condition, there was a stronger increase in baseline pupil dilation, suggesting higher memory effort, as well as a stronger decrease in task-evoked pupil dilation, suggesting less listening effort, over the course of a SWIR.

These results suggest that noise attenuation reduces resources for speech processing or listening effort for people with normal hearing. The freed-up resources are then devoted to memory processing and speech encoding. This can lead to better recall performance. We will also discuss these results and the possible underlying mechanisms based on different memory systems.

**CP320 Ecological Momentary Assessment App for the Open-source Speech processing Platform (OSP)**

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**Background:** EMA incorporates multiple experience sampling methods, including real-time outcomes assessments with human subjects. These assessments are typically done via surveys sent to participants multiple times a day across a trial period to gather relevant, in situ feedback about their experiences. EMA has been proven to improve many drawbacks of retrospective recall, diary studies, and other traditional methods. Researchers adopting EMA implementations within the clinical psychology domain and recent hearing-related topics suggest the critical need for developing and supporting EMA within opensource ecosystems. We have developed an EMA system enabling researchers to (i) design EMA surveys using a simple text-based authoring tool and share with others for extensions; (ii) incorporate contexts (e.g., background conditions, GPS, number of talkers, etc.) to dynamically control the surveys in real time, and (iii) log audio parameters (raw, spectra, subband energies, etc.) before, during, and after a given survey.

**Approach:** cEMA, is an interactive graphical user interface that runs on OSP’s embedded web server framework. We developed an offline authoring tool for Windows, Mac, and Linux operating systems based on YAML, an acronym for “Yet Another Markup Language,” and also a recursive version “YAML Ain’t a Markup Language.” Researchers can leverage many YAML online tutorials and cEMA examples to become proficient in designing EMA investigations for hearing aids research. cEMA has configurable settings that allow researchers to specify surveys’ push logic (e.g., number of surveys per day,
Hearing aid technology improves over time, as well as standard and best practice at the clinics. How the factors together affect the outcome for the end user in real life is the final feedback. This information is necessary in order to understand to what extent the improvements reach out to the end user.

The information is collected from the hearing rehabilitations. The Swedish Quality Register within Hearing Rehabilitation collects information both from private and county organized dispensers. The register includes subjective and objective information. Each year the register collects data from about 130 clinics and in total 85,000 prescriptions. The data is structured on national level, county level, and clinic level.

The data is used to understand differences in quality between hearing aids, clinics and counties for the same group of users. The information is also used to track individual clinic, regions, and national data over time. Longitudinal analysis is a very useful tool for tracking the quality and understanding the impact of different factors.

This work will present observation over ten years of data, showing variation in function score of the hearing aid. The newest data will also show effects due to the covid-19 pandemic. The dynamic in performance between the clinics will also be illustrated and discussed.

Through the years, the hearing aid has been considered as a medical device and there has been a separation between consumer electronics and medical devices both regarding the devices and user data. Modern hearing aids today are more integrated with other consumer products e.g. mobile phones. The users can choose to be more active with their devices, both controlling the functionality and making adjustments for different listening environments, and thus improving the functionality. These results will be presented and discussed.

**Results:** The national function score reported by users is relatively stable between the years 2015-2017. After 2017, a positive trend is observed where the users are reporting significantly higher function score. The active group using mobile phones together with hearing aids reports a significantly higher function score compared to the passive group, for all age groups 40-49, 50-59, 60-69, 70-79, 80-89, 90-99. The effect of the covid-19 is observed as a temporal decrease in average age of the users fitted with hearing aids.

**CP321 What Ten Years of Longitudinal Observations of Hearing Aid Fittings Can Tell us, and Integration Effects between Hearing Aids and Mobile Phones**

Peter Nordqvist*

1KTH Royal Institute of Technology - Hearing Bridge

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**CP322 Auditory Enhancement as a Measure of Adaptation to Real-World Acoustic Environments**

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Auditory enhancement refers to a spectral contrast aftereffect that facilitates the detection of novel events in an ongoing background and may also play an important role in achieving perceptual constancy in the face of changing acoustic environments, different talkers, and background noise. Under certain conditions, enhancement effects can reach 20 dB in listeners with normal hearing. Here we measured auditory enhancement under similar conditions in people with cochlear hearing loss, as well as in groups of age-matched and younger listeners with normal hearing. Our first results replicated the large (15-20 dB) effects previously observed in young normal-hearing listeners, but showed no significant enhancement in the group of hearing-impaired listeners. These results suggest that an effect that may play an important role
in everyday listening environments is severely reduced by cochlear hearing loss. However, follow-up studies in age-matched and younger normal-hearing listeners, with stimuli presented at the same high sound pressure level that had been used with the hearing-impaired listeners, also showed much less enhancement, suggesting strongly level-dependent effects in enhancement. Thus, hearing-impaired listeners may suffer a loss of enhancement, and hence poorer ability to adapt to real-world acoustic variability, due at least in part simply to the higher levels at which sound must be presented to be audible. [Work supported by NIH grant R01 DC012262.]

**CP323 Remote Psychoacoustic Tests Using Gaussian Processes for Personalised Hearing-Aid Fitting**

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The COVID-19 pandemic highlighted the need of having remote hearing testing and hearing-aid fitting to guarantee service delivery. They are also useful to provide high-quality service in remote or under-developed areas. However, remote tests are susceptible to level calibration and headphone responses. The standard procedures, often using up/down adaptive tracking, in which the stimuli are presented with a systematic combination of frequency/intensity, make many tests too long to be feasible for clinical practice. These problems are addressed in the present study by developing and evaluating audiogram, notched-noise and temporal masking curve tests that use Gaussian processes (GPs) and are implemented to be run on a web server. Since the notched-noise and temporal masking curve tests are based on the perception of the difference between signal and the masker, they are more robust than audiogram when the calibration is not known. The GPs presented stimuli only with informative combinations of frequency and intensity, which led to a reduction in testing time (50 trials in about 4 minutes) without losing accuracy (root-mean-square differences of 5 dB). Participants with previously known audiograms (having normal hearing, mild and moderate hearing loss) performed the remote tests. In particular, the audiogram was measured between 250 to 8000 Hz (Schlittenlacher et al., 2018), notched-noise tests were performed between 500 to 4000 Hz and with a set of 6 notch widths (i.e., 0|0, 0.1|0.1, 0.2|0.2, 0.3|0.3, 0.2|0.4, and 0.4|0.2) (Stone et al., 1992; Schlittenlacher et al., 2020), and temporal masking curves were measured between 500 to 4000 Hz and from -10 to 100 ms gap, with the linear reference (or off-frequency) threshold at 1600/4000 Hz (Lopez-Poveda and Johannesen, 2012; Johannesen et al., 2014). At the time of writing, three participants have finished the experiment. These pilot results showed that thresholds could be established for audiogram, and for the wider notches in the notched-noise tests. The condition 0/0 (i.e., no notch) was not successful, so we changed the strategy for the first few trials for the next participants. The remote audiograms were compared with those previously obtained in the clinics. The notched-noise tests were used to estimate the auditory-filter shapes. The temporal masking curve tests were used to estimate the inner- and outer-hair cell dysfunction. We have been able to assess the feasibility of remote tests, and to collect comprehensive information on participants’ hearing.

**CP324 Eye-Gaze Behaviour of Hearing-Impaired Listeners in Conversational Turn-Taking**

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Eye-gaze behaviour reflects cognitive processing and attention and can thus help us understand the experience of hearing-impaired listeners in realistic communication situations. Previous research indicates that when listeners follow a multitalker conversation, they tend to gaze at the active talker. Here, we examine this behaviour further by describing how the gaze behaviour varies across the stages of conversational turn-taking.

We analysed eye-tracking data from 22 hearing-impaired listeners who followed a realistic, recorded, audiovisual two-person conversation. The conversation was classified into time periods with voice activity by a single talker (solo), both talkers (overlap), or neither talker (silence). Periods of overlap and silence were further classified with respect to whether they occurred during a transition between the talkers or within a single talker’s turn. For each conversation classification, we calculated the proportion of gaze-time towards each talker. Although participants tended to select the same gaze-target at any given moment, individual listeners varied in their tendency to conform to the group’s behaviour. During solos, the group-selected target was usually the active talker, but not always. During silence and overlap, the group tended to look at the most recent turn-holding talker, but behaviour within the group was less reliable than during solos. These results will be discussed in light
of their implications for future hearing assistive technology.

**CP325 Development and initial validation of the Hearing Aid Feature Importance Evaluation (HAFIE) questionnaire**

Hasan Saleh*, Paula Folkeard†, Selina Liao*, Susan Scollie§

*University of Western Ontario

In this study, a novel questionnaire called the Hearing Aid Feature Importance Evaluation (HAFIE) was developed and initially validated. The aim of developing this questionnaire was to provide a structured, evidence-based methodology for hearing aid recommendations and selection. This may be achieved by allowing clinicians to gather patient attitude and self-reported importance ratings for different modern hearing aid features.

Initial questionnaire items were designed using the statements generated in a concept mapping investigation of the drivers of user preference between hearing aids at higher and lower technology levels (Saleh et al., 2021). A series of focus group interviews were conducted with experienced hearing care professionals (n=10) to assess these items and gather suggestions regarding further questionnaire design and content. The items were modified based on the focus group results and a scan of currently available hearing aid features.

Validation of this initial 34-item version of the questionnaire was conducted using an anonymous online survey tool (Qualtrics). Respondents were adults self-reporting hearing difficulties (N = 218, median age = 48 years). Exploratory factor analysis was used to assess the factor structure of the dataset, using principal axis factoring and an Oblimin rotation. Three factors were identified, dividing the hearing aid features into the subscales: “Advanced connectivity and streaming”, “Physical features and usability”, and “Sound quality and intelligibility”. Seven items did not load well onto any factor or had high factor loadings on more than one factor and were therefore removed. This resulted in a 27-item questionnaire with three subscales. Reliability of each of these subscales was assessed via Cronbach’s alpha and item-total correlation, and each was found to be appropriate.

**Reference:**


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**CP326 The Art of Asking – Considerations when Using Ecological Momentary Assessment in Hearing Research**


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Ecological Momentary Assessment (EMA) is increasingly popular as a methodology in hearing research. With the use of EMA, valuable insights have been gained into people’s auditory ecology, perceived hearing difficulties, and hearing device use and benefit. However, it is sometimes difficult to compare results from different studies because the methodology used varies considerably. It is sometimes difficult to compare results from different studies, since the methodology used varies considerably.

Papers from other fields give general EMA recommendations, but we identified no paper that focuses on EMA methodology in hearing research. Some challenges when using EMA are special to our field. For example, hearing problems are often most severe in social situations, where immediate reporting may be perceived as especially disruptive or impolite. Hearing difficulties are highly dependent on acoustic context and matching of objective data (such as measured sound pressure levels) and subjective data (such as ratings of difficulty) is an important issue.

Over the last two years, an international group of researchers experienced with EMA in hearing research has gathered online for a monthly workshop series, where we have exchanged information about our past EMA studies and learned from each other’s experiences. We have compared our study designs to better understand the implications of different design choices on outcomes, compliance, and data quality.

One outcome of the workshop is a suggested list of questionnaire items and response alternatives that can be used in EMA studies across different research questions. In this poster, we present these questionnaire items with the goal to inspire other researchers who are designing EMA studies. We are hoping for feedback and discussions on the items and on the usefulness of such a collection of questionnaire items for
the hearing research community as a way of achieving consistency in methods and the ability to compare findings across studies.

**CP327 Using Real Group Conversations in Laboratory Testing**

Karolina Smeds¹, Petra Herrlin¹, Eline Borch-Petersen², Frédéric Marmel¹, Florian Wolters¹, Anja Kofoed Pedersen¹
¹ORCA Europe, WSA, ²WSA

There is currently scientific focus on improving the ecological validity of laboratory hearing assessments. Most work has highlighted the selection of listening situations and playback using multi-speaker setups. Our work has instead centered on investigating the listening tasks that can be used, specifically real conversations. Various outcomes can be obtained during a conversation. The actual conversation may be analyzed acoustically (e.g., turn-taking timing), linguistically (e.g., repetition behavior) or behaviorally (e.g., head and eye movements). Another option is to stage a conversation and use traditional psychoacoustical tests while conversing (e.g., paired comparisons of two hearing-aid settings). There are also various ways to elicit a conversation. Collaborative puzzle-solving, map-navigation, and spot-the-difference (Diapix) tasks have been used in conversations with two interlocutors. Here, we report on aspects of using real conversations in small groups of interlocutors. In one study, groups of three interlocutors subjectively evaluated four different ways to elicit group conversations, while acoustical analyses were used to explore how the elicitation method affected the conversations. In another study, groups of four interlocutors evaluated four staged scenarios in terms of perceived realism and conversation success. The experimental considerations for using group conversations in the laboratory will be discussed.

**CP328 Assistive Listening and Sound Quality during a Live Orchestra Performance**

Larissa Taylor*, Steve Armstrong², Dan Bosnyak¹, Ian Bruce¹
¹McMaster University, ²SoundsGood Labs

In addition to difficulties listening to speech, music sound quality can be greatly affected by hearing loss and hearing aid processing. Any non linear processing on hearing aids has the potential to distort important elements of music. Live music has additional challenges compared to recorded music, in both volume and the bandwidth of frequencies that are necessary for the best possible sound quality. In order for hearing aid processing to have optimal processing, sound quality for live music needs to be maximized, so ecologically valid studies during live performances are essential to fully characterize sound quality.

Preliminary studies in the LIVELab and an experiment conducted during an orchestra concert showed that while music sound quality judgments in hearing aid users are subjective and variable between subjects, those with high musical sophistication are more critical and consistent in their judgments. Two different assistive listening systems, a telecoil loop and a 2.4 GHz RF streaming system, were tested during the concert to determine if such a system was beneficial for live music. Several different audio mixes from microphones amongst the musicians were rated during the concert for sound quality and loudness to find an optimal mix for the assistive listening systems. No clear optimal mix for improving sound quality was found, but the high musical sophistication group did show a preference for having the assistive listening system compared to the hearing aid microphones alone. In order to have more detailed and paired comparisons, a follow up study using a MUSHRA listening test with recordings from the initial concert will be performed, and initial analysis of results presented.

**CP329 Listening Effort Measured in Multiple Subjects Simultaneously during a Live Improv Performance in Background Noise and Reverberation**

Larissa Taylor*, Henry Luo², Ian Bruce¹
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Listening effort, or the amount of cognitive effort required to listen to a sound of interest, is an important measure of hearing performance, especially for hearing aid users. Hearing loss leads to increased listening effort in noisy situations and ideally hearing aid processing would reduce this effort in order to also reduce the fatigue that hearing impaired individuals experience in these situations.

The goal of the listening effort study presented here was to collect multiple measures of listening effort in an ecologically valid scenario, testing the effects of background noise, reverberation, and hearing aid directional processing on listening effort and head movement. The experiment consisted of an improvised performance on stage which the subjects listened to while background noise and reverberation in
the room changed to give a range of listening difficulty during the performance. Subjective ratings were collected on tablets during the performance, and ECG and motion capture data were also collected. To avoid the variability introduced due to age and varying degrees of hearing loss, for this initial study young normal hearing listeners were used. Two types of directional hearing aid processing were compared to the unaided condition.

Preliminary results presented at IHCON 2018 showed effects of directional hearing aid processing on listening effort as measured by heart rate variability as well as subjective ratings.

New results from further analysis of the data, including resting heart rate variability as a covariate as well as motion capture data results, have since been obtained. These results show an effect of background noise level and reverberation on subjective listening effort, an effect on physiological listening effort, as well as a right ear bias for head direction in increased background noise and reverberation. Hearing aid type showed a significant effect on deviation angle from the speaker on stage, that is the difference between where the subject was looking and the location of the actor speaking on stage. This deviation angle effect demonstrates a natural trend towards a right ear bias as listening became more effortful due to reverberation and background noise.

**CP330 Functional Near-Infrared Spectroscopy (fNIRS) as a Measure of Listening Effort and Speech Intelligibility in Adults with Hearing Loss**

Jonathan Vaisberg*, 1, Bilal Sheikh1, Frank Russo2, Jinyu Qian1
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There is broad consensus in the hearing community that listening effort is an important outcome for measuring hearing performance. Many individuals reporting hearing difficulties describe listening effortfully, despite presenting with normal auditory thresholds. Despite well-established methods like threshold testing and speech repetition, there is little consensus on the best ways to measure listening effort. This study seeks to measure listening effort using a relatively new neuroimaging technique called functional near-infrared spectroscopy (fNIRS).

fNIRS emits near-infrared light through the scalp. Some of this light will be scattered through cortical tissue before travelling back to detectors. Brain activation in a local cortical area can be assessed through relative changes in light intensity due to absorptive properties of oxygenated and deoxygenated blood. fNIRS is attractive for hearing research due to its relative silence, mobility, and resistance to electronic equipment like hearing aids. fNIRS has long been used to assess cognitive effort and has recently been extended to hearing paradigms. For example, increasing blood oxygenation in the prefrontal cortex (PFC) has been associated with increasing verbal n-back working memory demands in older adults using hearing aids (Rovetti et al., 2019), as well as changes in signal-to-noise ratio (SNR) and semantic context in a speech recognition task with normal hearing listeners (Rovetti et al., 2021).

The purpose of the current study is to measure oxygenation in the PFC using fNIRS in a sample of older adults with hearing loss using a simplified experiment based on Rovetti et al. (2021). Participants will be asked to repeat the final word from low-context sentences in noise at two SNRs; a hard SNR (individually-measured SNR-50), and an easy SNR (SNR-50 + 10 dB) both without and with hearing aids set to a directional mode. Speech and noise will be presented from the participant’s front and rear, respectively, to fully utilize the directional benefit. We hypothesize that more PFC oxygenation will be associated with more challenging conditions (hard SNR/aided) compared to less challenging conditions (easy SNR/unaider), particularly as expressed in the left PFC.

Pilot data from normal hearing listeners has revealed the expected trends with greater oxygenation in the left PFC with more challenging SNR, and less oxygenation in the left PFC with amplification in the less challenging SNR. These changes in oxygenation may reflect changes in listening effort, and support the use of fNIRS to investigate listening effort. Next steps include testing and analyzing with adults with hearing loss.

**CP331 Real-Life and Real-Time Hearing Aid Experiences – Insights from Smartphone-Collected Data**

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Modern hearing aids analyze the acoustical environment they are in, as sound is captured by their microphones. This can inform hearing aid settings tailored to the environment, and potentially, switching automatically between settings when the acoustical environment changes. Real life communication extends
beyond the characteristics of the environment, though, and a person’s listening intention could play a crucial role in satisfaction with hearing technology as well. Also, after a first fit, a listener experiences hearing aids in their own natural environment for the first time. During the next appointment, the listener would explain potential points of improvement regarding the fitting or physical fit to the clinician, who, in turn, translates this feedback into fine tuning or other actions. This process is prone to memory bias and communication barriers: the user needs to remember how they experienced the hearing aids, in what situation, and they need to describe these experiences in their own natural language at a different point in time. Recently, however, smartphone connected hearing aids have allowed listeners to provide feedback to their clinician in (near-) real time, in their own natural environments, using their own natural language.

The aim of this project is to explore how listeners experience hearing aids in real-life and real-time. We explore how satisfaction differs for different combinations of self-reported listening activities and hearing aid classifier states. We also explore how hearing aid wearers describe these experiences using their own natural language, identifying recurring themes in descriptions. Results of a retrospective analysis of 30 127 smartphone collected feedback sessions show that (1) listeners more often provide feedback in low rather than high complex acoustic environments; (2) the proportion of positive ratings increases when the complexity of the listening intention and the acoustic environment decreases; (3) positive feedback is correlated with how hearing aids foster the listening experience, while negative feedback is correlated with technical, device-related themes. Clinical implications and future directions from the insights of this large database of real-life, real-time self-reported hearing aid experiences will be explored.

CP332 Concurrent Matrix Test: Determine Speech-Recognition in Multi-Talker Situations
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Unlike real-life conversations, testing speech-recognition in diagnostics or rehabilitation of a hearing impairment is mostly performed with one target speaker from one direction. In complex real-life communication, aspects like e.g., competing talkers, turn-taking, selective and divided attention, play an important role and should be considered in audiological testing. The Concurrent Matrix Test (CCOLSA) was developed to simulate the use case of a group conversation with three talkers (Heeren et al, 2022). The spatially separated speakers alternately present sentences of a Matrix sentence test (Kollmeier et al, 2015). One specific name of the Matrix test material, the name ‘Kerstin’ in the German version, is used as the call sign, reflecting that the listener in the group conversation is called ‘Kerstin’ and therefore is addressed by that call sign. The task of the listener is to repeat the last words of all sentences from that talker until another talker begins a sentence with ‘Kerstin’. Since the alternation of the talkers was implemented with an adjustable overlap time between sentences across talkers, the call sign and the target words overlap. Hence, detecting the call sign and repeating the target words must be performed as a dual-task paradigm. Experiments with normal-hearing listeners showed that this approach allows for analyzing turn-taking and attentional factors by comparing the call-sign detection task and the speech-recognition task either as single or as dual tasks under the same listening conditions. Speech-recognition scores significantly differed between single and dual-task paradigms (word-recognition scores of 93-95% versus 77-88%) and across different overlap times: Increasing the overlap time yielded decreased word-recognition scores in the dual-task paradigm. The difficulty of the test can be adjusted by the temporal overlap between call sign and target word, allowing for usage at high signal-to-noise ratios.

The poster presents the approach of the CCOLSA and shows single and dual-task results of the test paradigm with young normal-hearing listeners (Heeren et al, 2022). In addition, the application in evaluation of hearing devices as well as the application of an adaptive measurement procedure are discussed. [ Funded by the Deutsche Forschungsgemeinschaft (DFG, German Research Foundation) Project-ID 352015383, SFB 1330 C4. ]

References:

CP333 Hearing Aid Benefit Associated with Individual Auditory Perceptual Profiles through Ecological Momentary Assessment
Background: Individuals with similar audiograms may receive fairly similar hearing aid (HA) gain settings, validated and fine-tuned with real-ear measures for better precision. Still with a “perfect” fit, some perceive benefit in speech understanding, while others do not, suggesting untargeted inter-individual differences. Hearing loss may result in one or more deficits in auditory perceptual abilities that are heterogenous among individuals and, along with cognitive abilities, may be important factors affecting HA benefit. The goal of this study was to identify “auditory perceptual profiles” based on individual differences in auditory perceptual abilities and to determine any associations with subsequent HA benefit. A goal, with precision audiology, is to leverage perceptual profiles to target compromised perceptual abilities in hearing aid fittings.

Methods: A condensed test battery using the Portable Automated Rapid Testing (PART) platform assessed the auditory perceptual and cognitive abilities of twenty older adults with mild to moderately-severe hearing loss. Assessments included measures of frequency selectivity, spectro-temporal processing, temporal fine structure (TFS) and binaural processing, temporal envelope perception, spatial release from masking (SRM), and measures of working memory and fluid intelligence. A step-up design was used in which listeners were evaluated unaided for two weeks followed by three weeks aided. Ecological momentary assessment (EMA) surveys reflecting real day-to-day listening experiences were collected four times per day for the first (unaided) and last (aided) two weeks. Individuals were fit with HAs using clinical best practices. An objective hearing in noise test and standard HA questionnaires were administered prior to and after HA use.

Results: Cluster analyses identified three auditory perceptual profiles from individuals’ auditory perceptual and cognitive abilities. One profile (A) indexed poor performance on all perceptual and cognitive tasks and was associated with minimal HA benefit. Profiles B and C were associated with good perceptual performance and greater HA benefit. Profile B differed from C in which measures of HA benefit were significant. Working memory and SRM were major factors separating the “best” profile (B) from the other two (A and C); whereas the other two profiles were separated by performance on frequency selectivity, spectro-temporal sensitivity, and TFS processing.

Conclusion: Auditory perceptual profiles were associated with different degrees of HA benefit, especially as measured under ecologically valid conditions using EMA. Auditory perceptual abilities may be important precursors in identifying realistic expectations and, in the long-term, HAs may precisely target auditory percepts to improve benefits and successful HA adoption.

CP334 Modulation of Sound Localization Parameters by Pulling the Tensor Tympani and Stapedius Muscles
Sunil Puria*, Michael E. Ravicz1, Nam Hyun Cho1
1Mass Eye and Ear

Introduction: The three-bone flexible ossicular chain may allow alterations of middle-ear (ME) sound transmission via independent or concerted action of its two attached muscles for sound localization. It is well-known that ME sound transmission has a wide bandwidth and a delay (e.g., Puria and Allen, 1998). We hypothesize that ME muscles can modulate sound transmission latency in addition to their well-documented attenuation of gain. We studied this question by mechanically pulling the tensor tympani (TT) and stapedius (ST) muscles independently in human cadaver ears.

Methods: The middle-ear cavity of human-cadaver temporal bones was opened to access the TT and ST bodies. Forceps gripped the TT and the ST bodies to apply static pull with a load cell and micromanipulator. Tones (0.1-10 kHz) were delivered to an ear-canal coupler with a microphone to measure ear-canal pressure (Pec). A Thorlabs Ganymede-III-HR 905-nm optical coherence tomography system measured the eardrum shape and umbo velocity Vu. Stapes velocity Vs was measured with a laser-Doppler vibrometer. From these measurements, we determined Vu/Pec, Vst/Pec, and the eardrum shape from muscle pulls. For frequencies between 200-700 Hz, we computed the group delay for each muscle pull that ranged from 10-100 gram force, a physiologically likely range.

Results: A static TT muscle pull produced attenuation in sound-driven umbo and stapes motion magnitude of 12-15 dB at frequencies below about 1 kHz for the highest pull forces. The baseline ME delay was reduced (phase increased) by 150-250 us. A static ST muscle pull attenuates the stapes motion by 6-7 dB below about 0.6 kHz and the baseline ME delay was reduced by 30-100 us. The ST muscle pull affected...
mostly the stapes motion, while TT muscle pull affected the umbo and stapes motions roughly equally. TT muscle pull also caused a static medial umbo displacement and TM shape change, which is likely the mechanism for the observed transmission changes.

**Discussion:** These results demonstrate that sound transmission delay through the human ME is altered by TT and ST muscle pulls that mimic muscle activation. The results suggest that ME transmission delay can be reduced by up to 250 us; this is a large fraction of the interaural time difference produced by the head (~ 800 us). Thus, a role for TT and ST muscle activity might be to modify low-frequency sound-localization cues and future hearing aids might incorporate this into their signal processing. [Work supported by Amelia Peabody Charitable Fund.]

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11:00 a.m. - 12:00 p.m.

**SESSION C2. Part II: New Approaches to Hearing-Aid Fitting and Service Delivery**

Session Chair: Peggy Nelson

**C2-1. Release from Hyperacusis: The Promising Results from an Initial Field Trial.**

*David Eddins*,†, *Dana Cherri*,†, *Carrie Secor*,†, *Steve Armstrong*,‡, *Craig Formby*,†

†University of South Florida, ‡Soundsgood Labs

Hyperacusis, an unusual intolerance to everyday sounds, can be a debilitating condition. Affected individuals often present in the clinic wearing earplugs to limit offending sound exposures. Unfortunately, their mis- and over-use, combined with unhealthy sound avoidance behaviors, can exacerbate the hyperacusis condition. These counterproductive management strategies pose a major barrier for transitioning patients into best-practices sound therapy treatments often implemented with sound generators. At IHCON 2018, we described a novel treatment protocol, including structured counseling, that addresses this challenge by combining features of a well-fitting earplug to protect against potentially offending sound levels, a sound generator to provide sound therapy, unity gain to provide comfortable exposure to low level sounds, and loudness suppression to limit exposure to high-level, offending sounds. To encourage the use of sound therapy as much as practical, we provided users with a thin-tube, open-dome configuration in environments at low risk for offending sound levels and the option for the thin tube coupled to the custom earplug in high-risk environments. The protocol featured systematic release of loudness suppression based on treatment-induced changes in loudness judgments for “loud, but ok” running speech.

Here we report the first laboratory-based field trial of this protocol. Twelve subjects completed the trial. Average, pretreatment hearing threshold levels were within the normal range and did not change posttreatment. Pretreatment judgments of “loud, but ok” levels ranged from 30 to 80 dB HL for warble tone stimuli and from 26 to 65 dB HL for spondees. The six-month trial included monthly visits for testing and treatment-induced device adjustment. Exceptions due to COVID-19 will be discussed. Initial fitting and follow-up of the transitional intervention involved a closed-loop adjustment and verification protocol to ensure precision measurement of unaided gain, occluded gain, the sound generator response, and the aided unity gain needed to overcome earplugging, loudness suppression function and setting, and sound tolerance for running speech measured in the ear canal. Posttreatment, patients demonstrated benefit on multiple indices. Most dramatic was their average increase of 34 dB in loudness tolerance for running speech. This equates to an average incremental treatment effect (release from loudness suppression) of 5.6 dB per month. Subjective indices of hyperacusis changed accordingly, with significant improvements on the Hyperacusis Questionnaire, the Tampa Scale of Hyperacusis,
the Noise Avoidance Questionnaire, and Questionnaire on Hypersensitivity to Sound. Subjective feedback indicated a return to many normal daily activities posttreatment and high satisfaction with the intervention (NIDCD R21DC015054).

C2-2. Amplitude Compression for Listeners with Rollover at Above-Conversational Speech Levels

Michal Fereczkowski*,1, Stine Christiansen1, Raul Sanchez-Lopez2, Tobias Neher1
1University of Southern Denmark, 2Interacoustics Research Unit

Hearing aids provide level-dependent gain to improve speech audibility and loudness perception. While more audibility typically leads to better speech intelligibility at lower levels, the same is not true for above-conversational levels where decreases in intelligibility can be observed. Traditionally, such ‘rollover’ has been interpreted as a sign of retro-cochlear deficits (e.g., due to auditory nerve tumors). More recently, however, rollover has also been observed in listeners with normal audiograms and in hearing-aid users with different degrees of hearing loss. This has led to the idea that cochlear synaptopathy could be involved in speech intelligibility decrements at above-conversational levels.

Here, the relation between rollover and aided outcome was investigated in 37 older experienced hearing-aid users. Performance-intensity functions were measured based on word recognition scores obtained at several above-conversational levels with individual linear amplification. Aided outcome was assessed using the Hearing-In-Noise Test (HINT) and the International Outcome Inventory for Hearing Aids (IOI-HA) questionnaire. The analyses revealed rollover in 35 out of 74 test ears. Furthermore, rollover was associated with poorer aided HINT scores obtained at a moderate level. Also, a trend for rollover to be associated with lower IOI-HA interaction scores was observed.

A subset of the participants with rollover were then tested with three amplitude compression settings. Two of these were designed to avoid rollover by applying slow- or fast-acting compression with high compression ratios around the ‘sweet-spot’ in the participants’ performance-intensity functions. The third setting made use of NAL-NL1 gains and thus served as a reference. Speech intelligibility was assessed at multiple levels in quiet and in noise. Pairwise preference judgments were also collected. As expected, sweet-spot compression outperformed the reference setting at high levels and gave comparable results at moderate levels.

Overall, these findings highlight the value of supra-threshold hearing diagnostics for personalized hearing-aid fitting. [This research was carried out as part of the ‘Better hEAring Rehabilitation’ (BEAR) project. Support from the Innovation Fund Denmark (Grand Solutions 5164-00011B), Oticon, GN Hearing, WS Audiology and other partners (Odense University Hospital, Aalborg University Hospital, Aalborg University, and Force Technology) is sincerely acknowledged.]

C2-3. Spectro-Temporal Modulation Sensitivity as a Predictor of Realistic Aided Speech Reception and Effects of Directionality and Noise Reduction

Johannes Zaar*,1, Lisbeth Birkeland Simonsen2, Søren Laugesen3
1Eriksholm Research Centre and Technical University of Denmark (DTU), 2Interacoustics Research Unit (IRU) and Technical University of Denmark (DTU), 3Interacoustics Research Unit

A range of studies have demonstrated a connection between spectro-temporal modulation (STM) sensitivity and speech reception in noise. However, some studies reported problems establishing an STM test paradigm that is applicable across a wide range of age and hearing loss. Furthermore, the considered speech tests typically lacked realism and it remains unclear how STM sensitivity could be exploited for prescribing hearing-aid (HA) processing, espe-
cially regarding advanced features such as combined directionality and noise reduction algorithms (in the following termed “NR” for simplicity). The present study investigated the connection between STM sensitivity, aided speech reception, NR benefit, and subjective NR preference in 30 elderly Danish listeners with varying degrees of sensorineural hearing loss. A suitable STM test design was chosen based on a previous study, using a band-limited carrier noise in combination with a spectral density of 2 cycles/octave and a 4-Hz temporal modulation rate, presented to subjects with ear-specific audibility compensation in a three-alternative forced-choice (3-AFC) paradigm. Speech reception thresholds (SRTs) were obtained in a spatialized speech-on-speech set up with hearing aids in three settings: NRoff (amplification only), NRmild (amplification and mild NR), and NRstrong (amplification and strong NR). Subjective NR preference was assessed by means of a preference rating between NRmild and NRstrong following a field trial. The results indicate that STM sensitivity was correlated with (i) SRTs measured in NRoff, (ii) the SRT benefit induced by NRstrong as compared to NRmild, and (iii) the subjective NR preference, with poor STM performance being connected to preference for NRstrong. The latter relation only appeared for a subset of 22 subjects (that sufficiently exposed themselves to noisy situations in the field trial) and seemed to be affected by cognitive aspects. A more clinically viable STM sensitivity measurement paradigm, heavily inspired by the clinical pure-tone audiometry measurement procedure, was designed using continuous carrier noise presentation with STM imposed at the audiologist’s request. In a follow-up study with the same subjects, this procedure showed excellent time efficiency with good test-retest reliability and a strong correlation with the previous 3-AFC procedure results. Overall, the present study suggests that the proposed approach to measuring STM sensitivity is suitable for clinical purposes and provides crucial information to the audiologist regarding supra-threshold hearing deficits. This can be useful for counselling, HA selection and adjustment of advanced HA features.

12:00 p.m. - 12:30 p.m.  Announcement of 2024 Technical Committee candidates / Introduction to CX Satellite Workshop

12:30 p.m. - 1:30 p.m.  Lunch

1:30 p.m. – 5:00 p.m.  Leisure

2:00 p.m. - 3:00 p.m.

SESSION CX. Short Talks: Open-Source Audio-Processing Tools Satellite Workshop

A satellite workshop on open-source audio-processing tools, with oral presentations, posters, and technology demos

**CX-1. The Tympan Open Source Audio Processing Platform**  
*William Audette*,†  Eric Yuan†, Ken McEnaney‡, Stephen Neely‡, Ryan McCleery‡, Marc Brennan‡, Joel Murphy†, Odile Clavier†  
†Creare LLC, ‡Boys Town National Research Hospital, ‡University of Nebraska, † tympan.org

Hearing-aid research is limited by the lack of wearable, reconfigurable and programmable audio processing platforms that can evaluate the real-time performance of new algorithms and innovative paradigms designed to help hearing impaired listeners in challenging conditions.
The Tympan is an open-source hardware platform that utilizes a powerful Teensy 4.1 600MHz processor that can perform 16-band stereo compression in real-time yet can be programmed in the user-friendly Arduino environment. A mobile app (Android/IOS) gives researchers the ability to configure algorithms on the fly, store listener feedback, and record audio using a simple interface. Researchers can also customize the mobile app by defining the user interface right in the Tympan firmware, without any knowledge of mobile app software development. An open-source behind-the-ear (BTE) earpiece, with dual microphones, allows users to conduct human studies in realistic environments, where listeners can modify the parameters of their hearing aid through the mobile app. During this special session, we will describe the Tympan platform design, report on computational performance measured in the laboratory and summarize the steps the team has taken to disseminate the platform and reach users in varied communities.

CX-2. Hearing-aid research with the open Master Hearing Aid (openMHA)
Hendrik Kayser*, Chaslav Pavlovic2, Volker Hohmann1
1Carl von Ossietzky University of Oldenburg, 2BatAndCat Sound Labs

The project "Open community platform for hearing aid algorithm research" (NIDCD R01DC015429) provides a software platform for real-time, low-latency audio signal processing: open Master Hearing Aid (openMHA). It contains a versatile set of basic and advanced methods for hearing aid processing, as well as tools and manuals enabling the design of own setups for algorithm development and evaluation. Documentation is provided for different user levels, in particular for research audiologists, application engineers and algorithm designers. The software runs on standard off-the-shelf hardware including lab setups and portable setups. Being of particular interest for the evaluation of new methods in real-world scenarios, a portable, integrated research platform for openMHA is provided in conjunction with the SBIR project R44DC016247: the Portable Hearing Laboratory (PHL). This contribution introduces openMHA and discusses current use cases and possible application scenarios of the software framework as well as the PHL. The first steps of using the software will be demonstrated including the setup of a basic configuration, manipulation of processing parameters at runtime - locally or from a remote devices such as other computers and smartphones. An overview over the tools, provided with the software package, that address the different user groups will also be given. The aim is to learn about the flexible configuration and remote control of openMHA running a typical hearing aid processing chain, to get an insight into basic steps required to build your own hearing aid research setup, and to gain an overview of application possibilities of openMHA in hearing research.

CX-3. Hearing Aid Research with the Portable Hearing Laboratory (PHL)
Chaslav Pavlovic*, Reza Kassayan1, Nick Michael2, Hendrik Kayser3, Volker Hohmann2
1BatAndCat Sound Labs, 2BatAndCat Sound abs, 3Carl von Ossietzky University of Oldenburg

We report here on the development of a portable hearing aid research platform that has been accomplished in the SBIR Phase II Project R44DC016247; as well as on the early development and plans for its Phase III continuation (currently completing its Year 1), which will develop the successor generation of the current platform with several extensions.

The project builds on the openMHA software (its distribution for ARM processors) included in an optimized Linux system developed in project R01DC015429 to provide a complete portable and wearable software-hardware master hearing aid device needed for development of new innovative solutions for assisted hearing. We refer to this device as the PHL for Portable Hearing Laboratory. The PHL consists of a ARM Cortex A8 based processing unit (open-source single-board computer BeagleBone Black) and a codec set able to support hearing aid architecture of up to 6 microphones and 4 speakers, a further development of the open-source audio board “Cape4all”. It is accompanied by a binaural 4- microphone extremely low noise
BTE hearing aid set. Additionally, it features stereo line in and line out connections and also interfaces to different headset form factors. In particular, it supports the ITE stereo headset developed at the University of Oldenburg (“Hearpiece”). The device can be remotely controlled with a smart phone and computers.

Phase III work on the PHL, greatly builds on the experience and results of a large number of studies by various laboratories across the world done on the current Phase II version of the PHL device. We will present the most important conclusions reached in these studies with respect to the PHL capability and performance. We will then focus on the current electroacoustic and software capabilities of both Phase II and Phase III device and the type of studies it already makes possible and will further enable.

**CX-4. Hearing Aids Objective Outcomes (HO 2 – /haːˈt/ ) Apps**

_Harinath Garudadri*1, Vy Nguyen1, Martin Hunt1, Wayne Phung1, Shrea Chari1, Varsha Rallapalli2_  
1University of California - San Diego, 2Northwestern University

**Background:** Modern digital hearing aids (HAs) are extremely effective in compensating for hearing loss (HL) in most situations except in those that hearing-impaired listeners need most – multiple talkers and background noise. The adage from business management “If you can’t measure it, you can’t improve it,” is very apt for modern HAs. There is a need to build successful, clinically relevant tools such as QuickSIN, Modified Rhyme Tests, Acceptable Noise Level, etc. in an integrated research environment using open-source subband amplification, feedback cancellation, frequency translation, and other such tools. Further, such tools shall support measuring objective outcomes in a repeatable, and reproducible manner.

**Approach:** We developed a suite of Hearing Aids Objective Outcomes (written as HO2 and pronounced as [hot] /haːˈt/ ) Apps. (i) We developed a multi-lingual data collection tool for audio and audio-visual stimuli based on minimal contrast sets (MCS) of words in a given language. Each word in a given subgroup of words in MCS differs in one and only one (acoustic) phonetic feature that results in a different meaning – phonemics of the given language. (ii) The resulting stimuli content is uploaded to HO2. (iii) HO2 provides functionality for the investigations to configure (a) unaided and aided conditions; the aided conditions comprising multiple fitting strategies for a given HA sound processing algorithm and a given fitting strategy for multiple HA sound processing algorithms; (c) select audio stimuli with and without video; (d) add different noise types at different SNR levels; and (e) automatically generate word-level accuracies, phonetic confusion matrices, and broad phonetic class confusion matrices. (iv) The broad phonetic classes are based on the phonemic classes for a given language and include both place of articulation and manner of articulation during speech production. (v) The confusion matrices enable researchers to use HO2 iteratively by selecting stimuli that address perception errors at individual and at group levels and investigate various static and dynamic aspects of subband amplification. (vi) HO2 can save the above conditions in a configuration file and share the file with other researchers for repeatability, reproducibility, and contributions to novel research paradigms.

**Results:** We have validated the tool for software correctness. We will present feasibility data from users, demonstrate HO2 Apps in our presentations, and provide hands-on training to interested IHCON attendees. Subgroups can focus on one or more features of HO2 Apps and share their configuration files with others.

**CX-5. Open-Source Speech Processing Platform (OSP) Researcher Apps**

_Harinath Garudadri*4, Dhiman Sengupta1, Martin Hunt1, Wayne Phung1, Alice Sokolova1, Kuanlin Chen1, Arthur Boothroyd2, fred harris1, Varsha Rallapalli3_  
1University of California - San Diego, 2San Diego State University, 3Northwestern University
**Background:** While modern digital hearing aids (HAs) are extremely effective in compensating for hearing loss (HL), frequent complaints from the users persist, notably poor speech intelligibility in noisy environments and high cost, among other issues. However, the signal processing and audiological research needed to address these problems has long been hampered by proprietary development systems, underpowered embedded processors, and the difficulty of performing tests in real-world acoustical environments. To facilitate existing research in hearing healthcare and enable new investigations beyond what is currently possible, we have developed a modern, open-source hearing research platform, Open Speech Platform (OSP). This contribution introduces the basic, and advanced signal processing algorithms and describes Researcher Apps aimed at advanced audiology research.

**Approach:** The basic HA features currently supported in OSP comprise three classes of subband amplification (i) temporal (FIR based) 6-band system that can be extended, (ii) FFT based multiband openMHA from Oldenburg, and the temporal (multirate) 11-band system; a suite of “least means squares”-based acoustic feedback cancellation (AFC). The advanced HA features in OSP include real-time beamforming between left and right microphones, and a novel frequency translation module. This module (called Freping, a portmanteau for Frequency Warping) is based on a modified all-pass network that performs frequency compression and expansion in each subband independently, based on a single parameter in the range of $+/-.1$. Positive values enhance the temporal fine structure, while negative values reduce the spectral footprint to accommodate elevated hearing thresholds in some bands. Low Freping values (around $+/-.01$) provide an added stable gain of about 10 dB, allowing audiologists to prescribe more amplification without “whistling” artifacts. Audio quality changes are barely perceivable at such Freping values. Higher Freping values (around $+/-.2$) may improve speech intelligibility at lower amplification prescriptions, but additional research is required to understand the benefits. These researcher apps have been used successfully in self-fitting research such as Goldilocks—a search and select approach or amplification.

**Results:** We will demonstrate Researcher Apps running in real-time. The app includes NAL-NL2 prescriptions for the standard audiograms developed by the “International Standards for Measuring Advanced Digital Hearing Aids” and the ability to adjust both amplification and Freping values in multiple bands. We will provide hands-on training to create multiple prescriptions that can be used in intelligibility and ecological momentary investigations using the OSP.

**CX-6. CCI-MOBILE: Bimodal Synchronization and Algorithm Testing with the CI/HA Research Platform**

*Ria Ghosh*$^{*1}$, Prof. John Hansen$^1$

$^1$University of Texas at Dallas

Previous studies have reported improved localization and better speech perception when the unilateral CI is coupled with a hearing aid in the contralateral ear. This is referred to as bimodal presentation of speech. A major obstruction to accurate source localization for bimodal CI users is the distortion of interaural time and level difference cues (ITD and ILD), and limited ITD sensitivity. Hence, it is necessary to develop and test algorithms that provide better localization and sound source identification cues. Various research interfaces developed by either academic or industry sponsored research teams support proposed signal processing and psychoacoustic investigations but have limited ability to efficiently validate bimodal algorithms. Platforms that support bimodal testing are either not portable or only provide limited features due to proprietary parameters/routines. Thus, the open-source, portable signal processing platform, CCI-MOBILE developed by UT-Dallas, enables electric and acoustic bimodal stimulations simultaneously, providing researchers the freedom to explore new technology and scientific paradigms.
In this study, we perform a systematic verification of the synchronized bimodal (electric-acoustic) output in an authenticated and efficient manner to ensure effective left/right processing pipeline timing support of algorithmic and experimental investigations focused on localization.

CCi-MOBILE captures audio in real-time through two behind the ear (BTE) microphones via an audio codec and streams the digitally processed signal through a UART port to the computing platform (PC/ smartphone). Sound processing routines send the RF data through one channel and digitized acoustic data through the other channel via the UART port. The delays and system clock settings are tuned such that the middle and inner ear frequency dependent delay are taken into consideration, while synchronizing the HA output with the CI output. The output to the CI is sent to the RF coil to deliver biphasic pulses in a continuous interleaved manner (CIS). The amplified HA output is sent to the HA transducer channel, after Wide Dynamic Range Compression (WDRC). Results were verified using an oscilloscope to demonstrate the timing validity of left and right output signals superimposed on each other, proving evidence of simultaneous CI/HA synchronization. Bimodal stimuli synchronization was also verified by comparing Electric and Acoustic Stimulation (EAS) outputs on an oscilloscope to demonstrate equalization of the input and output time delay for both channels, as well as with formal subjective tests.

3:00 p.m. - 4:30 p.m. Demos and Posters

**POSTER SESSION CX: Open-Source Audio-Processing Satellite Workshop**

**CXP01 Automatic analysis tools for exploring own-voice and near-ear sound pressure level distributions**

Jule Pohlhausen*, Inga Holube¹, Sven Franz¹, Jörg Bitzer¹  
¹Institute of Hearing Technology and Audiology, Jade University of Applied Sciences, Oldenburg, Germany

Exploring acoustic conditions and listening experiences of people in their everyday environments has drawn a lot of attention. Our laboratory developed the smartphone-based ecological momentary assessment (EMA) system olMEGA which stores smoothed acoustical features (root mean square, RMS; auto and cross-power spectral density, PSD; zero crossing rate, ZCR) on a reduced time scale to preserve the privacy of the test participants. The recordings were performed by two head-worn, near-ear microphones attached to glasses. Simultaneous to the recorded acoustical features, subjective assessments are taken in situ by surveys on the smartphone.

This contribution shows analysis results of 13 near-ear recordings of hearing-impaired participants (aged 50 to 75 years) in an EMA study. The collected acoustical features and subjective assessments of approx. 4 days per participant were extracted from the smartphones and transmitted to a database server for further analysis. The database server facilitates data management and enables international collaboration. Furthermore, new features can be extracted automatically that are of interest for studying the impact of natural, acoustic environments on the communication abilities of elderly people. For this contribution, an algorithm detecting the test participant’s own voice (Own Voice Detection, OVD) and an algorithm that extracts the sound pressure level (SPL) in order to measure the sound exposure were applied to the EMA data. The OVD is based on a machine learning algorithm and a set of acoustical features derived from the olMEGA features. The evaluation of the OVD with a manually labeled real-world recording of one full day showed reliable and robust detection results.

The analysis of the EMA data shows that the grand mean percentage of predicted own-voice audio segments (OVS) during one day was approx. 10% which corresponds well to other published data. The grand
median SPL across all participants and recording days was 50.3 dB(A). The OVS had a small impact on the median SPL over all data. However, for short analysis intervals, significant differences of up to 30 dB occurred in the measured SPL, depending on the proportion of OVS and the SPL of the background noise.

**CXP02 Influence of Number of Hearing Aid Compression Channels on Spatial Release from Masking**

Marc Brennan*, Sarah Garvey†, Ava Feller‡

1University of Nebraska

The objective of this study was to understand how access to spectral information under conditions of hearing aid amplification can influence spatial release from masking (SRM). Previous studies have demonstrated that listeners with normal hearing take advantage of differences in the spectrum of speech and noise, dips in the noise level, and spatial cues that can occur between sound sources to improve their understanding of speech. Listeners with sensorineural hearing loss are less able to take advantage of some of these spatial cues. Possibly by not considering the influence of access to spectral information under conditions of amplification, hearing aids provide—at best—marginal improvement in SRM. This experiment examined whether modifying the number of compression channels can improve SRM. The experimental conditions were based on prior data that documented improved measures of spectral resolution, estimated with psychophysical tuning curves, with 16-relevant to both an unaided condition and to 4-channels of compression. For this study, we obtained sentence recognition for AzBio sentences presented in front of each participant. Simultaneous masking from either a male or female speaking the rainbow passage were presented in both collocated and spatially separated (masker at 30 degrees) conditions. The maskers were fixed at 50- or 70-dB SPL while adaptively varying the level of the sentences. Test conditions included unaided and four and 16 channels of compression with the Tympan open-source hearing aid platform. We hypothesized that SRM would be greatest for the 16-channel compression condition, followed by the 4-channel compression condition, and poorest for the unaided condition. While data collection is ongoing, preliminary data for 13 subjects with sensorineural hearing loss suggests a similar SRM between the unaided and 4 channels of compression conditions. SRM for these two conditions was similar to 9 subjects with NH (unaided). Inconsistent with the hypothesis, SRM was poorest for the 16-channel compression condition, indicating that changes in access to spectral cues across conditions was not the primary contributing factor to sentence recognition. Potential alternative factors will be discussed, including the presentation of supporting evidence from electroacoustic analyses of interaural level and timing differences, audibility (speech intelligibility index), better ear advantage, and binaural unmasking. Initial results suggest that reductions in interaural level differences may have contributed to poorer SRM for the 16-relative to 4-channels of compression. These results provide proof of concept for the Tympan as a research platform.

**CXP03 Tympan Electroacoustic and Behavioral Measurements**

Sara Harris*, Odile Clavier2, Ryan McCreevy†, Marc Brennan2, Chip Audette4, Joshua Alexander4, Eric Yuan2, Stephen Neely†

1Boys Town National Research Hospital, 2Creare LLC, 3University of Nebraska, 4Purdue University

An audio processing platform called Tympan that allows implementation of custom algorithms directly on wearable hardware and modification of parameters via a simple user interface was developed as part of the NIH-NIDCD efforts to support innovation in hearing aid research through the development of open-source audio processing platforms. Recent revisions of the Tympan included the addition of wearable hardware with receivers and directional microphones at the ear-level. Tympan performance was compared to a commercial hearing aid with similar features. Hearing aid output of both devices was obtained following the procedures defined in the ANSI standard and Verifit 2 manual and met manufacturer specifications.

14 adults with mild-moderate sensorineural hearing loss participated in behavioral testing (7 female; age range 34-87 years). Gain was set according to the NAL-NL1 prescriptive method for both devices for each participant using the AudioScan Verifit 2. There was a good match between fit and target for both devices, except at 2 kHz where the output of the Tympan was higher than the prescriptive target. Two lists of 10 CASPA consonant-vowel-consonant words (Mackersie et al. 2001) were presented in quiet at three input levels (55, 65, 75 dB SPL). Percentage correct improved from 55 to 65 dB SPL and was better aided than unaided at 55 dB SPL. AzBio sentences (Spahr et al. 2012) were presented at 70 dB SPL in 10-talker babble at 5 dB signal-to-noise ratio. Word
recognition was better aided than unaided. The Tympan performed similarly to the commercial hearing aid in all conditions. Electroacoustic measurements included evaluation of the hearing aid speech perception index and speech quality index (HASPI and HASQI).

These results are consistent with results from a previous study comparing an earlier version of the Tympan to a different commercial hearing aid. Though the previous behavioral results implied that the two hearing aids performed equally, the qualitative impressions of the Tympan were poorer than the commercial hearing aid overall. In the current study, a sound quality questionnaire was included to quantify perceived characteristics of the devices. Several patterns were consistent in responses to all questions: (1) ratings increased as the difficulty of the test condition decreased, (2) ratings for difficult conditions were higher for aided compared to unaided conditions, (3) participants rated the Tympan similarly to the commercial hearing aid. The responses on the questionnaire provide evidence that, on average, participants thought the Tympan was similar to the commercial hearing aid.

CXP04 Ecological Momentary Assessment App for the Open-source Speech processing Platform (OSP)
Vy Nguyen*, Wayne Phung†, Dhiman Sengupta†, Martin Hunt†, Varsha Rallapalli†, Harinath Garudadri†
†University of California - San Diego, ‡Northwestern University

Background: EMA incorporates multiple experience sampling methods, including real-time outcomes assessments with human subjects. These assessments are typically done via surveys sent to participants multiple times a day across a trial period to gather relevant, in situ feedback about their experiences. EMA has been proven to improve many drawbacks of retrospective recall, diary studies, and other traditional methods. Researchers adopting EMA implementations within the clinical psychology domain and recent hearing-related topics suggest the critical need for developing and supporting EMA within open-source ecosystems. We have developed an EMA system enabling researchers to (i) design EMA surveys using a simple text-based authoring tool and share with others for extensions; (ii) incorporate contexts (e.g., background conditions, GPS, number of talkers, etc.) to dynamically control the surveys in real time, and (iii) log audio parameters (raw, spectra, subband energies, etc.) before, during, and after a given survey.

Approach: cEMA, is an interactive graphical user interface that runs on OSP’s embedded web server framework. We developed an offline authoring tool for Windows, Mac, and Linux operating systems based on YAML, an acronym for “Yet Another Markup Language,” and also a recursive version “YAML Ain’t a Markup Language.” Researchers can leverage many YAML online tutorials and cEMA examples to become proficient in designing EMA investigations for hearing aids research. cEMA has configurable settings that allow researchers to specify surveys’ push logic (e.g., number of surveys per day, specific times, and combinations) and surveys’ context logic (e.g., number of talkers, GPS, background noise, etc.). After researchers have finalized an EMA logic, it is ported to OSP enabling participants to complete surveys either on their smartphones or tablets. Survey results reside within OSP’s internal memory for added security and privacy. In addition, cEMA can passively sense and capture audio and environment characteristics during surveys. This provides researchers additional information on the number of social interactions, speaker turn-taking in discourses, etc., while addressing privacy and security concerns. Further, cEMA results serve as the ground truth to create labeled environments for machine learning and artificial intelligence tools related to emerging hearing aids.

Results: We present cEMA design methodology and provide hands-on training to IHCON audience. Researchers can use existing configuration files, edit them to add new functionalities, or create entirely new cEMA investigations using example YAML files. We will also provide training to analyze EMA surveys based on individual and group-level responses.

CXP05 Development and Feasibility of Using an Open-Source Portable Hearing Aid with EMA in the Real World
Erik Jorgensen*, Dhruv Vyas†, Yumna Anwar†, Octav Chipara†, Yu-Hsiang Wu†
†The University of Iowa

Objective: The purpose of this study was to investigate the feasibility of using a portable, open-source hearing aid paired with ecological momentary assessment and soundscape recording in the real world. Open-source hearing aids and ecological momentary
assessment will enable the transparent and collaborative development and real-world testing of hearing aid processing algorithms. This study describes platform development, compliance, feasibility, participant experiences, and potential challenges and use cases.

**Design:** This study used the Portable Hearing Aid Lab (PHL), a BeagleBone computer with wired receiver-in-the-canal earpieces, running the Open Master Hearing Aid software. Participants (with and without hearing loss) were asked to collect data in 5–10 complex listening environments over a 7–10-day period. In each environment, participants wore the PHL and completed two ecological momentary assessments on a smartphone over the course of 30 minutes. Between each momentary assessment for participants with hearing loss, the smartphone triggered the PHL to switch gain settings from an aided (fit to NAL-NL2 targets) to unaided (acoustically transparent) condition, or vice versa, testing the possibility of within-environment A/B comparisons of hearing aid algorithms in the real world. The PHL also recorded the environment.

**Results:** 12 participants with hearing loss and 10 with normal hearing completed the study. Most participants were able to operate the devices, but not universally. 266 EMA surveys were completed, but only 127 had associated recordings. Absent recordings were determined to typically be the result of user error. Only 1 participant (hearing loss group) declined to complete the study because the devices were too complicated. Participants with normal hearing completed EMAs in, on average, 7 complex listening environments and participants with hearing loss completed EMAs in, on average, 5 environments. Compliance did not differ significantly between groups. Data were collected at home (46%), in restaurants and bars (14%), transportation (10%), shops (8%), work (6%), and other environments, suggesting participants could use the device successfully in different places. Participants also wore the PHL for a variety of listening activities, including passive speech listening (26%), one-on-one conversation (22%), music listening (17%), and group conversation (9%). Compliance tended to increase over the course of the study as training was adapted.

**Conclusions:** Using an open-source hearing aid platform paired with EMA and sound recording for within-environment A/B hearing aid algorithm comparison is feasible. However, pitfalls were noted, particularly user error, user frustration, and device limitations. Training and streamlined interfaces and procedures are critical for real-world use.

**CXP06 Group Conversation Enhancement Using Wireless Microphones and the Tympan Open-Source Hearing Platform**  
Ryan Corey¹, Andrew Singer¹  
¹University of Illinois Urbana-Champaign

Some of the most challenging listening environments, especially for people with hearing loss, are group conversations in crowded spaces, such as restaurants. Conventional hearing aids perform poorly in noisy environments because their microphones capture a mixture of speech and unwanted background noise. Remote microphones can improve intelligibility in noise by transmitting speech directly from a talker to the ears of the listener. However, commercial remote microphones are unsuitable for group conversations because they work with only one talker at a time and do not preserve spatial cues such as interaural time and level differences. These spatial cues are especially important for group conversations because they help the auditory system to follow speech from multiple talkers.

We have recently proposed an immersive multitalker remote microphone system that combines the low noise of remote microphones with the realistic spatial cues of earpiece microphones [1]. A set of adaptive filters processes the remote microphone signals to match the magnitude and phase of the earpiece signals. Because it relies on the earpieces as references, the system does not need to explicitly localize or track the talkers and the enhanced remote signals can be seamlessly mixed with the live sound at the earpieces. A basic version of the system was implemented in real time on the Tympan open-source hearing aid development platform [2].

In this presentation, we extend the previous implementation to improve performance in real-world environments. In particular, we consider robustness against talker and listener motion, crosstalk between two or more talkers, delayed auditory feedback of the listener’s own speech, and double-talk. We also demonstrate a cooperative conversation enhancement system for multiple users with listening devices. We compare experimental results from laboratory equipment with the real-time implementation on the Tympan hardware platform. [This work was supported in part by an appointment to the Intelligence Community Postdoctoral Research Fellowship Program at the...
University of Illinois Urbana-Champaign, administered by Oak Ridge Institute for Science and Education through an interagency agreement between the U.S. Department of Energy and the Office of the Director of National Intelligence.]

References:

CXP07 In Situ Tuning of an Adaptive Feedback C canceler Using the Tympan Open Source Audio Processor
Joshua Alexander*, Steven Neely†, William Audette‡
1Purdue University, 2Boys Town National Research Hospital, 3Creare LCC

Modern hearing aid design is critically dependent on feedback management, particularly when the coupling to the ear is wholly or partially unoccluded (i.e., ‘open’). To this end, adaptive feedback cancelers (AFCs) are necessary for managing dynamic feedback paths. In order to maximize the effectiveness of AFCs, multiple parameters need to be optimized using models or human testing. We use the Peak Height Insertion Gain (PHIG; Alexander et al. 2022, J. Acoust. Soc. Am. 150(3), 1635-1651) feedback criteria to optimize the parameters of a normalized filtered-x least mean square (NFXLMS) algorithm in situ on a KEMAR manikin using the open-source Tympan processing platform. The NFXLMS algorithm, implemented on the Tympan firmware, uses open-fitting earpieces. The initialized parameters of the NFXLMS algorithm are first determined by an optimization procedure that minimizes the estimation error between the modeled feedback path and the AFC estimated feedback path. In situ measurements of the resulting PHIG as a function of hearing aid gain for a typical mild sloping to moderate hearing loss are then used as input into an optimization algorithm that selects the next set of AFC parameters for tuning. Added stable gain (ASG) is quantified using the parameters that minimize PHIG as a function of gain. In this presentation, results for PHIG and ASG with the Tympan will be compared to published benchmarks from commercial hearing aids.

CXP08 Hearing Aid Fitting and Listening Performance in Spatially Complex Scenes
Laura Hartog*, Dirk Oetting†, Julia Zimmer‡, Theresa Jansen‡, Jörg-Hendrik Bach‡, Hendrik Kayser‡
1Hörzentrum Oldenburg gGmbH, 2Carl von Ossietzky University of Oldenburg

The gap between real-world speech reception performance of hearing aid users and results of lab assessments is still large. Furthermore, people with similar hearing thresholds may experience very different benefit and satisfaction from hearing aids. Evaluation and deeper understanding of listening performance in realistic situations remains a crucial step to individualize hearing solutions.

Methods: We used the Portable Hearing Laboratory (PHL) as a research hearing aid. It combines ear-level hearing aid hardware with the open Master Hearing Aid (openMHA) software.

Virtual listening situations with varying acoustic complexity in a realistic, reverberant cafeteria environment were created with a multi-loudspeaker setup and higher-order ambisonics using the Toolbox for acoustic scene creation and rendering (TASCAR). Diffuse background noise and competing talkers were varied in number and level. The generated conditions ranged from a target speech source with two softer interfering sources (+6.3 dB SNR) to a target in loud background noise with two interfering sources (-7.5 dB SNR).

Based on a questionnaire completed in advance, 20 test subjects were classified into self-reported “high performer” and “low performer” groups. The subjects did a word scoring test and a subjective listening effort screening in each acoustic condition. We compared the performance between the unaided case, the subjects’ own hearing aids and the PHL. On the PHL hearing loss compensation was provided according to the trueLOUDNESS fitting method to restore binaural broadband loudness perception. In addition, three different signal enhancement algorithms were implemented: binaural coherence filtering, bilateral adaptive differential microphones (ADM), binaural minimum variance distortionless response (MVDR) beamforming.

Results: Overall, lower word scoring performance was found for the “low performer” group, which also seems to drop faster in more complex conditions than for the “high performer” group. This appears to be independent of the device (own hearing aid/PHL). “Low performers” also report higher listening effort in most of the conditions. Word scoring results indicate that spatial filtering provides a higher benefit for
the “low performers”. For this group, the optimal result was achieved on average with the research hearing aid.

Conclusions: Speech reception results measured in simulated spatially complex listening scenarios reveal self-reported differences in performance with hearing aids. The benefit from spatial filtering is prevalently higher for self-reported “low performers”. However, large inter-individual differences were observed and suggest that further investigation into individual effects is needed to enable individualized selection of speech enhancement methods in the hearing aid fitting process.

CXP09 Realtime Multirate Multiband Amplification for Hearing Aids
Alice Sokolova*, Dhiman Sengupta1, Martin Hunt1, Rajesh Gupta1, Baris Aksanli2, Varsha Rallapalli2, Fred Harris2, Harinath Garudadri1
1University of California - San Diego, 2San Diego State University, 3Northwestern University

Background: Subband amplification is a basic feature of modern hearing aids (HA) to compensate for individual hearing loss. However, the temporal and spectral nature of hearing loss poses many challenges for the HA signal processing designers. Larger number of frequency channels offers more precision and accuracy for fulfilling HA prescriptions, especially for unusual hearing loss patterns, but increases the complexity and reduces battery life. There is also a lack of understanding on how to accurately satisfy attack and release time parameters in a real-time system based on ANSI 3.22 guidelines, to facilitate systematic investigation of these parameters on listener outcomes.

Approach: We present a novel multiband real-time amplification system for HAs using multirate signal processing to minimize the complexity and power consumption of processing multichannel audio. The system offers precise control of the temporal dynamics of WDRC. The filterbank is based on audiometric frequencies comprising eleven half-octave frequency channels spanning five octaves, from 250 Hz to 8000 Hz, uniformly distributed on the logarithmic scale, with each band having a proportionate bandwidth, mimicking the known properties of cochlear transduction. Further, each octave of audio channels is mapped to a different sampling rate, such that each group of bands is processed at the lowest possible sampling rate. Processing lower sub-bands at the original Nyquist rate results in redundant samples.

The proposed multirate system removes this redundancy and offers dramatic reduction in complexity. We implemented a new WDRC subsystem using an automatic gain control (AGC) using a Hilbert Transform for the envelope estimation in each band in the lower sampling domain. This approach accurately satisfies attack and release time specifications for the compression parameters as suggested from ANSI 3.22 guidelines.

Results: We implemented the proposed multirate HA algorithm on the Open Speech Platform (OSP) – an open-source system hearing loss research. On-target measurements on the OSP hardware show that the system runs in real-time, offers better frequency resolution, and with less complexity than prior sub-band amplification tools available in OSP. The proposed multirate filter bank provides a 14x reduction in complexity compared to a single-rate implementation, and has an algorithmic latency of 5.4 ms – well within the conventional threshold for real-time operation. The WDRC subsystem satisfies ANSI 3.22 guidelines within 0.5 ms. We present acoustic measurements (e.g., using Verifit Verification Toolbox, HASQI) to confirm the accurate static, and dynamic behavior of the proposed system for the ISMADHA hearing loss profiles.

CXP10 Designing the Real-Time Master Hearing Aid (RT-MHA) Framework for the Open Speech Platform (OSP)
Dhiman Sengupta*, Martin Hunt1, Harinath Garudadri1, Rajesh Gupta1
1University of California - San Diego

Background: The open-source audio processing tools for hearing aid (HA) research community came to a consensus that the best hardware platform for the next generation of HA research tools should be based on mobile computing platforms. These platforms include single board computers (SBC) like the Raspberry Pi, Qualcomm 410c, the Beaglebone, etc. The reasoning behind choosing these SBC platforms is that they are the ideal combination between the available computation, energy efficiency, and programmability. These SBC platforms achieve the ideal combination by using multiple super-efficient CPU cores, usually two or more, among other processing elements, making these computing platforms, unlike traditional computers. Therefore, it is essential to design the research tools with the hardware in mind.

Approach: This work makes two critical contributions to this area. The first contribution is a detailed
analysis of the best way to set up the operating environment for these SBC platforms to get the most out of the hardware, no matter the open-source HA framework used. Our analysis found that we achieve the best performance when partitioning the computation resources between real-time (RT) and non-RT applications. Using this essential information and others gathered from the analysis, we designed the RT master hearing aid (RT-MHA) framework, the second critical contribution of this work. RT-MHA is a part of the Open Speech Platform (OSP), which has been designed and optimized to utilize the resources available on the SBC best. The RT-MHA framework gives HA algorithm researchers the most usable computational resources while minimally impacting the performance of non-RT applications like the embedded web server on the OSP platform.

Results: We show the different impact mechanisms on a modern-day SBC has on the real-time performance of the MHA algorithm. These mechanisms include the scheduler on Linux, the partitioning of computing resources, and the idling mechanism on SBC. Then we will describe how the RT-MHA framework can optimize the workload given what we found in our analysis to get up to 2 times more computation while being able to guarantee real-time operation.

CXP11 Design Modifications for Distributed Economical Manufacturing and Standards Compliance of an Open-Source Hearing Aid for Children
Kavyashree Venkatesh*, Deval Karia1, Manish Arora2
1Indian Institute of Science, 2Indian Institute of Science, Bangalore India

Speech and language training, with consistent use of appropriate amplification devices are crucial factors for successful rehabilitation of hearing-impaired children. The overall rehabilitation journey is economically and socially challenging endeavour for patients in resource-constrained settings. The cost of hearing aids (HA) can range from ₹20,000 (~$260) to ₹3,00,000 (~$3800), making them unaffordable for patients at the base of the economic pyramid. Societal stigma associated with hearing impairment further compounds these economic challenges. An intervention in the form of an affordable HA, coupled with a smartphone application was proposed in our earlier work to address these issues. This work focuses on the design approach of an affordable HA, derived from the open-source Tympan platform.

Functional and performance requirements for the HA design were derived from a WHO guidance document (World Health Organization, 2017. Preferred profile for hearing-aid technology suitable for low-and-middle-income countries.), and benchmarking of commercially available HAs in the Indian market for profound hearing impairment. These specifications were limited to essential parameters for appropriate amplification. To build upon an existing open-source hearing aid platform and a comparison of multiple available platforms was conducted, which pointed to Tympan Rev-D (an open-source hearing aid developed by Tympan) as the most suitable one.

Tympan Rev-D is a body worn hearing aid based on Teensy-3.6 processor, however, technical, and financial constraints within the Indian manufacturing ecosystem led to Design for Manufacturing (DfM) changes. Further changes were also undertaken to meet derived requirements. The developed HA, built upon Tympan Rev-D was subjected to a battery of tests in accordance with relevant standards (ANSI S3.22-2009 Specifications) using the Fyre-Phonix analyser and the results were found to be meeting all derived requirements. In summary, device had OSPL90 of 128dB, Full-On-Gain of 52dB, Equivalent Input Noise of 26dB, Total Harmonic Distortion of <1.4 % with suitable Frequency Response. Based on clinical immersion activities, a headband model was designed with children as the target user. The DfM changes led to a reduction in the cost of manufacturing by an order of magnitude (~10x) and enabling local manufacturing in India without compromising any of the technical requirements. The team is currently undertaking efforts for size reduction, and a device embodiment design catering to a child’s lifestyle. These design changes will be made available in the open-source domain. This can facilitate distributed economical manufacturing of open-source hearing aid hardware and its wider community adoption.

CXP12 Live OScope: A Debugging Tool for Master Hearing Aid (MHA) Algorithms for the Open Speech Platform (OSP)
Dhiman Sengupta*, Martin Hunt1, Arthur Boothroyd2, Rajesh Gupta1, Harinath Garudadri1
1University of California - San Diego, 2San Diego State University

Background: Developing and implementing signal processing and master hearing aid (MHA) algorithms on open-source audio processing research tools come with challenges, especially when deployed on an embedded system. It is challenging because we have little to no visibility in most open-source audio pro-
cessing research tools once the algorithms are deployed in an embedded system. Our experience found that the most challenging part is understanding why an algorithm does not work when deployed even though it was tested thoroughly in a simulated environment. Therefore, we feel these research tools need an intuitive interface for debugging these algorithms. For that reason, we extended the Open Speech Platform (OSP) framework to include a debugging tool called the Live OScope.

**Approach:** We were inspired by the oscilloscope, an analog tool used to debug signals, when we designed the Live OScope feature of the OSP framework. The Live OScope allows algorithm developers to insert test points in their algorithms for monitoring during run time. The Live OScope feature has two components: the recorder and the display. The recorder is a part of the OSP framework; it continuously records the data from the test points into a circular buffer of a predefined length, usually 1-3 seconds. The recording happens until the display component asks for the data. At which time the data gets streamed from the buffer in its entirety to the display component. After the streaming has completed, the recorder component starts sampling the data again. The display component, an external tool running on a remote device, displays the waveforms captured on the embedded device. Like an oscilloscope, the display component can refresh at a set interval and save the waveform captures.

**Results:** We first show that the Live OScope feature has minimal impact on the real-time performance of the embedded system. Next, we will demonstrate the utilities of this tool through three case studies that heavily utilized this tool, including how this tool is currently used to calibrate the microphones of these embedded devices. Finally, we will have a live demonstration of the Live OScope tool.

5:00 p.m. – 7:00 p.m.

**SESSION C3. Outcome Measures Reflecting Real-Life Listening**

**Session Chair:** Karolina Smeds

**C3-1. Understanding Communication Difficulties from an Egocentric Perspective**

*Christi Miller*, Calvin Murdock, W. Owen Brimijoin, Vamsi Krishna Ithapu, Nava Balsam, Thomas Lunner

1Reality Labs, Meta

An Augmented Reality (AR) platform is a system of interdependent technologies (e.g., audio, eye-tracking, computer vision, etc.), which enable digital objects to be placed in our real-world surroundings. These digital objects may provide assistance by overlaying enhancements to natural auditory objects in the scene, but the classic hearing device problems of estimating listener effort and identifying signals-of-interest remains. An AR platform in the form of glasses could support a large number of widely spaced microphones, forward and eye-facing cameras, inertial measurement units and other motion tracking hardware, and many other sensors. These sensors could be used to shed light on what sounds a listener wishes to hear, and whether they are having difficulty hearing them, but only if this information is optimally combined with a deeper understanding of natural conversation behavior.

To this end our team has taken advantage of an AR glasses platform to create a number of egocentric datasets capturing conversation in difficult listening situations, utilizing similar types of data that future AR hearing devices could be able to capture. In a recent study, we used this approach to study the effects of noise level and hearing loss on communication behaviors. Communicators with and without hearing loss were recruited in groups (i.e., they were familiar with one another), and participated in a 1-hour conversation while background levels randomly varied in a mock restaurant space. A glasses research device, Aria, collected egocentric data with a variety of sensors (i.e., microphones, forward-facing cameras, eye-tracking cameras,
inertial measurement units), combined with close-talk microphones. Hypotheses were established a-priori about how behavior would change with increases in noise level and/or hearing loss, and regarded metrics from voice activity, head motion/position, and eye gaze. The data is being analyzed using human and automated annotations, combined with statistical and machine learning approaches with the eventual goal of leveraging these statistics to better understand what signals listeners wish to hear and how much difficulty they are having during conversations.

**C3-2. Listening efficiency: A novel outcome measure reveals how hearing-impaired listeners are disproportionately affected by moderate, ecologically relevant levels of background noise**

*Ian Wiggins*

1University of Nottingham

What matters in real-life listening is not just whether an attended target is heard correctly, but also the amount of effort expended in doing so. These two dimensions can be usefully integrated in a measure of “listening efficiency”: conceptually, the amount of accuracy achieved per unit of effort expended. A measure of listening efficiency may have practical utility for assessing aided performance because it: 1) jointly reflects both accuracy and effort; and 2) is sensitive to differences in performance at or near ceiling levels of speech intelligibility.

We have developed a novel approach to quantifying listening efficiency based on the rate of evidence accumulation towards a correct response in a linear ballistic accumulator model of choice decision making. Estimation of this measure within a hierarchical, Bayesian framework confers further benefits, including full quantification of uncertainty in parameter estimates, as well as improved estimation at the individual-subject level through the borrowing of power from the group level. We have implemented this model using the statistical computation platform Stan.

In a series of experimental studies, we examined speech-in-noise performance of normally-hearing (NH) listeners, hearing-aid (HA) users, and cochlear-implant (CI) users. Participants listened to reverberant target sentences in continuous background noise (open-plan cafeteria recording) at ecologically relevant signal-to-noise ratios (SNRs): +20 dB (“Easy”), +10 dB (“Medium”), and +4 dB (“Hard”). For the NH and HA groups, despite mean accuracy remaining high (>94%) in all conditions, listening efficiency was markedly poorer in the HA group, the gap between NH and HA growing wider with each increase in background noise level. CI users were also highly sensitive to background noise level, but in addition were much less efficient listeners compared to the NH and HA groups even in favourable acoustic conditions (+20 dB SNR).

We argue that listening efficiency is a conceptually well-motivated measure that is easy to measure in practice and is well suited to quantifying performance in realistic acoustic conditions. It thus holds promise as a tool to support the development and evaluation of a new breed of hearing technologies that aim to alleviate suprathreshold listening difficulties.

**C3-3. Comparing In-Ear EOG for Eye-Movement Estimation with Eye-Tracking: Accuracy, Calibration, and Speech Comprehension**

*Sergi Rotger-Griful*¹, *Martin A. Skoglund*¹, *Martha M. Shiell*¹, *Gitte Keidser*¹, *Martin Andersen*², *Mike Link Rank*²

¹Eriksholm Research Centre, Oticon A/S, Denmark, ²T and W Engineering A/S, Denmark

This presentation details and evaluates a method for estimating the attended speaker during a two-person conversation by means of in-ear electro-oculography (EOG).
Twenty-five hearing-impaired participants were fitted with moulds equipped with 6 EOG dry electrodes (in-ear EOG) and wore eye-tracking glasses while watching a video of two life-size people in a dialogue solving a Diapix task. The dialogue and background noise were directionally presented at 60 dB SPL. During three conditions of steering (none, in-ear EOG, conventional eye-tracking), participants’ comprehension was periodically measured using multiple-choice questions. Based on eye movement detection by in-ear EOG or conventional eye-tracking, the estimated attended speaker was amplified by 6 dB. In the in-ear EOG condition, the estimate was based on one selected channel pair of electrodes out of 36 possible. A novel calibration procedure introducing three different metrics was used to select the measurement channels. The in-ear EOG attended speaker estimates were compared to those of the eye-tracker.

Across participants, the mean accuracy of in-ear EOG estimation of the attended speaker was 68%, ranging from 50% to 89%. Based on offline simulation, it was established that higher scoring metrics obtained for a channel with the calibration procedure were significantly associated with better data quality. Results showed a statistically significant improvement in comprehension of about 10% in both steering conditions relative to the no-steering condition. Comprehension in the two steering conditions were not significantly different. Further, better comprehension obtained under the in-ear EOG condition was significantly correlated with more accurate estimation of the attended speaker.

In conclusion, this study shows promising results on the use of in-ear EOG for visual attention estimation with potential for applicability in hearing assistive devices.

C3-4. Audio-Visual Scene Analysis in Listeners with Normal and Impaired Hearing

Axel Ahrens*, Nadia Fons Christensen1, Adam Westermann2, Virginia Best3, Torsten Dau4, Tobias Nehrer1

1 University of Southern Denmark, 2 WS Audiology, 3 Boston University, 4 Technical University of Denmark

In many everyday listening situations, listeners face the challenge of having to follow a conversation that may be embedded in a mixture of competing speech. Listeners with hearing loss are known to have large difficulties with such a task, and several environmental factors may increase this difficulty, including the number of talkers and reverberation. Here, we investigated how well young listeners with normal hearing and older listeners with hearing loss can follow a conversation in audio-visual scenes differing in terms of the number of concurrent talkers and the amount of reverberation. The number of talkers was either one, three or five. The reverberation conditions reflected an anechoic room, a living room and an unfinished-concrete room. The participants’ task was to find and locate an ongoing story (based on the general topic) in a mixture of other stories. Three-dimensional audio-visual scenes were simulated and presented via head-tracked headphones and virtual-reality glasses. The hearing-impaired participants were tested with individual audibility compensation based on the NAL-RP fitting rule. The primary outcome measure was the time taken to identify the location of the target talker (i.e., the response time). The secondary outcome measure was the localization accuracy. For both groups, response times were longer when more talkers were present. Small amounts of reverberation did not affect the response times, while more reverberation led to increased response times when many talkers were present. While response times were longer for participants with hearing loss, no group differences with regards to the effects of the number of talkers and reverberation were observed. Regarding localization accuracy, participants with hearing loss showed poorer performance overall, and there was a clear effect of the number of talkers but not of reverberation. Overall, this new task may provide insight into the challenges experienced by listeners with hearing loss when tracking a conversation of interest in a complex audio-visual scene.
What is Effort in the Context of Interactive Communication, and How Should It Be Considered in Hearing Rehabilitation?

Timothy Beechey*

1University of Nottingham

Effort is an important consideration in hearing rehabilitation, including as a measure of hearing device benefit. Effort is most often thought of within hearing science in relation to listening, but hearing devices are frequently used to assist with interactive communication. It is therefore important to consider how effort may manifest in interactive communication: the form effort may take, the function of effort and how effort expenditure might affect other outcome measures. Any theory of effort in interactive communication should take account of behavior to an extent not typically considered in the study of listening effort. In relation to listening, effort has come to be treated as synonymous with cognitive resource allocation. But a listener may employ behaviors which extend outside the head to improve auditory perception. Such behaviors range from turning one’s head, to turning up the volume of a television, or closing a window. That is, a listener can move their body and interact with the environment. It seems reasonable to expect that such extra-cognitive tactics may be more effective, in many situations, than reallocation of cognitive resources because they can improve signal-to-noise ratios. In interactive communication many behaviors are afforded to an interlocutor which are not available during one-way listening, such as asking questions, seeking repetitions and clarifications, and eliciting changes the behavior of a conversation partner. The availability of behavioral strategies means that in many cases interlocutors can maintain successful communication in highly adverse conditions. When communicative behaviors are available, communication ability may be limited by interlocutors’ capacity and willingness to employ effective behaviors, not only by sensory impairment. This presentation will describe experimental results illustrating how communication can be maintained by behavioral strategies and will sketch a theory of communication effort based on the concept of affordances drawn from Gibsonian ecological psychology, and perceptual control theory which holds that behavior affects perception in addition to being affected by it. Implications for measuring hearing performance and hearing device benefit will be discussed.

7:00 p.m. - 8:00 a.m. Dinner
8:00 p.m. - 10:00 a.m. Posters and Social

Saturday, August 13

7:00 a.m. – 8:00 a.m. Breakfast
8:30 a.m. – 10:00 a.m.
D1-1. Machine Learning for Speech Signal Processing on Hearing Devices

_Timo Gerkmann*\(^1\)

*Universität Hamburg

Background noise, reverberation, and competing speakers often present a major challenge for users of hearing devices. To mitigate these effects and facilitate speech communication, modern devices typically employ signal enhancement algorithms. In recent years, the advent of deep learning techniques has dramatically transformed the field of signal enhancement, and what used to be considered beyond reach is now well within the realms of possibility. In this talk we will present some of the recent trends proposed and investigated by our group within this context.

We begin with recent advances in single-microphone source separation, showing how modern machine learning approaches allow for high-quality separation of competing speakers. We then address algorithmic latency which is an important factor for hearing devices. Algorithmic latency depends on the segment lengths employed for spectral analysis and synthesis. While in traditional magnitude-centric approaches shorter segments decrease performance, we show that neural networks allow for enhancing both magnitude and phase on short segments yielding both a low algorithmic latency and an improved performance. Next, we question the optimality of the traditional signal processing chain of beamforming and postfiltering in multimicrophone speech enhancement. We show that with neural networks more powerful nonlinear joint spatial-spectral filters can be learned that outperform the traditional sequential spatial and spectral processing. Finally, we present the very recent and powerful approach of score-based generative models, where we were among the first groups to tailor this approach for speech enhancement with impressive results. We show that this method can be flexibly used in both denoising and dereverberation tasks.

D1-2. Immersive Conversation Enhancement Using Binaural Hearing Aids and External Microphone Arrays

_Ryan Corey*\(^1\), Manan Mittal\(^1\), Andrew Singer\(^1\)

*University of Illinois Urbana-Champaign

Hearing aids alone perform poorly in crowded, noisy environments such as restaurants and conference centers. Devices equipped with microphone arrays can use beamforming to reduce noise by focusing on a single talker of interest. However, beamforming can only isolate one talker at a time and that talker must have a known direction relative to the listener, making it difficult to use in dynamic group conversations. Furthermore, high-performance microphone arrays must be physically large in order to meaningfully reduce noise, making array-equipped personal devices impractical for day-to-day use.

However, thanks in part to the rise of hybrid meetings and classrooms during the COVID-19 pandemic, many spaces are now equipped with high-performance recording equipment including large microphone arrays. Although intended for conferencing applications, these arrays could also be connected to hearing aids to improve intelligibility in noisy environments. Using future high-throughput, low-latency wireless protocols, hearing aids could automatically connect to in-room infrastructure to improve intelligibility without requiring the user to carry any extra devices.
In this presentation, we demonstrate a group conversation enhancement system designed for crowded spaces such as a cafeteria. One or more conversation participants wear binaural hearing aids that can receive and process external signals. An array processing algorithm uses a set of conference-room microphone arrays to isolate speech originating from a designated area, e.g. a single dining table, and remove sounds from elsewhere in the room. An adaptive filter spatializes the array output signals so that the spatial cues, including interaural time and level differences, of all talkers match the corresponding cues at the earpiece microphones. The filter is continuously updated to track motion of both the talkers and listeners. The proposed system is demonstrated using recordings of live human talkers conversing in a noisy environment with behind-the-ear microphones and several large microphone arrays. [This work was supported by the Discovery Partners Institute and by an appointment to the Intelligence Community Post-doctoral Research Fellowship Program at the University of Illinois Urbana-Champaign, administered by Oak Ridge Institute for Science and Education through an interagency agreement between the U.S. Department of Energy and the Office of the Director of National Intelligence.]

**D1-3. Incorporation of External Microphones in Hearing Aid Processing for Robust Noise and Interferer Reduction**

*Wiebke Middelberg, Simon Doclo*

*1University of Oldenburg, Dept. of Medical Physics and Acoustics and Cluster of Excellence Hearing4all, Germany*

In many speech communication applications, background noise and competing speakers decrease the intelligibility of a target speaker, especially for hearing-impaired persons. Hence, reducing noise and interfering speakers is important in hearing aids. To improve the speech enhancement performance, it has been proposed to exploit one or more external microphones in conjunction with the hearing aid microphones. A promising processing scheme incorporating external microphones into a beamformer is the so-called generalized sidelobe canceller with external speech references (GSC-ESR) [1,2]. The GSC-ESR relies on the assumption that the target position relative to the hearing aid user is known but does not require knowledge about the position of the external microphone(s) nor the relative position of the target speaker to the external microphone(s). Based on pre-processed hearing aid and external microphone signals a joint beamformer is computed. To actively suppress interfering speakers, we have proposed a minimum power distortionless response (MPDR) implementation of the GSC-ESR, which outperformed a minimum variance distortionless response (MVDR) implementation in terms of interferer reduction when the interfering speaker is dominant. However, it was also observed that the MPDR implementation led to undesired speech distortion at high signal-to-interferer ratios (SIRs), whereas the MVDR implementation exhibited a large robustness at high SIRs. To merge the interferer reduction performance of the MPDR implementation and the robustness of the MVDR implementation, in this contribution we propose an SIR-dependent hybrid processing scheme. We experimentally investigate the influence of different weighting approaches of the MVDR and MPDR implementations of the GSC-ESR to achieve optimal noise and interferer reduction with a high robustness in different acoustic scenarios.

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**References:**


D1-4. Performance of an Automatic Head-Movement Detector during Multi-Talker Conversations with Hearing Aids

Nathan Higgins*1, Angkana Lertpoompunya1, Erol Ozmeral1, David Eddins1
1University of South Florida

Head movement while listening to a multi-talker conversation is a uniquely important source of information regarding listener intent. Better understanding of an individuals’ listening goals has the potential to improve individualized real-time signal processing in complex listening environments and improve the user experience. One of the first steps towards realizing these larger goals is the development of automated head-movement analysis and the impact of individual differences.

To simulate a natural multi-talker listening environment, audio and visual stimuli were extracted from a freely available, 4-talker video podcast available on the internet. The conversational content was subdivided into 24 individual trials, each was 90 s in duration. Participants (10 normal hearing, 20 hearing impaired) wearing hearing aids were seated in the center of a 24-channel loudspeaker array (15° separation) that included video monitors co-located at the 13 frontal hemifield positions. Audio and video stimuli for each conversational talker were presented from one of four locations (+/- 15, 30, 45, 90°) relative to the participant throughout the trial. Participants were instructed to follow the conversation as it jumped from talker to talker, and they were given multiple choice questions on the content following each trial. Head movements were collected using an Optitrack V-120-Trio camera.

Raw head-movement time courses for each trial were smoothed and transformed to velocity using a 200 ms sliding window. Yaw, roll, and pitch trajectories were combined into a single weighted average, and local peaks (>2.5 °/s, 0.5s minimum separation) were identified and categorized as a Yes or No based on temporal proximity (within 2 s) to a switch in talker location. Local time-courses (+/- 600 ms around the peak) for all head-movement events were used to train a two-class (Yes, or No) support-vector-machine classifier implemented in MATLAB. Two separate classifiers were trained: 1) with a leave-one-subject-out process (across subjects), and 2) with a leave-one-trial-out process (within subject).

Detection-rate as a function of jump size (change in talker location) was used to quantify the sensitivity of each classifier. Large jump sizes (greater than 60°) were detected at a high rate (exceeding 80%) by both classifiers, for all participant groups. Small jump sizes (less than or equal to 60°) were detected at a significantly higher rate by the within-subject trained classifier than the across subject classifier. Results indicate that large head-movements are relatively consistent across individual, but accurate detection of smaller head movements requires pre-defined subject-specific information.

D1-5. Model-Based Hearing-Enhancement Strategies for Cochlear Synaptopathy Pathologies

Fotios Drakopoulos*1, Viacheslav Vasilkov1, Alejandro Osses Vecchi1, Tijmen Wartenberg1, Sarah Verhulst1
1Ghent University

It is well known that ageing and noise exposure are important causes of sensorineural hearing loss, and can result in damage of the outer hair cells or other structures of the inner ear, including synaptic damage to the auditory nerve (AN), i.e., cochlear synaptopathy (CS). Despite the suspected high prevalence of CS among people with self-reported hearing difficulties but seemingly normal hearing, conventional hearing-aid algorithms do not compensate for the functional deficits associated with CS. Here, we present and evaluate a number of auditory signal-processing strategies designed to maximally restore AN coding for listeners with CS pathologies. We evaluated our algorithms in subjects with and without suspected age-related CS to assess whether physiological and behavioural markers associated with CS can be improved. Our data
show that after applying our algorithms, envelope-following responses and perceptual amplitude-modulation sensitivity were consistently enhanced in both young and older listeners. Speech-in-noise intelligibility showed small improvements after processing but mostly for young normal-hearing participants. Since our hearing-enhancement strategies were designed to optimally drive the AN fibres, they were able to improve temporal-envelope processing for listeners both with and without suspected CS. Our proposed algorithms can be rapidly executed and can thus extend the application range of current hearing aids and hearables, while leaving sound amplification unaffected. [Work supported by European Research Council ERC-StG-678120 (RobSpear)]

10:10 a.m. - 11:20 a.m. Coffee Break and …

**POSTER SESSION DP**

**DP101 In Situ Tuning of an Adaptive Feedback Canceler Using the Tympan Open Source Audio Processor**
*Joshua Alexander*¹, *Steven Neely*², *William Audette*³
¹Purdue University, ²Boys Town National Research Hospital, ³Creare LLC

Modern hearing aid design is critically dependent on feedback management, particularly when the coupling to the ear is wholly or partially unoccluded (i.e., ‘open’). To this end, adaptive feedback cancelers (AFCs) are necessary for managing dynamic feedback paths. In order to maximize the effectiveness of AFCs, multiple parameters need to be optimized using models or human testing. We use the Peak Height Insertion Gain (PHIG; Alexander et al. 2022, J. Acoust. Soc. Am. 150(3), 1635-1651) feedback criteria to optimize the parameters of a normalized filtered-x least mean square (NFXLMS) algorithm in situ on a KEMAR manikin using the open-source Tympan processing platform. The NFXLMS algorithm, implemented on the Tympan firmware, uses open-fitting earpieces. The initialized parameters of the NFXLMS algorithm are first determined by an optimization procedure that minimizes the estimation error between the modeled feedback path and the AFC estimated feedback path. In situ measurements of the resulting PHIG as a function of hearing aid gain for a typical mild sloping to moderate hearing loss are then used as input into an optimization algorithm that selects the next set of AFC parameters for tuning. Added stable gain (ASG) is quantified using the parameters that minimize PHIG as a function of gain. In this presentation, results for PHIG and ASG with the Tympan will be compared to published benchmarks from commercial hearing aids.

**DP102 The Tympan Open Source Audio Processing Platform**
¹Creare LLC, ²Boys Town National Research Hospital, ³University of Nebraska, ⁴tympan.org

Hearing-aid research is limited by the lack of wearable, reconfigurable and programmable audio processing platforms that can evaluate the real-time performance of new algorithms and innovative paradigms designed to help hearing impaired listeners in challenging conditions. The Tympan is an open-source hardware platform that utilizes a powerful Teensy 4.1 600MHz processor that can perform 16-band stereo compression in real-time yet can be programmed in the user-friendly Arduino environment. A mobile app (Android/IOS) gives researchers the ability to configure algorithms on the fly, store listener feedback, and record audio using a simple interface. Researchers can also customize the mobile app by defining the user interface right in the Tympan firmware, without any knowledge of mobile app software development. An open-source behind-the-ear (BTE) earpiece, with dual microphones, allows users to conduct human studies in realistic environments, where listeners can modify the parameters of their
hanging aid through the mobile app. During this special session, we will describe the Tympan platform design, report on computational performance measured in the laboratory and summarize the steps the team has taken to disseminate the platform and reach users in varied communities.

DP103 Determining Hearing-Aid Preset Configurations with Clustering Strategies
Chelzy Belitz\(^*\), Prof. John Hansen\(^1\)
\(^1\)University of Texas at Dallas

According to the results of a 2008 survey, of the nearly 35 million people in the United States who reported hearing difficulty, only 24.6% owned hearing aids \([1]\). Further, numerous surveys suggest that among the minority group who report both having hearing difficulty and own hearing aids, a percentage still opt not to wear it, with various international surveys suggesting as many as 4.2% to 24% of individuals who own a hearing aid electing not to use it. Numerous articles cite various reasons for non-use of hearing aids, including perceived value, fit/comfort, maintenance, healthcare professional attitudes, and financial reasons, among others \([2, 3, 4, 5]\). This study aims to suggest methods of increasing hearing aid accessibility and retention using machine learning clustering strategies as a means of obtaining effective device starting configurations. By potentially reducing time-to-convergence for best operation, providing potential settings for over-the-counter sound amplification, and streamlining the process of obtaining a hearing aid, it may be possible to further increase accessibility of hearing aids to a wider range of people and to minimize the reasons cited for non-use among the hearing aid-owning population. Using a massive corpus of over 90,000 pure-tone audiogram and the resulting hearing aid fitting data collected from over 500 hearing centers throughout the United States, this study suggests machine learning-based clustering protocols to determine a limited set of hearing aid initial configurations. Hearing aid fitting settings are clustered using distance-based algorithms. The resulting clusters are then used to suggest a set of optimal fittings which best represent each data cluster. These serve as the hearing aid preset configurations. Then, a classifier is used to determine top one or top two recommendations based on the result of the pure-tone audiogram, with a top-one accuracy of over 60% and top-two over 90% \([6]\). To limit the scope of the work, initial analysis is done using just four starting configurations, with later analysis suggesting 10 total clusters as an optimal number of clusters to give the most effective acoustic space coverage \([7]\).

DP104 Deep Neural Networks for Speaker Separation and Noise Reduction for Hearing Impaired Listeners
Lars Bramsløw\(^*\), Gaurav Naithani\(^2\), Tuomas Virtanen\(^2\)
\(^1\)Eriksholm Research Centre, \(^2\)Tampere University

Deep neural networks (DNN) have demonstrated substantial user benefits for speech-in-noise enhancement for hearing-impaired listeners and voice-on-voice enhancement. Recently, deep neural network algorithms have shown great potential in tasks like blind source separation of a single-channel (monaural) mixture of multiple voices. The idea is to train the algorithm on relatively short samples of clean speech, thus learning the characteristics of each voice. Once trained for those specific voices, the network can then be applied to mixtures of new speech samples from the same voices. The present work has applied several low-latency DNN architectures for 1) separating two known voices and 2) enhancing speech from more common noise types: a party noise and a shopping centre ambient noise. Speech from a 12-voice Danish HINT corpus was used for training, from which two male and two female voices were used in a HINT listening test, recording the word scores.

In experiment 1, two known voices were separated, using three different DNN algorithms. A 37%-point word score recognition in 15 hearing-impaired listeners was shown when selecting one voice and 13%-point in the same listeners when presenting the two separated voices dichotically to the two ears. These statistically significant benefits indicated a large potential for separating two voices in quiet.

In experiment 2, voice in noise was scored by 21 hearing-impaired listeners. Five different DNN architectures, employing both voice-dependent and voice-independent training. In party noise, a modest, but statistically significant, word score improvement of 16%-point was found for a voice-dependent and a voice-independent DNN type, while two other DNN types showed an improvement of 12%-point. In the more stationary shopping centre noise, no improvements were found. It was assumed that the more modulated party noise provided more glimpsing opportunities for the DNN algorithm, compared to the shopping centre noise.
**DP105 Influence of Number of Hearing Aid Compression Channels on Spatial Release from Masking**

Marc Brennan*, Sarah Garvey, Ava Feller

*University of Nebraska

The objective of this study was to understand how access to spectral information under conditions of hearing aid amplification can influence spatial release from masking (SRM). Previous studies have demonstrated that listeners with normal hearing take advantage of differences in the spectrum of speech and noise, dips in the noise level, and spatial cues that can occur between sound sources to improve their understanding of speech. Listeners with sensorineural hearing loss are less able to take advantage of some of these spatial cues. Possibly by not considering the influence of access to spectral information under conditions of amplification, hearing aids provide—at best—marginal improvement in SRM. This experiment examined whether modifying the number of compression channels can improve SRM. The experimental conditions were based on prior data that documented improved measures of spectral resolution, estimated with psychophysical tuning curves, with 16-relative to both an unaided condition and to 4-channels of compression. For this study, we obtained sentence recognition for AzBio sentences presented in front of each participant. Simultaneous masking from either a male or female speaking the rainbow passage were presented in both collocated and spatially separated (masker at 30 degrees) conditions. The maskers were fixed at 50- or 70-dB SPL while adaptively varying the level of the sentences. Test conditions included unaided and four and 16 channels of compression with the Tympan open-source hearing aid platform. We hypothesized that SRM would be greatest for the 16-channel compression condition, followed by the 4-channel compression condition, and poorest for the unaided condition. While data collection is ongoing, preliminary data for 13 subjects with sensorineural hearing loss suggests a similar SRM between the unaided and 4 channels of compression conditions. SRM for these two conditions was similar to 9 subjects with NH (unaided). Inconsistent with the hypothesis, SRM was poorest for the 16-channel compression condition, indicating that changes in access to spectral cues across conditions was not the primary contributing factor to sentence recognition. Potential alternative factors will be discussed, including the presentation of supporting evidence from electroacoustic analyses of interaural level and timing differences, audibility (speech intelligibility index), better ear advantage, and binaural unmasking. Initial results suggest that reductions in interaural level differences may have contributed to poorer SRM for the 16-relative to 4-channels of compression. These results provide proof of concept for the Tympan as a research platform.

**DP106 Preliminary Evaluation of Fall Risk Assessment Methods Using Ear-Wearable Devices**

Justin Burwinkel*, Matan Sivan, Roy Rozenman, Archelle Georgiou

*Starkey

Hearing aids are among the few wearable devices that can be worn, comfortably and unobtrusively, for extended periods of time. Interestingly, individuals seeking treatment from hearing healthcare clinics were found to have a greater risk of falling than their age-matched peers (Críter and Honaker, 2016). Lin and Ferrucci (2012) also reported an independent association between the severity of hearing impairment and reports of falls, even when adjusting for demographic, cardiovascular, and vestibular function.

In recent years, motion sensors have been successfully embedded into commercially-available hearing aids for the purposes of tracking the wearer’s daily physical activity and to automatically send notifications to caregivers when a fall event is detected (Burwinkel et al., 2020). We postulate that these advanced ear-wearable devices may also be suitable for chronically monitoring the wearer’s postural stability during structured and unstructured activities. For example, wearable motion sensor data, obtained during instrumented falls risk screening tasks, has previously been used to detect significant differences between fallers and non-fallers (Howcroft, 2016; Zakaria et al., 2015). Evidence also suggests that daily physical activity level and the experience of near falls are significantly associated with incidence of future falls (Inouye et al., 2007; Nagai et al., 2017).

The authors will present preliminary findings demonstrating the technological feasibility of using machine learning and hearing aids with embedded motion sensors to monitor a wearer’s physical activity (n=20), postural stability (n=10), and assess future risk for falls (n=71).

**DP107 Withdrawn by author**
The use of auditory models is important for designing speech and audio processing algorithms for hearing assistive devices. These auditory models are often parameterized by a set of parameters relating to auditory function, e.g., hair cell loss or synaptopathy. In practice, the computational load of these auditory models can be very high thus limiting the feasibility of using the models as bio-inspired loss functions for deep learning based hearing loss compensation strategies or denoising strategies. Previous efforts have addressed this problem by training a neural network [1] for each parameter configuration of the auditory model which greatly reduces the computation time of the auditory model and allows for direct and efficient computation of the gradient when using the auditory model as a loss function but requires a new network to be trained whenever the parameterization changes. We propose an approach where a single neural network is trained, once and for all, to accurately simulate auditory models across their parameter spaces by conditioning the weights of the network on the parameters of the respective auditory models. This approach enables greater flexibility than training a single model for each parameter configuration, as any parameterization can be acquired on the fly. We showcase the approach on two different auditory models, the filterbank model of the cochlea and inner hair cells by Zilany et al [2], and the transmission line model of the cochlea by Verhulst et al [3]. The accuracy of the neural network is shown to be robust across both unseen inputs and different hearing losses.

References:

DP109 Spatially-Selective Spectro-Temporal Post-Filtering via Short-Time Target Cancellation for Enhancement of Hearing Aid Processing
Marcos Cantu*, 1 Volker Hohmann1
1 Carl von Ossietzky University of Oldenburg

The approach to spatially selective (i.e., directional) speech enhancement described and evaluated in this contribution involves combining adaptive beamforming and real-time capable spectro-temporal post-filtering (i.e., time-frequency masking). The well-established “Adaptive Differential Microphone” (ADM) hearing aid algorithm is used for the beamforming, while our causal, efficient and memoryless “Short-Time Target Cancellation” (STTC) processing is used for the spatially selective spectro-temporal post-filtering. ADM was chosen as an exemplar hearing aid algorithm because it is a state-of-the-art directional speech enhancement algorithm, has been found to provide exceptional speech quality, and is widely used in commercially available digital hearing aids. Our STTC processing can be used either to filter the binaural signals at the ears (i.e., the “STTC” condition) or as a post-filter (i.e., the “STTC+ADM” condition) for the ADM adaptive beamforming. Evaluation results, using both instrumental measures and listening experiments, indicate an additive effect, with overall better performance for the combination of the two (i.e., “STTC+ADM”) than for either processing scheme by itself. The two signal processing approaches are additive and compatible in part because they are both spatially selective; i.e., they each provide directional speech enhancement. Our STTC processing consists of a minimalist motif that uses Short-Time Fourier Transforms (STFTs) from any given pair of microphones to compute a continuously-valued time-varying spectral gain (i.e., ratio mask). Thus far we have implemented and evaluated our STTC processing with both a small (11 mm) dual-microphone endfire array, as in a typical earworn hearing aid, and with a larger (120 mm) broadside array of four eyeglass-integrated microphones. Evaluations with mixtures of multiple (2,3,5 and 7) concurrent talkers, in both a semi-anechoic chamber and a reverberant classroom setting, indicate that the endfire array implementation has better attenuation of sound sources from the rear hemisphere whereas the broadside array implementation has better attenuation of sound sources to the sides. Our “STTC+ADM” processing has been implemented in real-time, with low (<20 ms) latency, using the open Master Hearing Aid (openMHA) real-time signal processing platform. In future work, we plan to carry out listening studies, using these real-time implementations via openMHA, in complex, dynamic and ecologically valid acoustic scenarios. We will also combine our STTC processing with alternative, and arguably more effective and sophisticated, beamforming algorithms for hearing aids. The approach taken here, computing a spatially-selective ratio mask via real-time processing,
may be used to enhance the performance of hearing aid algorithms in especially challenging listening situations.

**DP110 Cnn-Based Comparative Framework for Non-Linguistic Sound Classification and Enhancement in Normal Hearing, Hearing Aid and Cochlear Implant Conditions**

*Ram Charan Chandra Shekar*¹, *Prof. John Hansen*¹
¹University of Texas at Dallas

Most Hearing Aid (HA) research efforts are focused on improving speech perception and less on the acoustic environmental context. According to the World Health Organization, approximately 466 million people worldwide suffer from hearing loss, and only 17% use HA on a regular basis [1, 2]. Non-linguistic sounds (NLS) generally refer to a wider range of soundscape representations of the environmental context of sound (e.g., music, animal sounds, sounds in nature – wind, etc). NLS perception enables autonomy, environmental awareness, and subject comfort. Unequivocal NLS identification plays a pivotal role in engaging the listener to respond to safety/threat scenarios, environmental context awareness, and plays an important role in improving hearing related quality-of-life. Advancements and improvements in speech perception cannot be directly used to infer improvement in NLS perception since NLS activates a distinctly different portion of the brain versus speech (Wernicke’s Area) and cannot be easily described using phonemes or similar low-level representations. In our previous effort, a CNN-based (Convolutional Neural Network) model is considered for comparative analysis of NH vs CI-simulated audio classification performance [3]. In our current study, a similar effort is explored to compare NLS classification in normal hearing (NH) vs HA/CI processed sounds. Here, HA processed audio are used to extract sound representations from a pretrained CNN in HA condition. Next, these HA processed sound representations are used to train SVM in HA processed condition. The classification performance of HA processed sounds are comparatively assessed versus the CI-simulated and NH conditions. An NLS enhancement algorithm was proposed in [4] based on that study finding that suggested NLS identification and perception is correlated with spectro-temporal properties, namely: fractal dimensions, harmonicity, mean peak and mean spectral centroid factors among NH listeners [5]. Here, a proposed NLS enhancement algorithm is formulated that focuses on preserving spectro-temporal characteristics that are important for NLS identification and perception in CI conditions, where performance of NLS enhanced classification was evaluated using CNN-based framework. Here, the NLS enhancement algorithm is applied to estimate the optimal filter-gains that preserve perceptually important spectro-temporal characteristics in HA processing. Next, NLS enhanced classification is evaluated using the CNN-based comparative framework to evaluate NH, HAs and CI classification. The proposed comparative NLS classification framework for NH, HAs and CI conditions contribute towards: (i) advancement of NLS recognition studies in assistive hearing devices, (ii) development of a community shared testbed for comparative NLS studies, and (iii) NLS enhancement studies among HA/CI listeners.

**DP111 Group Conversation Enhancement Using Wireless Microphones and the Tympan Open-Source Hearing Platform**

*Ryan Corey*¹, *Andrew Singer*¹
¹University of Illinois Urbana-Champaign

Some of the most challenging listening environments, especially for people with hearing loss, are group conversations in crowded spaces, such as restaurants. Conventional hearing aids perform poorly in noisy environments because their microphones capture a mixture of speech and unwanted background noise. Remote microphones can improve intelligibility in noise by transmitting speech directly from a talker to the ears of the listener. However, commercial remote microphones are unsuitable for group conversations because they work with only one talker at a time and do not preserve spatial cues such as interaural time and level differences. These spatial cues are especially important for group conversations because they help the auditory system to follow speech from multiple talkers.

We have recently proposed an immersive multitalker remote microphone system that combines the low noise of remote microphones with the realistic spatial cues of earpiece microphones [1]. A set of adaptive filters processes the remote microphone signals to match the magnitude and phase of the earpiece signals. Because it relies on the earpieces as references, the system does not need to explicitly localize or track the talkers and the enhanced remote signals can be seamlessly mixed with the live sound at the earpieces. A basic version of the system was implemented in real time on the Tympan open-source hearing aid development platform [2].
In this presentation, we extend the previous implementation to improve performance in real-world environments. In particular, we consider robustness against talker and listener motion, crosstalk between two or more talkers, delayed auditory feedback of the listener’s own speech, and double-talk. We also demonstrate a cooperative conversation enhancement system for multiple users with listening devices. We compare experimental results from laboratory equipment with the real-time implementation on the Tympan hardware platform.

This work was supported in part by an appointment to the Intelligence Community Postdoctoral Research Fellowship Program at the University of Illinois Urbana-Champaign, administered by Oak Ridge Institute for Science and Education through an inter-agency agreement between the U.S. Department of Energy and the Office of the Director of National Intelligence.

References:

**DP112 A Neural-Network Framework for the Design of Individualised Hearing-Loss Compensation**
Fotios Drakopoulos*, Sarah Verhulst

Even though the human auditory system is known to be complex and highly non-linear, hearing aids (HAs) still rely on simplified descriptions of the auditory system or on sensorineural hearing loss (SNHL) estimations, such as hearing thresholds or perceived loudness, to yield optimal acoustic amplification to HA users. Standard amplification strategies succeed in restoring inaudibility of faint sounds, but sometimes fall short of providing precise treatment outcomes for complex sensorineural deficits such as presbycusis or cochlear synaptopathy (CS). To address this challenge, we adopt a deep-neural-network (DNN) version of a biophysically realistic model of human auditory processing (CoNNear). CoNNear accurately simulates individual SNHL and comprises a fully differentiable description that can be used to design individualised HA strategies from the ground up. DNN-based audio-processing models (DNN-HA) can be trained that can optimally process sound to restore hearing in impaired auditory peripheries. In this study, we evaluate the restoration capabilities of our framework using simulated auditory-nerve (AN) responses of normal and impaired auditory peripheries. We compare different loss functions designed to optimally compensate for a mixed outer-hair-cell loss and CS impairment, each time focussing on the enhancement of different auditory features. After evaluating which trained DNN-HA model yields the best restoration outcomes on simulated AN responses and speech intelligibility, we applied the same training procedure to two milder hearing loss profiles, separately for OHC loss and CS. Our results show that a simulated restoration of AN population responses was possible in all cases, with OHC loss proving easier to compensate than CS. Several objective metrics were considered from the literature to estimate perceptual benefits after processing, with the results holding promise for improved understanding of speech-in-noise processing for hearing-impaired listeners. Since our framework can be tuned to the hearing-loss profiles of individual listeners, we enter an era where truly individualised and DNN-based restoration strategies can be developed and be tested experimentally. [Work supported by European Research Council ERC-StG-678120 (RobSpear) and FWO grant G063821N Machine Hearing 2.0.]

**DP113 Exploiting an External Microphone for Binaural Direction of Arrival Estimation for Multiple Speakers**
Daniel Fejgin*, Simon Doclo

In speech communication applications such as hearing aids, accurately estimating the direction of arrival (DOA) of competing speakers is of crucial importance. To achieve robust DOA estimation of multiple speakers in noisy and reverberant environments, several learning- as well as non-learning-based methods have been proposed, which make use of, e.g., interaural time and level differences, generalized cross correlation functions or relative transfer functions (RTFs). Recently, a low-complexity RTF vector estimation method was proposed in [1], which exploits the availability of an external microphone and assumes that the spatial coherence between the noise components in the external microphone signal and the hearing aid microphone signals is low. Based on this RTF vector estimation method, in [2] we proposed a binaural DOA estimation method for a single speaker by selecting the direction for which the Hermitian angle between the estimated RTF vector and a database...
of prototype anechoic RTF vectors is minimized. In this contribution we extend the DOA estimation method from [2] to the multi-speaker case. Instead of averaging the Hermitian angle over all frequencies, we consider only a subset of frequencies, where it is likely that one speaker dominates over all other speakers, noise, and reverberation. The DOAs of the speakers are then estimated by determining the peaks of the Hermitian angle spectrum, assuming the number of speakers to be known. We compare the effectiveness of coherence-based quantities for frequency bin subset selection, more in particular the generalized magnitude squared coherence and two binaural coherence-to-diffuse ratio estimators from [3]. Using recordings of speech and diffuse-like babble noise in acoustic environments with mild to severe reverberation and different signal-to-noise ratios, we demonstrate the performance of the proposed binaural DOA estimation method for scenarios with two competing speakers. [This work was funded by the Deutsche Forschungsgemeinschaft (DFG, German Research Foundation) under Germany’s Excellence Strategy - EXC 2177/1 - Project ID 390895286 and Project ID 352015383 - SFB 1330 B2.]

References:

DP114 Binaural Mvdr-Ic Beamformer for Real-Time Hearable Device
Yoh-ichi Fujiyama1, Nobuhiko Hiruma2, Yoshitaka Murayama3
1RION Co., Ltd., 2Rion Co., Ltd, 3Cear, Inc.

Binaural hearing plays an important role in the recognition and understanding of spoken language in a diffuse noise field with various interferences and background noises. The advantage of binaural hearing is particularly the ability to distinguish angular differences in sound information coming from different directions. Meanwhile, it is known that this angular difference information becomes ambiguous with hearing loss, i.e., the minimum audible angle is wider than normal hearing (Häusler et al., 1983). MVDR beamformer (Capon, 1969), which extracts the desired speech without distortion and preserves the direction of arrival information, has been used as an effective method of the purpose for SNR improvement and binaural cue preservation with spatial awareness. However, interference can also give the sense that they are coming from the same direction, which may impair comprehension of the desired speech.

To solve this problem, Marquardt et al. (2018) proposed MVDR-IC and MVDR-MSC considering interaural and magnitude squared coherence.

However, it has difficulty in terms of computational cost and processing latency when considering its use for hearing aids, which require a low processing latency, because the MVDR beamformer requires to compute speech and noise correlation matrices on a frame-by-frame basis. We evaluated the effectiveness of MVDR-IC method on a general-purpose SoC by applying a frequency-domain preconfigured filter (Hiruma et al., 2022), which is based on the situation with desired speech from front-right or front-left, into side-branch type of signal flow in order to realize the low latency in true real-time processing.

Objective Evaluation: The effectiveness of the embedded MVDR-IC feature was verified on an actual device in several SNR situations with background noise and different DOA of interference, and IC trade-off settings. PEASS (Emiya et al., 2011 and Vincent et al., 2012), which is objective measurement for the evaluation of audio source separation that correlates better with human assessments than general ratio measurements, was used as perceptual metric. It was found that spatial separation of interference and desired speech with minimal distortion is possible by selecting appropriate IC tradeoff parameters.

Subjective Evaluation: The spatial separation and awareness between desired speech and interference were investigated for eight normal hearings. It was found that using the embedded real-time MVDR-IC contributes to improving the recognition of desired speech without loss of spatial impression.

From these evaluations, the MVDR-IC beamformer with preconfigured filter is applicable not only to hearing aids but also to real-time speech listening systems.

DP115 Tympan Electroacoustic and Behavioral Measurements
An audio processing platform called Tympan that allows implementation of custom algorithms directly on wearable hardware and modification of parameters via a simple user interface was developed as part of the NIH-NIDCD efforts to support innovation in hearing aid research through the development of open-source audio processing platforms. Recent revisions of the Tympan included the addition of wearable hardware with receivers and directional microphones at the ear-level. Tympan performance was compared to a commercial hearing aid with similar features. Hearing aid output of both devices was obtained following the procedures defined in the ANSI standard and Verifit 2 manual and met manufacturer specifications.

14 adults with mild-moderate sensorineural hearing loss participated in behavioral testing (7 female; age range 34-87 years). Gain was set according to the NAL-NL1 prescriptive method for both devices for each participant using the AudioScan Verifit 2. There was a good match between fit and target for both devices, except at 2 kHz where the output of the Tympan was higher than the prescriptive target.

Two lists of 10 CASPA consonant-vowel-consonant words (Mackersie et al. 2001) were presented in quiet at three input levels (55, 65, 75 dB SPL). Percentage correct improved from 55 to 65 dB SPL and was better aided than unaided at 55 dB SPL. AZBio sentences (Spahr et al. 2012) were presented at 70 dB SPL in 10-talker babble at 5 dB signal-to-noise ratio. Word recognition was better aided than unaided. The Tympan performed similarly to the commercial hearing aid in all conditions. Electroacoustic measurements included evaluation of the hearing aid speech perception index and speech quality index (HASPI and HASQI).

These results are consistent with results from a previous study comparing an earlier version of the Tympan to a different commercial hearing aid. Though the previous behavioral results implied that the two hearing aids performed equally, the qualitative impressions of the Tympan were poorer than the commercial hearing aid overall. In the current study, a sound quality questionnaire was included to quantify perceived characteristics of the devices. Several patterns were consistent in responses to all questions: (1) ratings increased as the difficulty of the test condition decreased, (2) ratings for difficult conditions were higher for aided compared to unaided conditions, (3) participants rated the Tympan similarly to the commercial hearing aid. The responses on the questionnaire provide evidence that, on average, participants thought the Tympan was similar to the commercial hearing aid.

**DP116 Hearing-aid research with the open Master Hearing Aid (openMHA)**

*Hendrik Kayser*, **Chaslav Pavlovic**, **Volker Hohmann**

1. **Carl von Ossietzky University of Oldenburg**, 2. **BatAndCat Sound Labs**

The project "Open community platform for hearing aid algorithm research" (NIDCD R01DC015429) provides a software platform for real-time, low-latency audio signal processing: open Master Hearing Aid (openMHA). It contains a versatile set of basic and advanced methods for hearing aid processing, as well as tools and manuals enabling the design of own setups for algorithm development and evaluation. Documentation is provided for different user levels, in particular for research audiologists, application engineers and algorithm designers. The software runs on standard off-the-shelf hardware including lab setups and portable setups. Being of particular interest for the evaluation of new methods in real-word scenarios, a portable, integrated research platform for openMHA is provided in conjunction with the SBIR project R44DC016247: the Portable Hearing Laboratory (PHL). This contribution introduces openMHA and discusses current use cases and possible application scenarios of the software framework as well as the PHL. The first steps of using the software will be demonstrated including the setup of a basic configuration, manipulation of processing parameters at runtime - locally or from a remote devices such as other computers and smartphones. An overview over the tools, provided with the software package, that address the different user groups will also be given. The aim is to learn about the flexible configuration and remote control of openMHA running a typical hearing aid processing chain, to get an insight into basic steps required to build your own hearing aid research setup, and to gain an overview of application possibilities of openMHA in hearing research.

**DP117 Artificial Neural Network Applications in Determining Critical Factors Related to Speech Recognition in Noise**
Listeners with normal hearing and hearing loss) often find it difficult to perceptually segregate a single target voice from its competition and this has been termed “the cocktail-party effect” (Cherry, 1953). The still largely-unsolved cocktail party problem faced by listeners in noisy environments is typically confounded by four factors: 1) spatially separation of the target from its competition, 2) differences in temporal and spectral properties between target and competing speech, 3) energy differences between target speech and competing noise, termed “energetic masking”, and 4) similarities or differences in linguistic content between target speech and competing speech, termed “informational masking”.

The purpose of the proposed study was to study the intelligibility of QuickSIN sentences under varying azimuth and SNR conditions. This study consisted of fifteen listeners from a younger age range (19-40 years), with normal otoscopy and middle ear function, normal hearing on audiogram, and 4) normal localization ability. Participants were administered the QuickSIN test in a sound field environment with target speech (sentences) presented at 0 degrees azimuth (front loudspeaker) versus babble (noise) presented from three different locations (0 degrees or in front of listener, 90 degrees or to right of listener, and 180 degrees or directly behind the listener). Sentences recorded at varying SNRs (+5, 0, and -5) were used for presentation via a calibrated audiometer. The QuickSIN test was chosen for this study because it is a standardized, validated tool to measure speech recognition in noise.

Statistical analyses were completed on speech intelligibility data obtained under the nine conditions (3 SNRs X 3 noise locations). The between-subjects factors were SNR and noise location and results indicated statistically significant (p<0.05) effects of SNR but no significant effects of location or interaction of location*SNR. Results of Artificial Neural Network analyses used two main predictors (SNR, location) in the input layer that were linked to an output layer (speech intelligibility), linked via a single hidden layer comprised of four units. For final estimation by the output layer, only three of the hidden nodes showed strong synaptic weight.

Results suggest that even when the target speech and masker originate from different directions at unfavorable SNRs, this spatial separation advantage is lost. Energetic masking has greater influence than noise location on speech intelligibility in noise.

Based on study results, hearing aid algorithms that focus on improving speech intelligibility in noise show greater promise than those algorithms that focus on spatial recognition of sources.
on their configuration and processing arrangement. A stronger NR system can provide a higher output SNR and a low level of residual noise compared to a more conservative NR system. However, it might attenuate soft speech components to a point where they cannot be amplified by dynamic range compression. A fast-acting compression system can increase audibility compared to a slow-acting compression system but might amplify the residual noise after NR. A serial arrangement can provide an increased amount of compression but also amplifies more residual noise compared to a parallel arrangement. This study investigated the influence of these choices by evaluating a large set of systems using noise reduction methods (including ideal, model-based, and convolutional neural networks) combined with either fast-acting, slow-acting, or adaptive compression settings in either parallel or serial arrangements. The systems were tested with noisy speech and evaluated using objective metrics (e.g. the effective compression ratio and the change in SNR). Each system was compared to a reference system that used ideal ratio-mask for noise reduction combined with fast-acting compression in a serial arrangement. The reference system was considered to provide the highest amount of both noise reduction and compression. Each system was compared to the reference system in terms of the similarity of their objective metrics. This was done by calculating the likelihood that the objective metrics from each system came from the same distribution as those of the reference system. The results showed that the NR stage had the largest effect in terms of similarity to the reference system, followed by WDRC and the processing arrangement. In addition, the results suggested that a sufficiently effective NR method was more similar to the reference system when fast-acting compression was used in a serial arrangement, while the model-based NR method was most similar to the reference system when adaptive compression was used.

DP120 Hearing Aid Research with the Portable Hearing Laboratory (PHL)

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We report here on the development of a portable hearing aid research platform that has been accomplished in the SBIR Phase II Project R44DC016247; as well as on the early development and plans for its Phase III continuation (currently completing its Year 1), which will develop the successor generation of the current platform with several extensions.

The project builds on the openMHA software (its distribution for ARM processors) included in an optimized Linux system developed in project R01DC015429 to provide a complete portable and wearable software-hardware master hearing aid device needed for development of new innovative solutions for assisted hearing. We refer to this device as the PHL for Portable Hearing Laboratory. The PHL consists of a ARM Cortex A8 based processing unit (open-source single-board computer BeagleBone Black) and a codec set able to support hearing aid architecture of up to 6 microphones and 4 speakers, a further development of the open-source audio board “Cape4all”. It is accompanied by a binaural 4-microphone extremely low noise BTE hearing aid set. Additionally, it features stereo line in and line out connections and also interfaces to different headset form factors. In particular, it supports the ITE stereo headset developed at the University of Oldenburg (“Hearpiece”). The device can be remotely controlled with a smartphone and computers.

Phase III work on the PHL, greatly builds on the experience and results of a large number of studies by various laboratories across the world done on the current Phase II version of the PHL device. We will present the most important conclusions reached in these studies with respect to the PHL capability and performance. We will then focus on the current electroacoustic and software capabilities of both Phase II and Phase III device and the type of studies it already makes possible and will further enable.

DP121 Automatic analysis tools for exploring own-voice and near-ear sound pressure level distributions

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Exploring acoustic conditions and listening experiences of people in their everyday environments has drawn a lot of attention. Our laboratory developed the smartphone-based ecological momentary assessment (EMA) system oMEGA which stores smoothed acoustical features (root mean square, RMS; auto and cross-power spectral density, PSD; zero crossing rate, ZCR) on a reduced time scale to preserve the privacy of the test participants. The recordings were per-
formed by two head-worn, near-ear microphones attached to glasses. Simultaneous to the recorded acoustical features, subjective assessments are taken in situ by surveys on the smartphone.

This contribution shows analysis results of 13 near-ear recordings of hearing-impaired participants (aged 50 to 75 years) in an EMA study. The collected acoustical features and subjective assessments of approx. 4 days per participant were extracted from the smartphones and transmitted to a database server for further analysis. The database server facilitates data management and enables international collaboration. Furthermore, new features can be extracted automatically that are of interest for studying the impact of natural, acoustic environments on the communication abilities of elderly people. For this contribution, an algorithm detecting the test participant’s own voice (Own Voice Detection, OVD) and an algorithm that extracts the sound pressure level (SPL) in order to measure the sound exposure were applied to the EMA data. The OVD is based on a machine learning algorithm and a set of acoustical features derived from the oMEGA features. The evaluation of the OVD with a manually labeled real-world recording of one full day showed reliable and robust detection results.

The analysis of the EMA data shows that the grand mean percentage of predicted own-voice audio segments (OVS) during one day was approx. 10% which corresponds well to other published data. The grand median SPL across all participants and recording days was 50.3 dB(A). The OVS had a small impact on the median SPL over all data. However, for short analysis intervals, significant differences of up to 30 dB occurred in the measured SPL, depending on the proportion of OVS and the SPL of the background noise.

DP122 Robust Optimization Methods for Fixed Virtual Sensing Feedback Active Noise Cancelling Controllers Targeting In-Ear Headphones
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Active Noise Cancellation (ANC) applied to headphones aims at minimizing the environmental noise at the listener’s ears. In order to achieve this goal, the ANC controller generates a sound wave through the loudspeaker of the headphone to cancel the sound wave of the environmental noise that leaks into the ear canal. In this contribution, we consider an in-ear headphone equipped with one inner microphone and one loudspeaker, and we use two virtual sensing approaches, namely the remote microphone technique (RMT) [1] and the virtual microphone arrangement (VMA) [2] to derive a fixed feedback ANC controller that minimizes the sound pressure at the ear drum instead of at the inner microphone. Moreover, based on multiple measurement sets of the acoustic paths between the loudspeaker, the noise source and the ear drum, we derive design and stability constraints that account for variability in the acoustic paths, e.g., produced by inter-subject variability. Simulation results for diffuse noise and considering multiple reinsertions of the in-ear headphone show an improvement of the noise attenuation at the ear drum by using the virtual sensing approaches and a better control of the amplifications by using the RMT approach, when compared to a classical approach optimized at the inner microphone [3]. [This work was funded by the Deutsche Forschungsgemeinschaft (DFG, German Research Foundation) under Germany’s Excellence Strategy - EXC 2177/1 - Project ID 390895286 and Project ID 352015383 - SFB 1330 C1.]

References:

DP123 Adaptive Speech-In-Noise Enhancement in Hearing Aids with Deep-Learning-Based Noise Reduction
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Understanding conversations in complex environments remains a very commonly reported difficulty for hearing-aid (HA) users. Modern HA technology typically addresses this issue by enhancing the signal-to-noise ratio (SNR) between the HA input and output, using noise-reduction (NR) techniques consisting of a combination of advanced beamforming (BF) and postfiltering (PF) algorithms. BF algorithms have traditionally been driving the clinical benefits provided
by HA NR, while PF algorithms have shown limited measurable benefits. This study investigated whether using a PF technique based on deep-neural-network (DNN) processing could improve performance in terms of SNR enhancement and speech intelligibility in noise. Another aim was to map the pattern and range of SNR and short-term objective intelligibility (STOI) improvements achievable by an adaptive NR system combining BF and DNN-based PF strategies when using either compressive or linear amplification. The Hagerman and Olofsson phase-inversion technique was applied in an ecologically-valid speech-in-noise environment to calculate the HA output SNR for input SNRs ranging from -10 to +20 dB. The results showed that the DNN-based PF could provide a relatively stable SNR enhancement of about 2 dB across all input SNRs on top of the enhancement provided by the BF, largely outperforming the traditional PF approach. In a listening test with 20 HA users, the DNN-based PF approach was also found to increase speech intelligibility in diffuse noise significantly. A wide range of SNR enhancement patterns were obtained depending on the chosen amount of BF and PF activation, the combination of both algorithms providing up to 10 dB SNR enhancement at the most adverse input SNRs. The STOI estimates confirmed an improvement of speech cues in noise when activating NR. In line with existing literature, applying compression slightly decreased the output SNR at positive input SNRs. These findings indicate that DNN-based PF approaches can contribute substantially to the SNR-enhancement benefits of advanced HA NR systems. They also demonstrate a wide range of possibilities for adjusting NR settings in premium HA technology, enabling fine-grained individualization in terms of the amount of SNR enhancement desired for each user. However, determining the optimal amount of help in noise for a given user remains a clinical challenge. An ongoing study aiming towards that goal measured speech intelligibility in noise for identical combinations of HA settings and the same test setup as used here. The relationship between their findings and the present results will be discussed on another poster by Jürgens et al.

References:

DP125 Designing the Real-Time Master Hearing Aid (RT-MHA) Framework for the Open Speech Platform (OSP)
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Background: The open-source audio processing tools for hearing aid (HA) research community came to a consensus that the best hardware platform for the next generation of HA research tools should be based on mobile computing platforms. These platforms include single board computers (SBC) like the Raspberry Pi, Qualcomm 410c, the Beaglebone, etc. The reasoning...
behind choosing these SBC platforms is that they are the ideal combination between the available computation, energy efficiency, and programmability. These SBC platforms achieve the ideal combination by using multiple super-efficient CPU cores, usually two or more, among other processing elements, making these computing platforms, unlike traditional computers. Therefore, it is essential to design the research tools with the hardware in mind.

**Approach:** This work makes two critical contributions to this area. The first contribution is a detailed analysis of the best way to set up the operating environment for these SBC platforms to get the most out of the hardware, no matter the open-source HA framework used. Our analysis found that we achieve the best performance when partitioning the computation resources between real-time (RT) and non-RT applications. Using this essential information and others gathered from the analysis, we designed the RT master hearing aid (RT-MHA) framework, the second critical contribution of this work. RT-MHA is a part of the Open Speech Platform (OSP), which has been designed and optimized to utilize the resources available on the SBC best. The RT-MHA framework gives HA algorithm researchers the most usable computational resources while minimally impacting the performance of non-RT applications like the embedded web server on the OSP platform.

**Results:** We show the different impact mechanisms on a modern-day SBC has on the real-time performance of the MHA algorithm. These mechanisms include the scheduler on Linux, the partitioning of computing resources, and the idling mechanism on SBC. Then we will describe how the RT-MHA framework can optimize the workload given what we found in our analysis to get up to 2 times more computation while being able to guarantee real-time operation.

**DP126 Live OScope: A Debugging Tool for Master Hearing Aid (MHA) Algorithms for the Open Speech Platform (OSP)**

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**Background:** Developing and implementing signal processing and master hearing aid (MHA) algorithms on open-source audio processing research tools come with challenges, especially when deployed on an embedded system. It is challenging because we have little to no visibility in most open-source audio processing research tools once the algorithms are deployed in an embedded system. Our experience found that the most challenging part is understanding why an algorithm does not work when deployed even though it was tested thoroughly in a simulated environment. Therefore, we feel these research tools need an intuitive interface for debugging these algorithms. For that reason, we extended the Open Speech Platform (OSP) framework to include a debugging tool called the Live OScope.

**Approach:** We were inspired by the oscilloscope, an analog tool used to debug signals, when we designed the Live OScope feature of the OSP framework. The Live OScope allows algorithm developers to insert test points in their algorithms for monitoring during run time. The Live OScope feature has two components: the recorder and the display. The recorder is a part of the OSP framework; it continuously records the data from the test points into a circular buffer of a predefined length, usually 1-3 seconds. The recording happens until the display component asks for the data. At which time the data gets streamed from the buffer in its entirety to the display component. After the streaming has completed, the recorder component starts sampling the data again. The display component, an external tool running on a remote device, displays the waveforms captured on the embedded device. Like an oscilloscope, the display component can refresh at a set interval and save the waveform captures.

**Results:** We first show that the Live OScope feature has minimal impact on the real-time performance of the embedded system. Next, we will demonstrate the utilities of this tool through three case studies that heavily utilized this tool, including how this tool is currently used to calibrate the microphones of these embedded devices. Finally, we will have a live demonstration of the Live OScope tool.

**DP127 Realtime Multirate Multiband Amplification for Hearing Aids**

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**Background:** Subband amplification is a basic feature of modern hearing aids (HA) to compensate for individual hearing loss. However, the temporal and spectral nature of hearing loss poses many challenges for the HA signal processing designers. Larger number of frequency channels offers more precision and
accuracy for fulfilling HA prescriptions, especially for unusual hearing loss patterns, but increases the complexity and reduces battery life. There is also a lack of understanding on how to accurately satisfy attack and release time parameters in a real-time system based on ANSI 3.22 guidelines, to facilitate systematic investigation of these parameters on listener outcomes.

**Approach:** We present a novel multiband real-time amplification system for HAs using multirate signal processing to minimize the complexity and power consumption of processing multichannel audio. The system offers precise control of the temporal dynamics of WDRC. The filterbank is based on audiometric frequencies comprising eleven half-octave frequency channels spanning five octaves, from 250 Hz to 8000 Hz, uniformly distributed on the logarithmic scale, with each band having a proportionate bandwidth, mimicking the known properties of cochlear transduction. Further, each octave of audio channels is mapped to a different sampling rate, such that each group of bands is processed at the lowest possible sampling rate. Processing lower sub-bands at the original Nyquist rate results in redundant samples. The proposed multirate system removes this redundancy and offers dramatic reduction in complexity. We implemented a new WDRC subsystem using an automatic gain control (AGC) using a Hilbert Transform for the envelope estimation in each band in the lower sampling domain. This approach accurately satisfies attack and release time specifications for the compression parameters as suggested from ANSI 3.22 guidelines.

**Results:** We implemented the proposed multirate HA algorithm on the Open Speech Platform (OSP) – an open-source system hearing loss research. On-target measurements on the OSP hardware show that the system runs in real-time, offers better frequency resolution, and with less complexity than prior sub-band amplification tools available in OSP. The proposed multirate filter bank provides a 14x reduction in complexity compared to a single-rate implementation, and has an algorithmic latency of 5.4 ms – well within the conventional threshold for real-time operation. The WDRC subsystem satisfies ANSI 3.22 guidelines within 0.5 ms. We present acoustic measurements (e.g., using Verifit Verification Toolbox, HASQI) to confirm the accurate static, and dynamic behavior of the proposed system for the ISMADHA hearing loss profiles.

**DP128 Hybrid speech enhancement for hearing aids with deep neural networks and MMSE-LSA**

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People using hearing aids face some problems in their daily lives owing to the ineffectiveness of hearing aids in noisy environments. Many hearing aid manufacturers have struggled with this problem. For example, classical speech enhancement (SE) systems represented by the spectral subtraction method and Wiener filtering are less complex but have drawbacks such as suppression of speech that is smaller than noise and the inability to suppress non-stationary noise. Recently, deep neural network (DNN)-based SE has successfully been applied to single-channel speech enhancement. However, hearing aids have limitations such as low complexity and small memory that make it difficult to implement a recent and rich DNN-based system in them. To implement a tiny DNN-based SE, there is a trade-off between speech quality and noise suppression. Therefore, we designed a tiny DNN-based SE that compensates for the drawbacks of classical SEs. We used the minimum mean-square error log-spectral amplitude (MMSE-LSA) estimator (Ephraim et al., 1985) as a classical SE. By mixing the estimated gains by each algorithm, the noise suppression performance was improved.

We designed a tiny fully-connected DNN model that can be implemented in the digital signal processing (DSP) for hearing aids (onsemi: EZAIRO 8300). Log power spectra were used as input. The model was then trained by some open datasets and in-house recorded data to predict the ideal ratio mask. Subsequently, to find the approximate mixing rate between the DNN-based and the classical SE, we examined how the short-time objective intelligibility (STOI) values changed when the mixing rate was varied. The STOI was the highest for the 1:1 mixture, better than the results of the DNN-based SE alone and the classical SE. It was also confirmed that the amount of noise suppression was higher than that of the classical SE. In addition to the objective evaluation, subjective experiments were conducted with hearing people in a controlled environment. The results indicated that the speech quality could be improved. The proposed method was also implemented in the DSP for hearing aids and confirmed to operate with low latency. Thus, it was confirmed that the tiny DNN-based SE could upgrade the classical SE for hearing aids. On-site, we will bring our demonstrator. The attendees will be able to compare the proposed method and the classical SE by switching the mode.
Learning Hearing Aid User's Contextual Preference via Ecological Momentary Assessment in Real Life

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Background: Modern hearing aid (HA) provides the options of multiple programs in which different fittings in signal processing are adapted by the audiologists. Typically, these microphone programs are fit by audiologists for various real-life scenarios (e.g., speech-in-noise, lecture). Ideally, the HA shall adapt its program to the exposed context and to the users’ individual preference, and therefore to enhance the hearing experience with HA. To achieve the vision, the HA users’ preference needs to be learnt. Nowadays, HA users change programs in accordance with their intents in daily sound environments. Changing a program could indicate the preference in a given context, to a certain degree. To complement this information, an ecological momentary assessment (EMA) feature for providing feedback on the hearing experience comes as a popular tool.

Objectives: This study aims first to verify whether it is feasible to learn contextual preference by combing the context information, the program change action, and the EMA feedback. If so, a model for predicting the contextual preference is desired.

Method: Each test participant was fitted with a pair of Oticon More HAs, in which four programs were provided. The four programs form a 2×2 design which contains two HA features – noise reduction (NR) and high frequency gain (HFG), and two variables for each feature (for NR: default vs maximum, for HFG: default vs additional 4/6 dB). Moreover, participants used an extension of the standard Oticon ON app with EMA features on their smartphones. Participants could evaluate the overall satisfaction of the hearing experience and HA performance, as well as the label the exposed environment, user intent and motion via EMA. A convolutional neural network (CNN) embedded in the app attempts to validate the user environment, intent and motion labelling by classifying audio spectrogram input. The HAs report sound exposure data (e.g., sound pressure level, signal-to-noise ratio, etc) to the smartphone every 20 seconds, particularly when the test participants trigger an EMA session during the test period.

Study Sample: We invited 30 test participants with mild-to-moderate hearing loss. Each test participant will use the test HA and the app for 4-6 weeks. The participants were encouraged to experience diverse ambient types and to fill in the EMA as frequent as possible.

Result: At the time of submitting this abstract, we are in the process of collecting data. Thus, results are not reported at this moment but to be expected at the conference.

Can Active Noise Control Improve Sound Quality in Hearing aids?

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The superposition of sound processed through a hearing aid and the direct sound leaking into the ear canal can cause comb-filtering effects that degrade the perceived sound quality. Hearing aid wearers often perceive the effect as unnatural or as environment distortions. Common method to mitigate comb-filtering effects include adjusting occlusion to limit direct sound input and adjusting hearing aid amplification to limit the amount of interaction. These two options are often employed together to achieve a delicate balance between wearer comfort, providing the required amplification, and sound quality.

Recently, active noise control (ANC) has become increasingly popular in commercial consumer earbuds. Here, an anti-phase signal is generated to cancel out signals that leak into the ear canal. While ANC has been investigated in hearing aids to reduce the amount of perceived occlusion by reducing the level of own voice in an occluded ear canal, ANC could also be utilized to mitigate comb-filtering effects by actively suppressing signals leaking into the ear canal.

In the present study we investigate the potential of ANC to mitigate comb-filtering effects in hearing aids as a new method to balance wearer comfort, providing the required amplification, and sound quality. Specifically, we simulate combining ANC with conventional hearing aid processing. We present both instrumental predictions and subjective evaluations of sound quality for a variety of hearing loss profiles.

Predicted Selection of Listening Programs Based on Sound Exposure

Tiberiu-ioan Szatmari*, Alessandro Pasta†, Kang Sun†, Jeppe Christensen†, Niels Henrik Pontoppidan†
User-centric adaptation of audiological preferences across different contexts is a challenging task, as traditional clinical measurements of audibility do not reflect the cognitive perception of speech nor the binaural loudness of sounds in different contexts. Listening programs enable hearing aid users to adapt device settings for specific listening situations, which increases the personalization of their listening experience. However, studies investigating the real-world use of listening programs are lacking.

This study aims to investigate whether the selection of a specific listening program can be predicted based on the sound exposure of a user. Specifically, a binary classifier predicts whether the default ("General") or the most frequently used additional listening program ("Speech in Noise") is selected based on real-world, time-series sound environment data. Thus, we defined a time-series classification framework and compared multiple machine learning (ML) -based prediction models.

A state-of-the-art feature extraction model, Mini-Rocket, was used to transform environment features for the classification task. Due to the high number of features created by MiniRocket (10000 per raw sample), principal component analysis (PCA) was performed for reducing dimensionality to 1000 features per raw sample. Class imbalance during training was addressed by stochastic downsampling of the majority class. The final dataset was comprised of 6,613 program selections from 131 distinct users.

Next, we implemented two-level cross-validation to evaluate three binary classifiers: logistic regression (also serving as a baseline), gradient boosted trees, and a deep neural network. Due to the imbalanced nature of the data set (i.e., higher number of program selections in "General" over "Speech in Noise"), three metrics suited for imbalanced class distributions were used: balanced accuracy, F1-Score and Precision-Recall Area Under Curve (PR-AUC).

Training results on a balanced validation set showed the best-performing classifier – deep neural network – outperforms the other classifiers. Testing results on an imbalanced, realistic set revealed the model can make predictions given unseen data (balanced accuracy approx. 73%), but became less robust to imbalanced data (PR-AUC approx. 0.5). Finally, the deep neural network model was able to correctly label 69% of the “Speech in Noise” selections and 77% of the “General” selections in the testing set.

By rethinking contextual adaptation of HA settings as a time-series classification task, we validated the role of the sound environment in program selection and established a baseline approach for further investigating prediction models.

**DP132 Supervised Learning-Based Multi-Frame Filtering for Binaural Speech Enhancement**

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In many speech communication scenarios, head-mounted assistive listening devices such as hearing aids capture not only the target speaker but also interfering sound sources, resulting in a degradation of speech quality and speech intelligibility. To alleviate this issue, several binaural speech enhancement algorithms such as the binaural multi-channel Wiener filter have been proposed [1], which exploit spatial correlations of both the target speech and noise components. Similarly, for single-microphone scenarios it has been proposed to exploit the fact that speech is highly correlated over time, resulting in the multi-frame Wiener filter (MFWF) [2]. In this contribution, we propose a binaural extension of the MFWF, which exploits both spatial as well as temporal correlations. Similarly to [3], the binaural MFWF is embedded into an end-to-end supervised learning framework, where the required parameters are estimated by temporal convolutional networks (TCNs) that are trained using the mean spectral absolute error loss function. Simulations are conducted to evaluate the binaural MFWF in terms of its binaural speech enhancement performance as well as its ability to preserve binaural target localization cues. This evaluation is performed on a dataset comprising binaural room impulses measured with behind-the-ear hearing aids in realistic environments as well as diverse noise sources at a broad signal-to-noise ratio range. The simulation results demonstrate the advantage of multi-frame filtering instead of single-frame masking as well as the advantage of employing the binaural MFWF structure instead of directly estimating the binaural multi-frame filter coefficients. [This work was funded by the Deutsche Forschungsgemeinschaft (DFG, German Research Foundation) under Germany's Excellence Strategy - EXC 2177/1 - Project ID 390895286.]

**References:**
Feedback cancellation based on the LMS algorithm is commonly used to suppress howling as an oscillation due to acoustic feedback in hearing aids. However, the estimation error of the feedback path increases when signal with high autocorrelation is input to the LMS. This estimation error causes an artifact called entrainment, a phenomenon in which the acoustic input is distorted. An adaptive feedback cancellation based on a prediction error method (PEM-AFC), which decorrelates the inputs, has been proposed to solve this problem (Spriet et al., 2005). Still, the convergence performance is insufficient in real-world use in some cases. Therefore, we proposed an improved PEM-AFC using binaural information (Ueda et al., 2021).

The proposed method (Prop-AFC) solves the convergence problem of PEM-AFC by detection of howling based on the interaural level difference, and by controlling the decorrelation performance (Kawamura et al., 2008) of the PEM based on the detection. We compared the performance of feedback cancellation algorithms using computer simulations of binaural acoustic scenarios. The direction of input with high autocorrelation and the feedback path was varied in these simulations. Our experiments covered four different feedback cancellation algorithms: traditional AFC (T-AFC), conventional PEM-AFC, frequency-shifted AFC (FS-AFC), and Prop-AFC.

The experimental results were evaluated objectively and subjectively. Three objective evaluation indices were used: (1) misalignment (MIS), (2) maximum stable gain (MSG), and (3) artifacts-related score of PEASS (APS). The MIS confirms the accuracy of the estimation of the feedback path, and the MSG demonstrates the effectiveness of the hearing aids feedback suppression algorithm. In addition, we utilized the APS (Emiya et al., 2011) as a new entrainment evaluation index. Regarding the subjective evaluation, we conducted pairwise comparison listening experiments of the results of Prop-AFC, T-AFC, and FS-AFC, respectively. Participants selected the one with less entrainment. PEM-AFC was excluded from the evaluation because it is equivalent to Prop-AFC when the feedback path is stable.

Both objective and subjective evaluation results showed that the overall score of Prop-AFC was better than that of the other three methods. These results indicate that Prop-AFC converged faster than other algorithms under conditions where it is difficult to suppress howling with the conventional method, while the entrainment elimination performance of Prop-AFC was equivalent to that of PEM-AFC. Furthermore, these tendencies were similar even when the direction of arrival of the signal with autocorrelation changed.
mmercially available HAs in the Indian market for profound hearing impairment. These specifications were limited to essential parameters for appropriate amplification. To build upon an existing open-source hearing aid platform and a comparison of multiple available platforms was conducted, which pointed to Tympan Rev-D (an open-source hearing aid developed by Tympan) as the most suitable one.

Tympan Rev-D is a body worn hearing aid based on Teensy-3.6 processor, however, technical, and financial constraints within the Indian manufacturing ecosystem led to Design for Manufacturing (DfM) changes. Further changes were also undertaken to meet derived requirements. The developed HA, built upon Tympan Rev-D was subjected to a battery of tests in accordance with relevant standards (ANSI S3.22-2009 Specifications) using the Fyre-Phonix analyser and the results were found to be meeting all derived requirements. In summary, device had OSPL90 of 128dB, Full-On-Gain of 52dB, Equivalent Input Noise of 26dB, Total Harmonic Distortion of <1.4 % with suitable Frequency Response. Based on clinical immersion activities, a headband model was designed with children as the target user. The DfM changes led to a reduction in the cost of manufacturing by an order of magnitude (~10x) and enabling local manufacturing in India without compromising any of the technical requirements. The team is currently undertaking efforts for size reduction, and a device embodiment design catering to a child’s lifestyle. These design changes will be made available in the open-source domain. This can facilitate distributed economical manufacturing of open-source hearing aid hardware and its wider community adoption.

**DP135 Using Inverse Reinforcement Learning for Online Personalization of Hearing Aid Compressions**

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In the currently practiced fitting of hearing aids, prescriptive compressions are used. These prescriptive compressions are normally defined by taking gain averages of a group of people having more or less the same hearing loss or audiogram. In real-world audio environments, people with the same hearing loss do not necessarily express the same preference. It is thus hypothesized that personalization of the hearing aid fitting process provides an improved hearing experience for hearing aid users. Our research group previously developed a human-in-the-loop machine learning approach to personalize the compression function of hearing aids. A deep neural network was trained based on a user’s preference to predict the reward as part of a reinforcement learning framework. This approach was not deployable online in the field as the training of the deep neural network needed to be conducted offline. In this work, an alternative human-in-the-loop machine learning approach is developed which is deployable online. In this approach, first a user’s preference model is set up via paired comparison feedbacks received from the user for audio signals compressed by different compression settings. Then, the maximum likelihood inverse reinforcement learning method is used to find the personalized compression settings based on the user’s preference model. A comparison between the outcome of the personalized settings and the prescriptive compression settings of DSLv5 was conducted on 10 hearing-impaired subjects. The results showed that the personalized settings were preferred about 10 times more than the prescriptive settings. A word recognition experiment was also carried out which showed that not only the personalization approach did not have any adverse impact on speech understanding in noisy conditions, but also it improved speech understanding. Conducting compression in a personalized way would lead to higher user satisfaction with their hearing aids.

**DP136 Machine Learning Reveals the Importance of Extended High Frequencies for Predicting Speech-In-Noise Recognition in Listeners with Normal Audiograms**

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Nearly 10% of patients seen in the clinic complain of listening difficulty despite having a normal audiogram. Although decades of research have shown an equivocal relationship between the audiogram and speech-in-noise recognition, clinically, the audiogram remains the gold standard for hearing loss assessment and management. Can audimetric thresholds predict the ability to recognize speech in noise for normal-hearing individuals? We answer this question by using machine learning (ML) models. Specifically, the goals were to: (1) compare the relative performance of one standard (GAM: generalized additive model) and four machine learning models (ANN: artificial neural network, RF: random forest,
XGBoost: gradient boosting, DNN: deep neural network) for predicting speech recognition threshold (SRT); and (2) identify the relative contribution of audiomeric frequencies and demographic variables (age, ear, and sex) for predicting the SRT.

Data included audiometric thresholds (0.25 through 16 kHz) and SRTs measured in multi-talker babble using the digit triplet test from 764 participants (1528 ears; age 4-38 years). Test-retest SRT was assessed in 100 participants. All participants had normal audiograms (thresholds ≤20 dB HL; 0.25 to 8 kHz) and normal middle ear function. SRT was used as the response variable for building the ML models, while age, ear, sex, and audiometric thresholds were used as predictor variables. Prior to developing the models, the entire dataset was divided into the training (80%) and testing (20%) datasets. The training dataset was used to develop the corresponding models, while the testing dataset was used to evaluate the model predictions using the mean absolute error (MAE), which measures the average magnitude of the errors in a set of predictions. MAEs from various ML models were compared using the Wilcoxon sign rank test. The Shapley Additive exPlanations analysis was used to build explainable ML models for ANN and XGBoost. Additionally, change-point analysis was performed to track the age-dependent variations in pure-tone averages and SRT.

Among the ML models studied, XGBoost performed significantly better than other methods (MAE=1.62 dB). ANN and RF yielded similar performance (MAE= 1.68 and 1.67 dB, respectively), whereas DNN showed the worst performance (MAE=1.94 dB). The MAE for GAM was 1.61 dB. These MAEs were comparable with the mean test-retest SRT difference (1.09 dB). Feature importance analysis showed that age and extended high frequency (10-16 kHz) thresholds were the main predictors for SRT. Results will be further discussed in the context of hearing difficulty in the presence of a normal audiogram.

DP137 Optimizing Hearing Aids for Music Perception in Noise Using Subjective and Objective Measures
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The perception of music is significantly compromised in the individuals using hearing aids (HA), which becomes further degraded in the presence of noise. Although HAs are heavily optimized for speech intelligibility in noise, they often are not well optimized for music perception, especially in the presence of noise. This study investigated the effect of signal-to-noise ratios (SNR) on music processed through HAs in two programs (1- Speech and 2- Music) using subjective and objective measures. The HA (fitted to NAL-NL2) processed music stimuli in the two programs at 5 SNRs (0, +5, +10, +15 and +20) were rated perceptually by 20 normal-hearing participants on a visual analogue scale of ‘Adjective descriptors of Music’ adapted from ‘Iowa Musical Background Questionnaire’, while the quality of music was objectively analyzed using a music quality index (HAAQI) using MATLAB code. Results showed that the music perception was better for higher SNRs (+15 and +20) than lower SNRs (0, +5 and +10) in both subjective and objective measures for both the programs. The effect of SNRs was more pronounced in the music program. These results emphasize the need for subjective and objective validation of HA programs for music listening in clinical setups.

DP138 Evaluation of a Deep Neural Network-Based Noise Reduction Algorithm in Cochlear Implant Listeners
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Complex acoustic environments with multiple competing talkers limit the ability of cochlear implant (CI) listeners to understand speech. Traditional single-microphone noise reduction algorithms, based on assumptions of signal statistics, can improve speech intelligibility for CI listeners in stationary noise but not in modulated babble noise. Deep neural network (DNN) based noise reduction solutions rely on a data-driven approach to train the network and promise hearing improvements over a wide variety of noisy conditions.

In this study a DNN-based noise reduction algorithm was evaluated in CI listeners. The network evaluated was a commercially available DNN-based noise cancellation application known as KrispTM (https://krisp.ai/). A double-blinded evaluation was conducted with 10 adult cochlear implant users by assessing intelligibility and sound quality of speech embedded in multiple different noise types, including stationary speech-weighted noise, 4- and 20- talker babble, and recordings of a cafeteria, a cocktail-party and street-side city noise. The study also investigated
the potential benefits at different signal-to-noise ratios (SNR). None of the speech or noise materials used in testing were used to train the network.

Speech reception thresholds, the SNR required to understand 50% of the speech material, improved by 4.4 dB in speech-weighted noise and 2.6 dB in 20-talker babble. No significant difference in 4-talker babble was found when compared with the unprocessed reference. Source quality ratings improved in all noise types evaluated, including 4-talker babble. Benefits varied with noise type and SNR of the input signal. While substantial benefit was observed over most SNRs, the DNN produced poor intelligibility at SNRs below approximately 0 dB. Objective metrics of speech intelligibility and sound quality were calculated on the processed signals and compared against the perceptual outcomes, providing insight into the effects of SNR.

While the study was limited to acute evaluation of a PC-based algorithm in laboratory conditions, the intelligibility and sound quality benefits observed with KrispTM demonstrate the potential of DNN-based noise reduction for significantly improving speech communications with Cis in noisy environments.

Results: We have validated the tool for software correctness. We will present feasibility data from users, demonstrate HO2 Apps in our presentations, and provide hands-on training to interested IHCON attendees. Subgroups can focus on one or more features of HO2 Apps and share their configuration files with others.

Background: Modern digital hearing aids (Has) are extremely effective in compensating for hearing loss (HL) in most situations except in those that hearing-impaired listeners need most – multiple talkers and background noise. The adage from business management “If you can’t measure it, you can’t improve it,” is very apt for modern Has. There is a need to build successful, clinically relevant tools such as Quick-SIN, Modified Rhyme Tests, Acceptable Noise Level, etc. in an integrated research environment using open-source subband amplification, feedback cancellation, frequency translation, and other such tools. Further, such tools shall support measuring objective outcomes in a repeatable, and reproducible manner.

Approach: We developed a suite of Hearing Aids Objective Outcomes (written as HO2 and pronounced as [hot] /ho:t/ ) Apps. (i) We developed a multi-lingual data collection tool for audio and audio-visual stimuli based on minimal contrast sets (MCS) of words in a given language. Each word in a given subgroup of words in MCS differs in one and only one (acoustic) phonetic feature that results in a different meaning – phonemics of the given language. (ii) The resulting stimuli content is uploaded to HO2. (iii) HO2 provides functionality for the investigations to configure (a) unaided and aided conditions; the aided conditions comprising multiple fitting strategies for a given HA sound processing algorithm and a given fitting strategy for multiple HA sound processing algorithms; (c) select audio stimuli with and without video; (d) add different noise types at different SNR levels; and (e) automatically generate word-level accuracies, phonetic confusion matrices, and broad phonetic class confusion matrices. (iv) The broad phonetic classes are based on the phonemic classes for a given language and include both place of articulation and manner of articulation during speech production. (v) The confusion matrices enable researchers to use HO2 iteratively by selecting stimuli that address perception errors at individual and at group levels and investigate various static and dynamic aspects of subband amplification. (vi) HO2 can save the above conditions in a configuration file and share the file with other researchers for repeatability, reproducibility, and contributions to novel research paradigms.

Results: We have validated the tool for software correctness. We will present feasibility data from users, demonstrate HO2 Apps in our presentations, and provide hands-on training to interested IHCON attendees. Subgroups can focus on one or more features of HO2 Apps and share their configuration files with others.

Background: While modern digital hearing aids (Has) are extremely effective in compensating for hearing loss (HL), frequent complaints from the users persist, notably poor speech intelligibility in noisy environments and high cost, among other issues. However, the signal processing and audiological research needed to address these problems has long been hampered by proprietary development systems, underpowered embedded processors, and the difficulty of performing tests in real-world acoustical environments. To facilitate existing research in hearing
healthcare and enable new investigations beyond what is currently possible, we have developed a modern, open-source hearing research platform, Open Speech Platform (OSP). This contribution introduces the basic, and advanced signal processing algorithms and describes Researcher Apps aimed at advanced audiology research.

**Approach:** The basic HA features currently supported in OSP comprise three classes of subband amplification (i) temporal (FIR based) 6-band system that can be extended, (ii) FFT based multiband openMHA from Oldenburg, and the temporal (multirate) 11-band system; a suite of “least means squares”-based acoustic feedback cancellation (AFC). The advanced HA features in OSP include real-time beamforming between left and right microphones, and a novel frequency translation module. This module (called Frep- ing, a portmanteau for Frequency Warping) is based on a modified all-pass network that performs frequency compression and expansion in each subband independently, based on a single parameter in the range of +/-1. Positive values enhance the temporal fine structure, while negative values reduce the spectral footprint to accommodate elevated hearing thresholds in some bands. Low Freping values (around +/-0.01) provide an added stable gain of about 10 dB, allowing audiologists to prescribe more amplification without “whistling” artifacts. Audio quality changes are barely perceivable at such Freping values. Higher Freping values (around +/-2) may improve speech intelligibility at lower amplification prescriptions, but additional research is required to understand the benefits. These researcher apps have been used successfully in self-fitting research such as Goldilocks – a search and select approach or amplification.

**Results:** We will demonstrate Researcher Apps running in real-time. The app includes NAL-NL2 prescriptions for the standard audiograms developed by the “International Standards for Measuring Advanced Digital Hearing Aids” and the ability to adjust both amplification and Freping values in multiple bands. We will provide hands-on training to create multiple prescriptions that can be used in intelligibility and ecological momentary investigations using the OSP.

Previous studies have reported improved localization and better speech perception when the unilateral CI is coupled with a hearing aid in the contralateral ear. This is referred to as bimodal presentation of speech. A major obstruction to accurate source localization for bimodal CI users is the distortion of interaural time and level difference cues (ITD and ILD), and limited ITD sensitivity. Hence, it is necessary to develop and test algorithms that provide better localization and sound source identification cues. Various research interfaces developed by either academic or industry sponsored research teams support proposed signal processing and psychoacoustic investigations but have limited ability to efficiently validate bimodal algorithms. Platforms that support bimodal testing are either not portable or only provide limited features due to proprietary parameters/routines. Thus, the open-source, portable signal processing platform, Cci-MOBILE developed by UT-Dallas, enables electric and acoustic bimodal stimulations simultaneously, providing researchers the freedom to explore new technology and scientific paradigms. In this study, we perform a systematic verification of the synchronized bimodal (electric-acoustic) output in an authenticated and efficient manner to ensure effective left/right processing pipeline timing support of algorithmic and experimental investigations focused on localization.

Cci-MOBILE captures audio in real-time through two behind the ear (BTE) microphones via an audio codec and streams the digitally processed signal through a UART port to the computing platform (PC/smartphone). Sound processing routines send the RF data through one channel and digitized acoustic data through the other channel via the UART port. The delays and system clock settings are tuned such that the middle and inner ear frequency dependent delay are taken into consideration, while synchronizing the HA output with the CI output. The output to the CI is sent to the RF coil to deliver biphasic pulses in a continuous interleaved manner (CIS). The amplified HA output is sent to the HA transducer channel, after Wide Dynamic Range Compression (WDRC). Results were verified using an oscilloscope to demonstrate the timing validity of left and right output signals superimposed on each other, proving evidence of simultaneous CI/HA synchronization. Bimodal stimuli synchronization was also verified by comparing Electric and Acoustic Stimulation (EAS) outputs on an oscilloscope to demonstrate equalization of the input and output time delay for both channels, as well as with formal subjective tests.

**DP141 Cci-MOBILE: Bimodal Synchronization and Algorithm Testing with the CI/HA Research Platform**

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IHAS August 9-10, 2022

IHCON August 10-14, 2022
DP142 Effect of Minimum Phase Processing on Binaural Hearing in a Spatial Release from Masking Paradigm

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In open hearing aid (HA) fittings, both the processed sound and the direct sound are transmitted into the ear canal. Due to processing delay, typically <10 ms, the sum of processed and direct sound results in a comb filtering of the signal, which negatively affects the sound quality. In order to mitigate this effect, some commercial HAs utilize minimum phase processing. The use of minimum phase filters results in a group delay that is frequency- and gain-dependent. In a binaural HA setting, gain differences between the two ears can result in a different delay at each ear, potentially altering the interaural time difference (ITD) in the processed sound. The potential alteration of the ITDs is likely to be more detrimental in closed fittings, where only the processed sound is presented. ITDs, alongside ILDs, are important binaural cues. The present study aimed at investigating the potential perceptual consequence of applying minimum phase processing on spatial perception in a closed HA fitting. For that purpose, the spatial release from masking (SRM) paradigm was employed, where listeners were assessed on the benefit (i.e., “release”), they received in speech intelligibility through the spatial separation of the masking signals. SRM was measured in 18 normal-hearing listeners with both (1) broadband stimuli and (2) lowpass filtered stimuli with a cutoff frequency of 1 kHz, where it was assumed that participants would primarily have access only to ITD cues. The results showed that the lowpass filtering resulted in a reduced SRM; however, there was no significant difference in SRM between the minimum and linear phase processing conditions with either stimulus type. The results therefore, did not reveal any detrimental effect of the minimum phase processing on speech perception. This indicates that, at least in the conditions tested, minimum phase processing can be a successful approach for optimizing sound quality in hearing aids without compromising speech intelligibility.

DP201 Practice Listening and Understanding Speech (PLUS): Co-Development and Usability of

Two Novel Auditory-Cognitive Training Interventions

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Listening to degraded auditory input is cognitively challenging, whether due to hearing loss alone, or alongside the use of amplification. Auditory training has been shown to result in improvements in speech perception and cognition for people with hearing loss and hearing aid users. Evidence from our own research suggests these improvements are driven by refinements in higher-order cognitive control processes (executive functions). A meta-analysis (Lawrence et al., 2018) suggests the largest benefits to cognition for people with hearing loss may be achieved via interventions that combine auditory and cognitive training approaches (auditory-cognitive training).

Two bespoke auditory-cognitive training interventions were developed with patients and the public, designed to target and enhance executive functions through; 1) refinement of perceptual and cognitive skills (phoneme discrimination n-back training), and 2) development of cognitive control when listening to speech (competing-talker training). For phoneme discrimination training, stimuli were those reported by Ferguson et al. (2014), presented via a 1-back or 2-back odd-one-out task. For competing-talker training, novel sentence stimuli inspired by the Coordinate Response Measure (CRM) were co-developed with ten hearing aid users. Participants took part in PhotoVoice activities and workshops. PhotoVoice is a qualitative research technique designed to document and reflect reality. Participants photographed their most challenging listening situations over 7 days, then attended workshops where their photographs and descriptions formed the basis of in-depth discussions used to generate situations and sentence stimuli. For stimuli recordings, talkers were recruited from local theatres and University performing arts courses. Audio-recording auditions informed the selection of 4 talkers (2 male, 2 female) that provided variations in voice pitch. Recordings took place in a custom anechoic chamber, with careful attention paid to matching talkers’ speech rate and stress patterns both across talkers, and within talkers across sentences, for each situation. Feedback from patients and the public helped shape the user instructions and visual design for both training tasks.
The training interventions were iteratively refined through extensive at-home usability testing with hearing aid users, via: Think Aloud interviews (n=8), at-home trials (n=8), semi-structured interviews (n=8), and training diaries (n=15). Suggested changes were tabulated and 136haracteriz for implementation using the MoSCoW criteria (Kuhn, 2009). The resulting interventions are highly usable and aligned to patient need. The training interventions will be provided to first-time hearing aid users across two UK National Health Service (NHS) audiology clinics to assess the feasibility of conducting a full-scale 136haracteriz trial of their clinical and cost-effectiveness.

DP202 Understanding the Relation between Real-Ear Measurement, the NAL-NL2 Fitting Target and SSQ in Clinical Rehabilitation
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There are many different factors that might be associated with the success of hearing aid (HA) use, such as hearing loss, previous experience with Has, age, or listening environments. In clinical practice, a HA fit is often evaluated by a Real Ear Measurement (REM) or by measuring speech performance. However, objective measures only are not sufficient to define the degree of success as they do not take into account self-perceived auditory functioning in real-life situations. In this study the aim was to investigate the effect of REM characteristics along with other (mediating) factors on self-perceived functioning.

Data was collected from subjects visiting the Erasmus Medical Centre between 2015 and 2020. The Speech Spatial and Quality (SSQ) questionnaire, a measure of self-perceived ability, was used to rate the success of the HA fitting. All subjects (n=508) completed a HA trial period that included pre- and post-SSQ results. The mean age was 63.9 years, mean hearing loss (PTA0.5,1,2,4) was 50.5 dB, and 44.1% had no previous HA experience. Data analysis included REM results, NAL-NL2 targets, and results of (un)aided speech intelligibility in quiet. The quality of the HA fit was defined as the Real Ear Aided Response Difference to Target (READT) at 65 dBSPL, and characterized by total variation (READTSD) and overall deviation (READTmean). Factors such as age and HA experience, were also included.

Results showed a positive increase in SSQ-scores on all SSQ-domains (p<0.001; t>0.65, i.e., large effects).

For SSQ-Speech domain, READTSD, unaided speech intelligibility and pre-SSQ had a significant and large effect on aided speech intelligibility. READT–SD had a small, but significant effect on post-SSQ scores for the Speech and Spatial domain. In case of the SSQ-Speech domain, this effect was fully mediated by aided speech intelligibility (with no direct effect of REM). READTmean had no impact on aided speech intelligibility or post-SSQ. Other factors (e.g. age, HA experience), showed small, but significant, effects on post-SSQ and/or aided speech intelligibility.

Self-perceived auditory functioning with newly fitted Has is primarily predicted by pre-SSQ scores, and to a lesser extent effect of READTSD. Yet, READTSD had an important impact on aided speech intelligibility. Considering the effect of READT on self-perceived auditory functioning and aided speech intelligibility, we conclude that maintaining the shape (READTSD) of the fitting target seems to be more important than the overall deviation (READTmean).

DP203 Efficacy of Individual Computer-Based Auditory Training for People with Hearing loss: An Updated Systematic Review and Meta-Analysis
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Auditory training can be described as a process of training the brain to listen through active engagement with sounds. It has been shown to result in improved speech perception over the course of an adult’s lifespan (Wright and Zhang, 2009). As such, auditory training can be offered prior to, or alongside hearing devices to help improve listening for people with hearing loss. Because auditory training interventions can be delivered via computers and the internet, they offer low-cost forms of self-management support that can be individually tailored and conveniently accessed by people with hearing loss. However, for auditory training to be considered effective, it should result in sustained improvements that extend beyond trained tasks, to benefit everyday listening abilities.
In 2013, a systematic review of the efficacy of computer-based auditory training interventions for adults with hearing loss identified 13 eligible studies and concluded that the published evidence was not robust and therefore could not be reliably used to guide intervention decisions at that time. The authors identified a need for high-quality evidence to further examine the efficacy of computer-based auditory training in this population.

Here we report an in-progress update to the systematic review with the addition of meta-analyses. Registered via Prospero (CRD42017076817), this updated review addresses the primary research question “Does evidence exist to support improvements in trained and untrained measures of speech perception, cognition, and self-reported communication or quality of life, as a result of individual computer-based or internet-based auditory training in adults with hearing loss (with or without hearing aids or cochlear implants)?” Twelve electronic bibliometric databases and trial registries were searched during December 2021. Randomised controlled trials, non-randomised controlled trials, cohort studies, and repeated measures designs were all considered for inclusion.

To date, we have identified 51 studies eligible for inclusion in the updated review and data extraction is ongoing. Meta-analyses will examine the magnitude of improvement (learning) for trained tasks and transfer of learning to untrained outcomes for the domains of speech perception, cognition, and self-reported hearing, across different training stimuli (speech/music) and participant device use (none/hearing aids/cochlear implants or bimodal). A narrative synthesis will describe key study characteristics and additional review findings. This review will provide a comprehensive synthesis of the robustness and estimates of efficacy of computer-based auditory training for adults with hearing loss, to help guide future evidence-based hearing healthcare and research.

DP204 Hearing Aids Combined with Educational Counseling versus Educational Counseling Alone for Tinnitus Treatment in Patients with Hearing loss: A Longitudinal Follow-Up Study

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Objective: Hearing aids (Has) and educational counseling (EC) are commonly recommended for tinnitus management in patients with hearing loss (HL) and persistent, bothersome tinnitus. However, the relevant studies are limited, and the effect of two treatments or the combination is ambiguous. This study aimed to investigate whether the combined use of Has and EC is more effective than EC alone on tinnitus relief.

Methods: A total of 72 adults with chronic, bothersome tinnitus and co-existing sensorineural HL were included. After receiving EC and Has prescription, 21 participants self-selected to purchase Has and receive EC (i.e., HA+EC group), while the remaining 51 refused to use Has and only received EC (i.e., EC group). They were asked to return for follow-up at 1 and 3 months after treatment initiation, and were encouraged to follow up frequently. At baseline, they all completed audiologic tests, tinnitus pitch matching, Tinnitus Handicap Inventory (THI), Tinnitus Evaluation Questionnaire (TEQ), Visual Analogue Scale (VAS) for loudness, Self-rating Depression Scale (SDS), and Self-rating Anxiety Scale (SAS). At each follow-up visit, both groups received EC sessions and completed THI, TEQ and VAS. Besides, International Outcome Inventory for Hearing Aids (IOI-HA) was administrated in the HA+EC group at 3-month follow-up. The primary outcome measure was THI, and tinnitus relief was defined as a 20-point or more reduction in THI. This trial has been registered with Chinese Clinical Trial Registry (ChiCTR1900022624).

Results: Demographic variables, status of insomnia, depression, and anxiety, degree of HL, tinnitus pitch, initial scores for THI, TEQ and VAS were homogeneous across two groups. THI, TEQ and VAS scores decreased significantly after treatments (Wilcoxon signed rank test, all p < 0.05), and both groups yielded a similar trend of reduction. Differences in time-to-event curves (i.e., cumulative tinnitus relief) between groups were insignificant (Log-rank test, p = 0.63). Moreover, there was no significant difference in the incidence of tinnitus relief between the two groups (61.90% and 47.06% respectively, p = 0.25). Participants in HA+EC group were generally satisfied with Has based on the mean score of 25.08 on IOI-HA. Conclusion: Receiving HA+EC and EC were equally effective for tinnitus management. There was insufficient evidence to support the superiority of the combined use of Has and EC for tinnitus over EC with no device.
Large-scale datasets, such as the UK Biobank and the Human Connectome Project, have proved extremely powerful for facilitating research into mechanisms of human health and disease. A limitation of most existing resources, however, is that hearing health phenotypes are captured at a rudimentary level, where even basic pure-tone audiometry data are rarely available. This severely limits the scope of the questions that can be asked of these datasets from an auditory perspective. The Nottingham Hearing BioResource (NHB) represents our effort to begin leveraging the power of large, open, accessible datasets in a way that could transform how we treat and manage hearing loss and hearing-related conditions (e.g., tinnitus, hyperacusis, Ménière’s disease) in future.

At its core, the NHB will provide a comprehensive collection of high-quality, person-centric samples and data focused on hearing health and related domains. Biological samples and data will be made available to academic and commercial researchers around the world in accordance with FAIR (findability, accessibility, interoperability, and reusability) principles. By marrying genetic and biomarker information with measures of noise exposure history, advanced audiological assessment (including extended-high-frequency audiometry, wideband tympanometry, otoacoustic emissions, electrophysiology, auditory perception), and longitudinal tracking of hearing and wider health outcomes, the NHB aims to support research that will radically improve our ability to: 1) understand individual risk; 2) diagnose individual pathology; and 3) predict individual outcomes. At the same time, the NHB will provide a database of well geno/phenotyped individuals who can be recruited in a targeted way into future clinical trials of emerging treatments for hearing loss, such as those based on gene, drug, or cell-based technologies.

With input from a network of international experts, we have been working on several important aspects of the NHB, including: 1) governance, ethical and data access arrangements that align with international norms and published best practice for biobanking; 2) data infrastructure and data standards that will allow for aggregation and interoperability with other datasets around the world; 3) protocols for advanced audiomteric assessment, with a particular focus on measures providing information “beyond the audiogram.” In relation to this last aspect, we will report the findings of a small-scale pilot study conducted to establish how the wideband middle ear muscle reflex (MEMR) can be robustly measured using standard clinical equipment.

We look forward to providing an update on progress with the NHB to date, to share learning amongst the hearing community and to prompt discussion.

**DP206 Hearing Characteristics of the Participants Enrolled in the Aging and Cognitive Health Evaluation in Elders (ACHIEVE) Randomized Controlled Trial**

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¹University of South Florida, ²Johns Hopkins University, ³University of Mississippi Medical Center

Novel approaches for reducing cognitive decline in older adults are needed given the aging population and the personal, socioeconomic, and public health implications of cognitive impairment in older adults. Hearing loss is independently associated with accelerated cognitive decline and thought to be the largest potentially modifiable risk factors for dementia. We will describe the patient-centered, best-practices audiological assessment and intervention for older adults with hearing loss, developed for the Aging and Cognitive Health Evaluation in Elders randomized controlled trial (ACHIEVE RCT; ClinicalTrials.gov Identifier: NCT03243422). The ACHIEVE RCT is the first large-scale trial to evaluate if best-practices hearing intervention can reduce cognitive and other functional declines in older adults. In addition to describing the hearing intervention, we will describe the baseline hearing function and performance (pure-tone audiometry, word recognition in quiet, speech perception in noise) and self-reported handicap for the 977 older adults enrolled at the four I study sites located in Washington County, MD; Jackson, MS; Forsyth County, NC; and Minneapolis Suburbs, MN. Participants were recruited from either the ongoing Atherosclerosis Risk in Communities (ARIC) longitudinal observational study (ClinicalTrials.gov Identifier: NCT00005131), which has been ongoing since 1985, or de novo from the community.

Participants better ear median 4-frequency pure-tone average was 38.8 [IQR: 33.8, 43.8], while the poorer ear pure-tone average was 42.5-dB HL [IQR: 37.5, 47.5]. Word recognition performance ranged from
16% to 100% in the better ear. Mean speech in noise performance measured with Quick Speech in Noise Test (QuickSIN) was 7.1 dB SNR Loss (SD= 5.2). Hearing Handicap Inventory for the Elderly – screening version (HHIE-s) indicated majority of participants had Mild-Moderate handicap (n=487). There were strong and statically significant correlations between measured hearing function and self-reported hearing ability. Interestingly, there were self-reported handicap differences based on geographical location and related to recruitment route (ARIC vs. de novo). The baseline audiometric and hearing-related self-report characteristics of the ACHIEVE participants will help to inform the primary analyses planned for the ACHIEVE RCT which are anticipated for completion in late 2022/early 2023.

**Results:** To date, 9 participants have been enrolled (4 OID, 5 Non-OID). Baseline hearing loss and neurocognitive results will be shared. Participants SPIN abilities pre-intervention and post-intervention will be shared. Objective results on the Words in Noise (WIN) test revealed a 2 dB SNR and 0.5 dB SNR average improvement, and subjective results on the Speech Spatial and Qualities of Hearing Scale (SSQ-12) revealed a 30-point and 4-point average increase for the OID and non-OID users respectively, from baseline to follow up.

**Conclusions:** Our preliminary data suggests a greater improvement on objective and subjective SPIN abilities in the OID users pre vs post OID intervention. As expected, the non-OID users show stable SPIN abilities at baseline and follow up testing. This is an ongoing pilot study that will inform future grant submissions and randomized controlled trials. Our results are expected to continue to provide preliminary evidence of OID hearing intervention on hearing and cognition performance and the mediation of this non-traditional hearing rehabilitation treatment on cognitive decline.

**DP301 Estimating Speech Intelligibility Using Google Speech-To-Text**

*TERENCE BETLEHEM*, **DANIEL MARQUARDT**, **MARTIN MCKINNEY**

*Starkey*

Speech enhancement algorithms are important for improving the listening experience of hearing aid users in adverse listening environments. To understand the performance of a speech enhancement algorithm, it is crucial to evaluate the performance perceptually with respect to signal quality and speech intelligibility (SI). Typically, SI is determined through subjective word recognition tests. These tests are time-consuming and hence can only be performed on limited occasions in the product development process. Alternatively, SI may be estimated using instrumental intelligibility metrics.

Instrumental intelligibility metrics are quick to compute but have limitations. They are typically intrusive, only applicable to certain speech corpuses, and limited in use to specific types of channel distortions and enhancement algorithms. The most accurate of the metrics (i.e., the most closely correlated with SI) are intrusive, i.e. they required a clean reference signal of the original uncorrupted speech to be available, in addition to either the channel or the distorted speech.
When evaluating algorithms using subject speech collected from noisy environments, however, the reference signal is often not available.

A commonly used metric for estimating SI is the short-time objective intelligibility (STOI). STOI was designed for predicting SI of time-frequency weighted noisy speech. Unfortunately, the ability to map STOI to SI is only available for the speech corpuses of IEEE and Dantale. Also, STOI is intrusive. A non-intrusive version of STOI, NI-STOI, was proposed [3] but is not as well correlated with SI as the intrusive version. More recently, SI has been accurately estimated using custom-made speech recognizers.

In this presentation, we evaluate a method for non-intrusively obtaining an estimate of speech intelligibility, by using a commercially-available speech recognizer. Using the QuickSIN speech-in-noise testing framework to determine SI, we study the ability of Google Speech-to-Text to predict the SI for speech-in-babble noise. We compare its performance against intrusive and non-intrusive metrics including STOI and NI-STOI. The scope of this study is listeners with normal hearing but will be extended to impaired listeners in future work.

DP302 Outcomes from Combined Logging Data from Hearing Aids and Clinical Databases
Niels Henrik Pontoppidan*, Jeppe H. Christensen¹, Louisa Murdin², Athanasios Bibas³, Dimitris Kikidis⁴, Giorgos Dritsakis⁴, Doris-Eva Bamis⁴
¹Eriksholm Research Centre, Oticon A/S, ²Ear Nose and Throat Service, Guy’s and St Thomas’ NHS Foundation Trust, London, United Kingdom, ³Department of Otolaryngology, National and Kapodistrian University of Athens, Greece, ⁴Ear Institute, University College London, United Kingdom

Background: A good dialogue between hearing care professionals and hearing aid users is generally believed to lead to better fit and yield better outcomes. However, presently the efficiency of the dialogue between hearing aid users and hearing care professionals about usage and fitting of hearing aids is constrained by the precision of the vocabulary describing sounds and environments, the memory about the situations, and the exact way hearing aids were used. Often, the hearing care professional will find it difficult to assess if the better ability to hear in a difficult situation three weeks ago compared to two weeks ago due to the situation or operation of the hearing aids. Furthermore, insights from developing hearing aid processing reveals that the interaction between the complexity of the scene impacts the perceived benefit of noise reduction which are difficult for the hearing aid user to perceive and describe.

Objectives: We wish to explore if data from everyday use of hearing aids combined with data from clinical databases provides a deeper insight into the actual usage patterns and benefits from hearing aids taking both context, fitting, and the other data that describes each individual user into consideration.

We then wish to explore if those insights can be transferred into new methods that hearing care professionals can use to provide more support for people with hearing problems.

Method: 400 hearing aids users used the EVOTION hearing aid and EVOTION mobile app for up to 12 months giving rise to more than 50 MIO multidimensional data vectors describing context and hearing aid settings combined with a comprehensive range of clinical data that describe the users by their hearing loss, age, previous hearing aid experience, additional medical data related to hearing, and questionnaires like GHABP, HUI3, MOCA, and HADS.

The data is analyzed using linear mixed models that allow for specifying fixed effects and random effects with continuous and categorical labels.

Results: The initial results from analyzing a homogenous sample of 40 out of the 400 participants indicates interactions between MOCA outcomes, daily usage, HL. However, we refrain from drawing too firm conclusions from the initial analysis of the limited data prior to conducting the full analysis before the IHICON 2022 meeting.

DP303 Developing a predictive model of adult hearing loss using self-reported outcomes
William Whitmer*¹, Polly Scutt²
¹Hearing Sciences – Scottish Section, University of Nottingham, Glasgow, UK, ²Nottingham NIHR Biomedical Research Centre

While there are options for the remote 140haracterized140m of hearing devices, even the most efficient methods (e.g., the Van Tassell method) require the use of calibrated equipment. Self-reported hearing outcomes have shown to be effective for screening but not necessarily for fitting. Here, we examine how well self-reported hearing ability, difficulty and handicap outcomes, together and on their own, are diagnostic predictors of adult pure-tone hearing thresholds.
This analysis used data collected 2002-2009 of 1008 adults recruited from local audiology clinics, aged 18-88 years with varying hearing and hearing-aid status, who underwent complete audiometric testing as well as responded to three questionnaires: the Speech, Spatial and Qualities of Hearing scale (SSQ), the Hearing Handicap Questionnaire (HHQ) and the Glasgow Hearing Aid Benefit Profile (GHABP). Predictors were age, sex, the three subscales of the SSQ, the HHQ and the unaided difficulty and handicap questions of the GHABP (averaged across situations). Multiple linear regression using stepwise selection procedure to select predictors was used to model both better-ear pure-tone four-frequency average thresholds (BE4FA) as well as audiometric categories based on the standard audiograms of IEC60118-15. Models were cross-validated (k = 10) and compared using Akaike information criteria.

A full model including all predictors fit the data reasonably well (e.g., BE4FA R2 = 0.63). A model consisting of only age and the GHABP difficulty score performed nearly as well as the full model, whereas other predictor combinations (e.g., HHQ, SSQ subscales) fared much worse. All models were limited in their ability to predict the most severe hearing losses. These results show how a small set of hearing difficulty questions could provide an efficient means to help characterize remote hearing healthcare. By incorporating separate ability, difficulty and handicap outcomes, the current analysis reveals differences in these oft considered compatible domains and highlights the importance in selecting the relevant hearing domain for diagnostic prediction.

11:20 a.m. – 12:40 p.m.

**SESSION D2. Epidemiological and Intervention Studies Related to Hearing Aids**

Session Chair: Inga Holube

**D2-1. Evidence for and the Development of a Best-Practices Hearing Intervention as a Possible Modifier for Cognitive Decline**

*Victoria Sanchez*<sup>1</sup>

<sup>1</sup>University of South Florida

Hearing loss is known to negatively impact communication and quality of life, and broader implications of hearing loss for the health and functioning of older adults are now consistently reported in epidemiologic studies. One critical result of over a decade of observational research is the finding of independently accelerated cognitive decline with hearing loss. In fact, hearing loss was cited as the largest potentially modifiable risk factors for dementia that may account for 8-9% of all dementia cases. Although a causal association remains to be determined, hypothesized mechanisms underlying the observed association between hearing loss and dementia include the effects of distorted encoding of sound on cognitive load, changes to brain structure and function, and/or reduced social engagement. Importantly, these mechanisms may be modifiable with comprehensive hearing intervention. We will summarize current research on the association between hearing loss and dementia and review the hypothesized underlying mechanisms. Then, current studies investigating the impact of hearing interventions on reducing cognitive decline and the risk of dementia in older adults will be discussed. We will focus specifically on the ongoing Aging and Cognitive Health Evaluation in Elders (ACHIEVE) study, which is the first large-scale trial to evaluate if best-practices hearing intervention can reduce cognitive and other functional declines in older adults (Clinicaltrials.gov NCT03243422; NIH funded R01AG055426, MPIs: Lin/Coresh). This multisite clinical trial recruited 977 adults with untreated mild-to-moderate hearing loss in 2018-2019. Participants were randomized and are currently receiving either a successful aging education control intervention or best-practice hearing intervention. We will describe the patient-centered, best-practices audiological assessment and intervention being used in the
ACHIEVE study through four general process areas: (1) Assessment and Goal Settings; (2) Technical Aspects of Treatment; (3) Orientation, Counseling, and Self-Management; and, (4) Outcomes Assessment. In addition to the intervention details, we will present characteristics of the enrolled ACHIEVE participants and describe the intervention quality control monitoring that has monitored the fidelity of the intervention. Participants are being followed for three years with final study visits scheduled in end of 2022, at which time the main trial results will then be available. Finally, we will describe follow-up studies and supplemental studies that are addressing research questions on telehealth delivery of hearing intervention and how provision of hearing intervention at milder hearing losses or at a younger age might impact brain health and cognition.

D2-2. Cognition and On-Beat Rhythmic Ability Support Speech-In-Noise Perception for Older Adult Hearing Aid Users

Chi Yhun La*, Ella Dubinsky1, Gurjit Singh2, Frank Russo1
1Toronto Metropolitan University, 2Toronto Metropolitan University, Phonak Canada, University of Toronto

Musicians tend to perform better than non-musicians on a range of auditory tasks, such as speech-in-noise (SIN) perception. However, the relative contributions to this advantage (i.e., musicians’ innate abilities and/or learned experience) remains contentious. Associations between working memory, frequency discrimination, and SIN performance are relatively well explored and understood, while the role of rhythmic abilities has been investigated to a much lesser extent—particularly in populations with hearing loss. Nonetheless, multiple studies suggest that the motor system and rhythmic abilities are correlated with better SIN abilities. This may have implications for individuals with hearing loss, who are typically characterized by poor frequency discrimination (i.e., pitch and timbre representations) but intact rhythmic abilities.

The present study is part of a larger study investigating the benefits of choir-based training and active music listening for adults with hearing aids. The present dataset explored pre-training correlations between outcome measures that were broadly stratified as speech perception, cognition, and psychoacoustic/musical tests. The aim of this study was to investigate the relative contributions of psychoacoustic outcomes on speech perception.

Participants were 38 older adult hearing aid users with a moderate bilateral hearing loss (M = 46.7 dB 4FAHL) aged between 57 and 90 years (M = 72.8 years, 25 female and 13 male). All participants passed the Montreal Cognitive Assessment (MoCA) screening for mild cognitive impairment and Alzheimer’s disease.

As expected, MoCA and SIN scores were moderately correlated, r(36) = -.49, p < .01, with better MoCA scores associated with improved speech reception thresholds. Interestingly, better on-beat rhythm perception was also moderately correlated with better SIN performance, r(36) = -.45, p < .01. To the authors’ best knowledge, the association between on-beat rhythm accuracy and SIN perception has not been previously reported for adults with hearing aids. Frequency- and spectral-psychoacoustic measures were not correlated with SIN.

In conclusion, cognition and on-beat rhythmic ability are associated with better speech-in-noise outcomes for older adults with a moderate hearing loss who use hearing aids. [This research was funded by a Mitacs training grant awarded to the second author that was partially sponsored by Unitron Canada.]

D2-3. Dementia and Hearing-Aid use: A Two-Way Street
Lauren Dillard*, Oliver Zobay, Gabrielle Saunders, Graham Naylor
1Medical University of South Carolina, 2University of Nottingham, Medicine, 3The University of Manchester, Manchester Centre for Audiology and Deafness

Introduction: There are two main causal pathways that may explain associations between reduced hearing aid use and incident dementia. First, hearing aid use may be protective against dementia via delaying diagnosis or reducing rate of cognitive decline. Second, living with dementia may make hearing aid use more challenging, leading to their irregular and/or discontinued use.

Methods: We used longitudinal electronic health records data from 380,794 Veterans aged ≥60 years who obtained hearing aids from the US Veterans Affairs healthcare system. Diagnostic data were ascertained from International Classification of Disease (ICD) codes and audiometric and hearing aid data were available. A metric of hearing aid use persistence was constructed from battery reorder data. Two main analyses were conducted to isolate effects of the pathways described above. The first analysis (n=72,180) included individuals without diagnosis of mild cognitive impairment (MCI) or dementia up to 3.5 years after hearing aid fitting. We used logistic regression models to determine the likelihood of diagnosed incident dementia from 3.5-5.0 years after hearing aid fitting. Models were adjusted for age, hearing acuity, hearing aid use persistence, obesity, stroke, diabetes, depression, bipolar disorder, and hypertension. The second analysis (n=350,918) included a) individuals with prevalent dementia prior to hearing aid fitting (n=6,418), and b) a referent group of individuals with normal cognitive function (no diagnoses of mild cognitive impairment or dementia; n=344,500). We used logistic regression models to determine likelihood of persistent hearing aid use at 3.2 years in individuals with prevalent dementia at the time of hearing aid fitting (vs normal cognitive function). Models were adjusted for age, hearing acuity, multimorbidity index, and previous hearing aid experience. Results are presented as odds ratios (OR) with corresponding 95% confidence intervals (95% CI).

Results: The first analysis showed persistent (vs. non-persistent) hearing aid users had reduced odds of incident dementia (OR 0.73 [95% CI 0.66, 0.81]). The second analysis showed individuals with prevalent dementia prior to hearing aid fitting (vs normal cognitive function) had reduced odds of hearing aid use persistence (OR 0.46 [95% CI 0.43, 0.48]).

Conclusions: Results lend evidence to both pathways described above, suggesting that a) persistent hearing aid use may reduce likelihood of dementia diagnosis, and b) dementia may reduce likelihood of persistent hearing aid use. Study findings highlight the need to prioritize accessibility and usability of hearing devices and hearing care processes for all individuals, regardless of cognitive ability.

12:40 p.m. – 1:30 p.m. Lunch
1:30 p.m. – 5:00 p.m. Leisure
5:00 p.m. – 5:10 p.m. 2024 Technical Committee results announced
SESSION D3. Statistical and Computational Approaches to Aided Outcome Prediction

Session Chair: Sarah Verhulst

D3-1. Blind Modelling of Speech Intelligibility and Listening Effort in Binaural Listening Conditions in Real-Time

Thomas Brand*, Saskia Röttges1, Christopher Hauth1, Jan Rennies2
1Medizinische Physik und Akustik, Universität Oldenburg, Oldenburg, Germany, 2Fraunhofer IDMT, Oldenburg Branch HSA, Oldenburg, Germany

The Binaural Speech Intelligibility Model (BSIM, Beutelmann et al. (2010). JASA, 127) can predict speech intelligibility (SI) as well as listening effort (LE) for spatially separated target speech and interferers. Several extensions have been introduced adapting BSIM to different acoustical conditions. One of these extensions (Rennies et al. (2019), Trends in Hearing, 23) distinguishes between the useful and the detrimental part of the binaural room impulse response (BRIR) and very accurately predicts the interaction of binaural and temporal processing in complex listening conditions with noise, reverberation, and echoes. However, a disadvantage of this BSIM version is that it requires auxiliary information about the speech, the noise, and the BRIR.

In this study, two blind versions of BSIM are presented that only require the mixed speech and noise signals and blindly predict human SI and perceived LE. These model versions base on a blind binaural front-end (Hauth et al. (2020), Trends in Hearing, 24), which assumes that the interfering noise has different modulation characteristics than speech and which predicts the effects of better-ear-listening and binaural unmasking. This binaural front-end applies different processing modes for negative and positive signal-to-noise ratios, and can be combined with different back-ends for estimating human speech recognition. Here, we present results using two blind back-ends that estimate SI and/or LE based on the output of binaural front-end without any further auxiliary information. These blind back-ends are the non-intrusive Short Time Objective Intelligibility measure (NI-STOI) and an automatic triphone recognizer with an a posteriori estimate of the certainty of the recognized triphones. These blind BSIM versions do not perfectly reach the performance of the non-blind reference model but they come very close. Even a short-term version is available which can be applied to arbitrary unknown speech and which updates SI and LE estimates in close to real time. This offers new fields of application, e.g., in hearing instruments. Therefore, the blind real-time BSIM has been implemented on a research hearing aid, where it continuously estimates SI and LE. These estimates can be used, for instance, for informing the selection of hearing aid processing strategies.

The main limitation of our model is that so far it only works for energetic masking and is not yet applicable to speech-to-speech conditions. [ Funded by the Deutsche Forschungsgemeinschaft (DFG, German Research Foundation) – Projektnummer 352015383 – SFB 1330 A1.]

D3-2. Plompz A and D-Model for Predicting the Limited Benefit of Hearing Aids revisited: Modelling Unaided and Aided Speech Recognition with Hybrid Auditory Model- And MI-Based Methods

Birger Kollmeier*, David Hülsmeier1, Anna Warzybok1
This contribution reviews a hybrid approach to model human speech recognition in various noises, for various degree of hearing impairment, and for the unaided and aided condition. The Framework for Auditory Discrimination Experiments (FADE, Schädler et al, 2018) utilizes a physiology-inspired Gabor-Filter feature extraction in combination with an HMM/GMM automatic speech recognition system as backend. This machine-learning-based recognition process implements a “generalized optimum detector approach” to model human speech recognition from basic principles in an unobtrusive and reference-free way. It simulates the detrimental effect of imprecisions in the signal representation that are intended to emulate the effect of the audiogram and of suprathreshold distortions.

Plomp (1978) introduced an empirical separation of the increased speech recognition thresholds (SRT) in listeners with a sensorineural hearing loss into an Attenuation (A) component (which can be compensated by amplification) and a non-compensable Distortion (D) component. Previous own research backed up this notion by speech recognition models that derive their SRT prediction from the individual audiogram with or without a psychoacoustic measure of suprathreshold processing deficits. To determine the precision in separating the A and D component for the individual listener with various individual measures and individualized models, SRTs with 40 listeners with a variation in hearing impairment were obtained in quiet, stationary noise, and fluctuating noise (ICRA 5-250 and babble). Both the clinical audiogram and an adaptive, precise sweep audiogram were obtained as well as tone-in-noise detection thresholds at four frequencies to characterize the individual hearing impairment. The prediction results indicate that the precisely measured swept tone audiogram allows for a more precise prediction of the individual SRT in comparison to the clinical audiogram (RMS error of 4.3 dB vs. 6.4 dB, respectively). The further refinement of including the tone-in-noise detection threshold with FADE led to a slight improvement of prediction accuracy (RMS error of 3.3 dB).

Hence, applying FADE is advantageous for scientific purposes where a consistent modelling of different psychoacoustic effects in the same listener with a minimum amount of assumptions and an aided performance prediction is desirable. For clinical purposes, however, an estimation of the expected D component using a linear regression appears to be a satisfactory first step towards precision audiology. Individually assessing the D-component does not only help to decrease the previously “unexplained” variability in SRT across listeners with the same audiogram. It also helps to set realistic expectations for the benefit from any hearing aid.

D3-3. Predicting Aided Intelligibility in Noisy Rooms: Extending the Hearing Aid Speech Perception Index (HASPI) to Binaural Listening

James Kates*, Kathryn Arehart†, Ramesh Muralimanohar‡, Mathieu Lavandier§
†The University of Colorado Boulder, ‡University of Lyon ENTPE

The Hearing Aid Speech Perception Index (HASPI) is a speech intelligibility metric that provides accurate predictions for speech degraded by additive noise, reverberation, spectral changes, and nonlinear distortion. The metric also predicts the effects of hearing-aid signal processing on speech intelligibility by virtue of incorporating a model of the auditory periphery that can represent impaired as well as normal hearing. HASPI is an intrusive metric that compares a degraded signal to a clean reference, and the current version of HASPI has been derived for monaural stimulus presentation over headphones. Here we extend HASPI to predict speech intelligibility for binaural listening.
The listening conditions comprise intelligibility scores for IEEE sentences obtained from both normal-hearing (NH) and hearing-impaired (HI) listeners. The binaural test conditions use simulated room acoustics combined with head-related impulse responses measured on a KE-MAR manikin. Symmetric and asymmetric spatial configurations of the speech and noise sources are presented for an anechoic room and for a concert hall. Speech is combined with multi-talker babble at three different SNRs, and the noisy reverberant speech is processed through three different simulated hearing aid settings: linear amplification, syllabic wide dynamic-range compression (WDRC) combined with mild noise suppression, and syllabic WDRC combined with strong noise suppression and frequency compression.

The proposed binaural HASPI first processes the reference signal and degraded signals through independent models of the auditory periphery. The reference signal for the binaural HASPI is the average of the anechoic speech at the left and right ears for a frontal source. The reference signal is passed through a model of the NH periphery while the left and right degraded signals use an HI periphery. Binaural interaction is represented by adjusting the time delays and amplitudes of the left and right degraded signals at the output of each peripheral auditory band to maximize the cross-correlation of their sum with the reference peripheral output in that band. The HASPI calculation then proceeds as for the monaural metric, using the delayed and weighted sum of the left and right degraded signals compared to the anechoic reference. We will discuss the overall accuracy of the binaural HASPI and how the binaural metric compares to the monaural metric computed for better ear listening. [Work supported by a research grant from GN ReSound to the University of Colorado at Boulder.]

D3-4. Latent factor analysis of hearing aid outcomes using multiple-indicator multiple-cause model

David Allen¹, Melanie Ferguson², Deanna Conner³, Jessica Cooper¹, Catherine McMahon⁴, Jorge Mejia¹, Jessica Monaghan¹, Ifeyinwa Okonkwo¹, Jermy Pang², Finnian Sonter², Raaya Tiko¹, Padraig Kitterick*¹

¹National Acoustic Laboratories, Sydney Australia, ²Faculty of Health Sciences, Curtin University/Ear Science Institute Australia, Perth, Australia, ³Hearing Australia, Sydney Australia, ⁴Macquarie University, Sydney Australia

Two key challenges when predicting outcomes after hearing aid fitting are that a multitude of factors can influence outcomes, and that benefit from hearing aids is a multi-dimensional construct. Measuring benefit also typically involves the use of multiple patient-reported outcome measures that are themselves often multi-dimensional, not designed to measure independent constructs, and are often highly correlated with each other. Dimension reduction approaches such as factor analysis can be useful in examining latent factors underpinning the pattern of scores across such measures. We demonstrate the application of multiple indicator multiple cause (MIMIC) modelling as a framework for studying the complex relationships between a range of predictive factors and outcomes following hearing aid fitting.

A total of 1965 adult participants were recruited from routine clients of Hearing Australia clinics across Australia. Of those, 1838 completed patient reported outcome measures prior to hearing aid fitting. Of the 819 clients who were fit with hearing aids, 593 completed a repeat assessment at 2 weeks post fitting, and 553 completed repeat assessment at 8 weeks. The predictive factors assessed included audiometric thresholds, readiness for hearing aids, auditory fatigue, complexity of auditory listening environments, hearing disability and handicap, social isolation, mental well-being, cognitive function. Outcomes assessed following hearing aid fitting included measures selected as being related to device experience (change in hearing disability, satisfaction with hearing aids, use of hearing aids, and benefits due to
hearing aids) and psychosocial benefit (changes in hearing utility and health utility, mental well-being, auditory fatigue, and social isolation).

An exploratory factor analysis identified that the outcome measures loaded onto two factors in line with the conceptualisation of benefit as comprising device experience and psychosocial-related effects. The MIMIC model was fit to the available data and used to examine the relationships between predictive factors and benefit scores. Participants experiencing lower social isolation and who listened in more complex auditory environments prior to hearing aid fitting, and those who had higher readiness for hearing aids, had higher device experience benefit scores. Greater psychosocial benefit was observed in those with greater hearing disability, more auditory fatigue, or better hearing in the better ear prior to hearing aid fitting.

MIMIC models, as illustrated in this study, provide a methodological framework that could be applied to the challenges of understanding variability in outcomes, predicting how individuals may benefit, and of identifying those who are most likely to benefit from hearing aids.

6:30 p.m. - 7:30 p.m.  Dinner
7:30 p.m. – 10:00 p.m.  Posters and Social

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Safe Travels. See you in 2024!
# IHAS/IHCON 2022 Attendees

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