IHCON 2014

International Hearing Aid Research Conference

August 13 – 17, 2014

Granlibakken Conference Center
Tahoe City, California
IHCON 2014

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Conference Coordinator: Barbara Serrano
# Student Scholarship Recipients

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<th>Affiliation</th>
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<tr>
<td>Inge Brons$^1$</td>
<td>Academic Medical Center</td>
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<td>Stephanie Cheung$^1$</td>
<td>McMaster University</td>
<td>Canada</td>
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<td>Cliston Cole$^1$</td>
<td>University of Illinois at Urbana-Champaign</td>
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<tr>
<td>Evelyn Davies-Venn$^1$</td>
<td>University of Minnesota</td>
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<td>Timothy Davis$^1$</td>
<td>Vanderbilt University</td>
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<tr>
<td>Ann-Marie Dickinson$^2$</td>
<td>The University of Manchester</td>
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<td>Julia Habicht$^1$</td>
<td>University of Oldenburg</td>
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<td>Hamish Innes-Brown$^1$</td>
<td>Bionics Institute</td>
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<td>Abigail Kressner$^1$</td>
<td>Georgia Institute of Technology</td>
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<td>Sara M. K. Madsen$^2$</td>
<td>University of Cambridge</td>
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<td>Nazanin Pourmand$^1$</td>
<td>University of Western Ontario</td>
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<tr>
<td>Tobias Weller$^1$</td>
<td>Macquarie University</td>
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<tr>
<td>Alan Wiinberg$^1$</td>
<td>Technical University of Denmark</td>
<td>Denmark</td>
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Student Scholarship funding is from the following sources:

$^1$ NIDCD  
$^2$ Action on Hearing Loss
Daily Schedule

Wednesday, August 13, 2014
5:00 PM Welcome Social
6:00 PM Dinner
7:30 PM Welcome Remarks
7:45 PM Keynote Address
8:30 PM Discussion
9:00 PM Social

Friday, August 15, 2014
9:45 AM Poster Session
11:15 AM Morning Session B
12:15 PM Lunch
5:00 PM Evening Session
7:00 PM Dinner
8:30 PM Social

Thursday, August 14, 2014
7:00 AM Breakfast
8:00 AM Morning Session A
10:00 AM Poster Session
11:30 AM Morning Session B
12:30 PM Lunch
5:00 PM Evening Session
7:00 PM Dinner
8:30 PM Social

Friday, August 15, 2014 (cont’d)
9:45 AM Poster Session
11:15 AM Morning Session B
12:15 PM Lunch
5:00 PM Evening Session
7:00 PM Dinner
8:30 PM Social

Saturday, August 16, 2014
7:00 AM Breakfast
8:00 AM Morning Session A
10:00 AM Poster Session
11:30 AM Morning Session B
12:30 PM Lunch
5:00 PM Evening Session
7:00 PM Dinner
8:30 PM Social

Sunday, August 17, 2014
7:00 AM Breakfast
8:00 AM Conference Concludes (check-out following breakfast)
Program Summary

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Wednesday, August 13

Welcome and Keynote Address
7:30 PM – 8:30 PM

Welcome Remarks: Sig Soli, Peter Blamey

Keynote Address

Stefan Launer
Hearing instrument technology: Which hearing problems are we not addressing yet?
Thursday, August 14

Session One:
Individual Factors and Hearing Aid Signal Processing
8:00 AM – 10:00 AM
Moderator: Brian C. J. Moore

Andrea Pittman
Hearing aid signal processing for children with hearing loss (Invited)

Pamela Souza
Individual sensitivity to distortion from combined signal processing

Dianne Van Tasell
User self-adjustment of a simulated hearing aid using a mobile device

Tobias Neher
Towards more individualized noise reduction and directional processing in hearing aids

Poster Session I 10:00 AM – 11:30 AM

Session Two:
Bone Conduction Devices
11:30 AM – 12:30 PM
Moderator: Sunil Puria

William Hodgetts
Prescription and verification of bone conduction implants: Present and future considerations (Invited)

Stefan Stenfelt
Binaural hearing abilities in normal hearing subjects with and without cross ear sound transmission
Session Three:
Hearing Loss and Cognition / Plasticity / Degeneration in Adults
5:00 PM – 7:00 PM
Moderator: Peggy Nelson

Anu Sharma  Changes in cortical resource allocation in age-related hearing loss (*Invited*)

Laurel H. Carney  A speech enhancement strategy based on midbrain response properties

Nirmal Srinivasan  Release from masking for small spatial separations in older listeners

Hamish Innes-Brown  The relationships between brainstem responses to complex sounds, perceptual sensitivity to temporal fine structure, and understanding speech in noise (*Scholarship*)
Friday, August 115

Session Four:
Stuart Gatehouse Lecture / Hearing in Noise
8:00 AM – 9:45 AM
Moderator: Michael Akeroyd

Graham Naylor
Audiological rehabilitation is about unique human beings with real lives: trends in the examination of individual differences and auditory ecology (*Gatehouse Lecture*)

William Whitmer
On a meaningful increase in signal-to-noise ratio

DeLiang Wang
A classification algorithm to improve speech recognition in noise for hearing-impaired listeners

Poster Session II 9:45 AM – 11:15 AM

Session Five:
Real World Challenges of Training and Service Delivery
11:15 AM – 12:15 PM
Moderator: Louise Hickson

Gabrielle Saunders
Health behavior change in adults with hearing impairment

Wouter Dreschler
A profiling system for the selection of hearing aids
# Session Six:
**Noise as a Cause and Challenge for People with Hearing Loss**

5:00 PM – 7:00 PM

Moderator: Todd Ricketts

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<td>Development of a therapeutic to protect the inner ear: From animal models to human trials <em>(Invited)</em></td>
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<td>Jelmer van Schoonhoven</td>
<td>Predicting the speech intelligibility in fluctuating and reverberant background noise using the Speech Transmission Index</td>
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<td>Joshua Bernstein</td>
<td>Spectrotemporal modulation sensitivity as a predictor of speech intelligibility in noise with hearing aids</td>
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<td>Gitte Keidser</td>
<td>Is a non-auditory profile associated with bilateral directional processing benefit in challenging listening situations?</td>
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Saturday, August 16

**Session Seven:**
**Objective Modelling of Hearing Aid Function**
8:00 AM – 10:00 AM
Moderator: Tim Trine

- **Birger Kollmeier**
  Understanding contemporary hearing aid algorithms and their relation to sensory factors and human performance (*Invited*)

- **James Kates**
  A unified approach to predicting speech intelligibility and quality

- **Inge Brons**
  Perceptual evaluation of noise reduction in hearing aids (*Scholarship*)

- **Abigail Kressner**
  The influence of structure in binary mask estimation error on speech intelligibility (*Scholarship*)

**Poster Session III** 10:00 AM – 11:30 AM

**Session Eight:**
**Bimodal Hearing for Children and Adults**
11:30 AM – 12:30 PM
Moderator: Jan Wouters

- **Lisa Davidson**
  Considerations in fitting and evaluating bimodal devices for pediatric CI recipients (*Invited*)

- **Lidwien C.E. Veugen**
  Frequency dependent loudness balancing and AGC matching to optimize hearing aid benefits in bimodal CI users
Session Nine:
Listening and Technologies in Real World Environments
5:00 PM – 7:00 PM
Moderator: Ben Hornsby

Justin Zakis  Robust wind noise detection
Marc Brennan  Optimizing forward-masked thresholds with amplification
Martijn Agterberg  Contribution of monaural and binaural cues to sound localization in listeners with unilateral conductive hearing loss (UCHL) and in total single sided deafness (SSD)
Sara M. K. Madsen  The effect of amplitude compression on the ability to hear out individual lines in music (Scholarship)

Sunday, August 17

CLOSE OF CONFERENCE

Check-out following breakfast
Hearing instrument technology: Which hearing problems are we not addressing yet?

Stefan Launer
Phonak AG, Switzerland

Basic physiological and psychoacoustic research two decades ago fundamentally altered our understanding of the underlying mechanisms of normal hearing and, even more so, impaired hearing. A major focus of research was on the physiological and psychoacoustical understanding of the mechanics of the inner ear in the healthy and impaired ear. In particular the detailed role of inner and outer hair cell mechanics received a lot of attention and served as a basis for explaining a lot of perceptual findings in both normal hearing and hearing impaired subjects. This research inspired and drove the conception and development of many new signal processing algorithms for hearing instruments. The introduction and fast development of microelectronics into hearing instrument technology allowed us to introduce a large variety of these different features aiming at improving speech intelligibility and hearing comfort in a variety and broad range of different listening conditions. In order to simplify the selection of optimal settings, modern hearing instruments use a function which allows them to automatically identify an acoustic scene and then to automatically select the optimal signal processing strategy for that scene.

In this talk I want to discuss research needs at different levels of auditory processing that could potentially help us to further improve auditory rehabilitation with hearing instruments.

First of all, we have a fairly good understanding of inner ear mechanics and the impact of hearing loss on it. However, recent research by Kujawa and Libermann has shown that this might fall short by not fully appreciating the impact of neural degeneration on auditory perception. Very little work to date exists trying to shed light on the potential impact of such a form of a “hidden” hearing loss on auditory perception in easy or challenging listening conditions. With the common measure of audibility for quantifying hearing impairment we might miss out on detecting such a form of a hearing loss. Research into a) understanding the perceptual consequences of hidden hearing loss and b) how to measure hidden hearing loss could help develop new solutions for people with various degrees of hearing losses including those with milder losses.

Secondly, it is remarkable how quickly the auditory system is able to identify and classify a sound into either being speech, music or environmental sounds. A better understanding of the underlying mechanisms could help us to improve the performance of hearing instruments for a) speech intelligibility in reverberant environments and b) identifying acoustic environments. One of the most remarkable abilities of the auditory system is the segregation of a mixture of sounds into different auditory objects in a variety of listening conditions including reverberant conditions. Today’s hearing instruments are capable of
improving speech intelligibility in many situations including those with multiple interfering sources. However, performance significantly breaks down once the environment becomes reverberant. The auditory system has a remarkable ability to also perform well even in highly dynamic and reverberant situations. Which features does it rely on that are robust against the impact of the transmission path? Understanding how the auditory system performs this function could inform the design of hearing aid processing for people with hearing loss who suffer in these environments.

Finally, auditory perception extends beyond the pure intelligibility of speech in various listening conditions. In daily communication situations, speech sounds do not only convey information about the content but also include information about the speaker’s emotional state or intent. Evidence suggests that people with hearing loss have a poorer ability to identify this information. How hearing loss affects identification of emotion and intent, and the way in which hearing aids affect the reception of this information, is important to understand.

Hearing instrument technology has improved over the past two decades and significantly improves the lives of hearing impaired people. The objective of this talk is to present and discuss the author’s perspective on various topics of potential research in the field of impaired auditory perception and communication which might help to further improve the quality of life of hearing impaired persons.

Thursday, August 14

SESSION ONE

Individual Factors and Hearing Aid Signal Processing

Moderator: Brian Moore

8:00 AM  Hearing aid signal processing for children with hearing loss

Andrea Pittman
Arizona State University

There is evidence that the listening and learning strategies of children with hearing loss compete with one another such that they are not able to detect and learn new information as well as their age-matched or hearing-matched peers. This competition may be responsible, in part, for the 2-year difference in vocabulary development between children with mild-to-moderate hearing losses and children with normal hearing. Our work focuses on understanding the listening and learning strategies of these children in an effort to help them make the most of every opportunity to learn new information. Results from several recent studies will be discussed with particular attention to the unique effects of hearing loss in childhood.

The presentation will also include a discussion of work funded by a grant from the Hearing Industry Research Consortium (IRC) entitled: Assessing Advanced Hearing Aid Features Using Behavioral Tasks that Vary in Cognitive Demand. In this two-year project, three forms of signal processing (wide-band amplification, frequency lowering, and digital noise reduction) are being examined to determine if hearing aid users derive increas-
Listening benefits from these features as the demands of the listening tasks increase. To date, 10 children and 7 adults with normal hearing have been enrolled as well as 11 children and 15 adults with hearing loss. Individuals with hearing loss participate in an unaided test session and in at least two aided test sessions depending on the configuration of the hearing loss. Specifically, listeners with steeply sloping high-frequency losses are tested with frequency-lowering technology whereas listeners with moderate high-frequency losses are tested with extended bandwidth technology. All listeners are tested with digital noise reduction. Preliminary results for two of the four auditory tasks will be described: 1) Non-Word Detection in which the listener counts the number of nonsense words embedded into four-word sentences, and 2) Auditory Lexical Decision in which the listeners repeat real and nonsense words and then judge the lexicality of each word (i.e., real or not real). Stimuli for all tasks were produced by the same female talker and presented in the sound field at 65 dB SPL in quiet and in steady-state noise at a +3 dB signal-to-noise ratio. Although data collection is ongoing, preliminary results suggest that, compared to adults with hearing loss, children with hearing loss have more difficulty detecting words that are unfamiliar to them, particularly in noise. Also, the errors that children with hearing loss make when judging the lexicality of words suggests a higher level of uncertainty and a stronger influence of context compared to the errors of adults with hearing loss. Finally, performance on both tasks improved with amplification in quiet but less so in noise. Clearer improvements in performance were observed with the use of wide-band amplification whereas lesser benefits were observed for digital noise reduction and frequency lowering.

Individual sensitivity to distortion from combined signal processing

Pamela Souza¹, Kathryn Arehart², Jing Shen¹, James Kates² and Ramesh Muralimanohar²

¹ Northwestern University, Evanston, IL
² University of Colorado at Boulder, Boulder, CO

Hearing aids aim to improve speech intelligibility via a combination of signal processing techniques, including multichannel wide-dynamic range compression [WDRC], digital noise reduction and frequency lowering. Although each strategy can enhance usable speech cues, each can also distort. While objective metrics can quantify the acoustic trade-off between distortion and audibility, susceptibility to distortion is also related to individual peripheral and cognitive abilities (Kates et al., J. Acoust. Soc. Am., 2013). However, previous data were based on listener response to a single processing strategy and may not reflect effects of wearable aids, where the signal-processing chain may result in aggregate or decremental distortion compared to single-processing approaches. This study explores the cumulative effect of combined distortions; specifically, the combination of fast-acting WDRC (which primarily alters temporal cues) with frequency lowering (which primarily alters spectral cues). Study participants are older adults with mild-to-moderate sensorineural hearing loss. Intelligibility scores are measured for low-context sentences presented in modulated background noise at a range of signal-to-noise ratios. Simulated amplification processing is used to vary the level of distortion introduced by WDRC and frequency lowering, separately and in combination. Acoustic distortion is quantified using a model of the auditory periphery that measures the fidelity with which the speech envelope and temporal fine structure are maintained after processing. We explore the relationship between cumulative distortion and individual peripheral and cognitive factors, including working memory, processing speed and executive function. Results to date indicate that when signal-processing approaches are combined, one approach
may dominate with regard to both distortion and speech perception, with smaller contributions from the secondary approach. Response to processing distortion is related to working memory as well as to processing speed. These results suggest that accurate predictions of processing benefit can be achieved by combining a distortion metric that measures the effects of the entire signal-processing chain in the hearing aid with measurements of individual cognitive function. [Work supported by NIH.]

9:00 AM  

**User self-adjustment of a simulated hearing aid using a mobile device**

_Dianne J. Van Tasell_1,2, Andrew T. Sabin2 and Kevin H. Franck2

1 University of Minnesota, Minneapolis, MN  
2 Ear Machine LLC, Chicago, IL

The work reported here addressed the question of whether elements of current hearing aid fitting methods could be replaced by intuitive self-fit controllers that allow users to adjust their own hearing aid dsp parameters using a mobile device. Current methods for fitting hearing aids comprise: 1) a pure-tone audiogram that is used to obtain 2) an initial prescriptive fit-to-target, followed by 3) further adjustment of signal processing parameters to address the user’s stated preferences for sound quality. Users’ abilities to adjust hearing aid signal processing parameters themselves in a variety of sound environments – and thereby to bypass audiogram, fitting software and verification steps – were investigated. A simple controller interface was implemented on an iPod Touch running a real-time signal processing simulation of a 9-channel WDRC hearing aid. The user manipulated two wheels: one wheel adjusted gain, compression, and MPO in all 9 channels simultaneously according to an algorithm based on demographic hearing loss data; the other wheel adjusted frequency response according to an algorithm based on analysis of common fine-tuning adjustments. Forty-eight subjects with sensorineural hearing loss, aged 27 – 94, adjusted the controllers for each of six speech passages (in quiet or noise) and three music passages. We have reported previously that on average subjects were reliable in their adjustments, chose gain appropriate to their hearing losses, and consistently preferred (in blind A/B comparisons) their own settings to those prescribed by NAL-NL2. We will focus here on the systematic differences among subjects’ self-selected settings and their preferences when: a) speech passages were heard at different SNRs; b) the passage was music vs. speech; c) the subject’s audiogram was or was not used to set initial controller positions; and d) visual cues were added to the controller interface. [Work supported by NIDCD grant R44 DC013093.]

9:30 AM  

**Towards more individualized noise reduction and directional processing in hearing aids**

_Tobias Neher_1, Kirsten Wagener2 and Markus Meis2

1 Medical Physics and Cluster of Excellence “Hearing4all”, Oldenburg University, Germany  
2 Hörzentrum Oldenburg, Germany

Recently, there has been growing interest in the personalization of medical interventions. As part of the Cluster of Excellence “Hearing4all”, we are investigating ways of tailoring hearing aid (HA) technology better to the needs of individual users (e.g. Neher et al, 2013, 2014). In the current study, we explored potential avenues for individualizing noise reduction and directional processing. We recruited 120 participants aged 60-85 years and screened them for a number of sensory and neuropsychological impairments. In addition, we administered a measure of reading span after Daneman and Carpenter (1980) to them.
For further testing, we selected 60 participants whom we could stratify into groups that were closely matched in terms of age, reading span (large vs. small), and pure-tone average hearing loss (mild vs. moderate). All participants were experienced HA users with symmetrical sensorineural hearing losses, good visual acuity, and no neuropsychological deficits.

Using laboratory simulations of a busy cafeteria situation, we then performed speech-in-noise measurements under various aided conditions. These were based on monaural and binaural algorithms for noise reduction and directional processing. In terms of outcome measures, we assessed speech intelligibility and overall preference at a number of fixed signal-to-noise ratios. Furthermore, we administered additional (visual) measures of cognitive function, noise sensitivity, and personality to our participants to be able to also assess the influence of these characteristics on our participants’ response to the various processing conditions.

We will discuss our results with a view towards developing more effective fitting strategies that ultimately may lead to greater satisfaction with HAs. [Funded by the DFG Cluster of Excellence EXC 1077/1 “Hearing4all”.

References:

Thursday, August 14

SESSION TWO

Bone Conduction Devices

Moderator: Sunil Puria

11:30 AM Prescription and verification of bone conduction implants: Present and future considerations

Bill Hodgetts
Associate Professor, Department of Communication Sciences and Disorders
University of Alberta
Program Director, Bone Conduction Amplification
Institute for Reconstructive Sciences in Medicine

The last 10 years has seen a massive expansion in the field of bone conduction implants (BCIs). From a time when there was only one small company offering technology to a small group of patients, clinicians and researchers, there are now many companies and many new clinicians and researchers helping patients with ever expanding candidacy criteria. However, unlike air conduction hearing aids, there is still a relatively underdeveloped literature with respect to how these bone conduction implants are to be verified and
prescribed. In this presentation I will review the challenges to prescription and verification for percutaneous devices (e.g., Baha and Ponto devices.) I will also review our approach to developing and implementing a BCI prescription (based on DSL V5) with the use of a skull simulator for verification. As we look to the future, there are now BCI technologies that have the active vibrator implanted under the skin (e.g., Bonebridge, Hakasson’s BCI.) I will explore the challenges and some of the solutions under consideration for verifying and prescribing output for these new devices.

12:00 PM  Binaural hearing abilities in normal hearing subjects with and without cross ear sound transmission

Stefan Stenfelt and Mehrnaz Zeitooni
Department of Clinical and Experimental Medicine
Linköping University, Linköping Sweden

Binaural hearing relies on the separation of the inputs to the two ears. In normal air conduction (AC) hearing, interaural differences in time and level (ITDs and ILDs) can be used by the auditory system to separate sound sources and provide benefit, for example in noisy situations. When the stimulation is provided by bone conduction (BC), as for example in bone conduction hearing aids (BCHA) or headsets using BC transmission, as sound from one stimulation position reaches both cochleae it is believed that the efficiency of binaural hearing can be impeded or even eradicated. Accordingly, due to this cross ear sound transmission, it is often argued that binaural hearing is not possible with BC stimulation and BCHAs are therefore most often fitted monaurally.

Binaural hearing was investigated in 30 participants with normal hearing as measured by audiometric thresholds. The binaural hearing ability was investigated for three conditions: (1) AC stimulation (earphones), (2) BC stimulation at the mastoid close to the ear canal, and (3) BC stimulation at the BCHA position (approximately 55 mm behind the ear canal opening). The interaural cross transmission was estimated for the two latter conditions. Spatial releases from masking, binaural intelligibility level difference, binaural masking level difference, precedence effect, and JNDs for ILDs as well as for ITDs were estimated for the three conditions.

The results showed that binaural processing exists in all three conditions, but the results are better in the AC condition than in the two BC conditions. When the results were compared between the two BC conditions, the mastoid position (closer to the ear canal) showed slightly better performance than the BCHA position. The BCHA position also showed the greatest cross ear transmission. The tests were further evaluated in three subjects with unilateral deafness. That evaluation showed that it was not binaural interaction at the cochlea (monaural cues) that caused the result for the BC stimulation but true binaural cues. The tests of ITDs and ILDs indicated that ITDs were elevated for BC stimulation compared with AC stimulation, but ILDs were comparable for all three conditions. Consequently, ILDs seem to be conserved with BC stimulation while the ITDs are impaired. Moreover, the results favor bilateral fitting of BCHAs, at least for persons with symmetrical and near normal cochleae.
SESSION THREE

Hearing Loss and Cognition / Plasticity / Degeneration in Adults
Moderator: Peggy Nelson

5:00 PM  Changes in cortical resource allocation in age-related hearing loss

Anu Sharma, Julia Campbell and Garrett Cardon
Brain and Behavior Laboratory
University of Colorado at Boulder, Boulder, CO

A basic tenet of neuroplasticity is that the brain will re-organize following sensory deprivation. Sensory deprivation appears to tax the brain by changing its normal resource allocation. A better understanding of the cortical re-organization that accompanies hearing loss may allow us to incorporate improvements in the design of prostheses to allow them to better accommodate altered cortical processing. Compensation for the deleterious effects of hearing loss may include recruitment of alternative or additional brain networks to perform auditory tasks. Our high-density EEG experiments suggest that age-related hearing loss results in significant changes in neural resource allocation, reflecting patterns of increased listening effort, decreased cognitive reserve, and/or changes in social emotional status which may be associated with dementia-related cognitive decline. Cross-modal plasticity is another form of cortical re-organization associated with deafness. Cross-modal plasticity occurs when an intact sensory modality recruits cortical resources from a deprived sensory modality to increase its processing capabilities as compensation for the effects of sensory deprivation. Deaf animals show recruitment of higher-order auditory cortical areas, by visual and somatosensory modalities resulting in enhanced capabilities for the recruiting modality, with likely negative consequences for auditory perception. Our results in humans suggest clear evidence of recruitment of higher-order auditory cortical areas by visual and somatosensory modalities in age-related hearing loss. Cross-modal cortical re-organization is evident in early stages of hearing loss and shows a strong negative correlation with speech perception in noise suggesting that recruitment by other sensory modalities may influence the variability in outcomes in adults with hearing loss, including those who are treated with amplification. Overall, our results suggest that compensatory cortical plasticity secondary to sensory deprivation has important neurological consequences and influences outcomes in age-related hearing loss. [Supported by NIH NIDCD.]

5:30 PM  A speech enhancement strategy based on midbrain response properties

Laurel H. Carney and Douglas M. Schwarz
Departments of Biomedical Engineering and Neurobiology & Anatomy
University of Rochester, Rochester NY

Speech enhancement algorithms have typically focused on compensating for changes due to hearing loss in the response of the auditory periphery. Significant transformations in neural representations occur at the level of the midbrain, which is the first point along the auditory pathway where neurons are tuned to amplitude-modulation frequency. We are developing and testing a strategy for speech enhancement that attempts to restore the midbrain-level representations, which are robust across a wide range of sound levels and
in the presence of noise in the healthy auditory system. This approach requires restoring the contrast in temporal fluctuations across frequency channels in the periphery. For example, peripheral responses to vowels are dominated by fluctuations near the fundamental frequency (F0, or voice pitch), but the amplitudes of the fluctuations vary across frequency channels in the healthy ear due to cochlear and synaptic nonlinearities, such as saturation and synchrony capture. In particular, peripheral frequency channels near formants have sustained responses that are dominated by a single harmonic and may be saturated, thus these channels have smaller F0-related fluctuations than the channels that are between formants. In the impaired ear, both elevated thresholds and reduced nonlinear cochlear amplification would diminish the contrast across channels. Our algorithm attempts to restore the contrast in peripheral fluctuations, and thus restore midbrain-level representations, by purposely saturating peripheral frequency channels near formants. Streamlined models for peripheral and midbrain processing are used to identify harmonic and formant frequencies in voiced speech. The peripheral model includes a bank of filters and saturating nonlinearities. The envelope of each frequency channel response is then band-pass filtered around F0, representing modulation tuning in the midbrain. Formants are identified based on the responses across frequency channels of the periphery and midbrain models. The harmonic closest to each formant is then amplified in an attempt to restore the contrast in fluctuations that would be present in the healthy periphery. Preliminary results of intelligibility tests in noise for listeners with and without hearing loss will be presented. Improvements from a preliminary version of this approach (Rao and Carney, 2014, IEEE TBME) will be described. [Supported by NIH-NIDCD DC010813.]

6:00 PM Release from masking for small spatial separations in older listeners

Nirmal Kumar Srinivasan\textsuperscript{1,2}, Frederick J. Gallun\textsuperscript{1,2}, Sean D. Kampel\textsuperscript{1}, Kasey M. Jakien\textsuperscript{1,2}, Samuel Gordon\textsuperscript{1} and Megan Stansell\textsuperscript{1}

\textsuperscript{1} National Center for Rehabilitative Auditory Research, Portland, OR

\textsuperscript{2} Dept. of Otolaryngology, Oregon Health & Science University, Portland, OR

It is well documented that spatial separation between target and interfering maskers results in significant release from masking in younger and older listeners. There have been fewer studies looking at the effect of small spatial separations between target speech and masker speech on release from masking. This study investigates the effect of very small spatial separations on spatial release from masking in older listeners. Virtual acoustic techniques as explained in Zahorik, 2009 were used to simulate two listening environments (reverberation times (T\text{60}) of 0 ms (anechoic) and 500 ms (reverberant)). Spatial release from masking was tested using stimuli drawn from Coordinate Response Measure (CRM; Bolia et al., 2000) speech corpus. All listeners were tested at 10 dB sensation level (dB SL) compared to their audiometric speech reception threshold. This low dB SL was used so that individuals with high amounts of hearing loss can be tested. Identification thresholds in terms of target-to-masker ratio (TMR) were obtained using a ‘progressive’ tracking procedure (Gallun et al., 2013) which involved presentation of 20 trials at 10 TMRs. Release from masking was measured by comparing threshold TMRs obtained with a target sentence presented from directly ahead of the listener and two masking sentences presented in one of the six spatial configurations: colocated with target (0°) or symmetrically separated by 2°, 4°, 6°, 8°, 10°, 15°, or 30°. Initial data analyses indicate lower TMRs at threshold for higher spatial separations (15° or 30°) and anechoic conditions. Also, there was no difference in TMRs at threshold for reverberant condition for very small spatial separations (2°, 4°, 6°, 8° or 10°) as compared to colocated condition. Best fitting filter functions based on masked results would be computed and compared in
conjunction with the earlier suggested filters (Marrone et al., 2008). [Work supported by NIH R01 DC011828.]

6:30 PM

The relationships between brainstem responses to complex sounds, perceptual sensitivity to temporal fine structure, and understanding speech in noise

Hamish Innes-Brown¹, Jason Gavrilis², Jeremy Marozeau¹ and Colette McKay¹
¹ Bionics Institute
² Department of Audiology and Speech Pathology, University of Melbourne

Aims: People with impaired hearing often have difficulties in hearing sounds in a noisy background or in segregating a single voice among many. Both these skills rely partly on the ability of the auditory system to process temporal fine-structure (TFS) information in the sound signal. Perceptual sensitivity to TFS cues can be measured behaviourally; however, it is unclear whether TFS sensitivity is also reflected in the complex auditory brainstem response. In this study we examined the relationships between perceptual sensitivity to TFS cues, brainstem encoding of TFS cues, and the ability to understand speech in noise. Fully understanding these links will allow the development of an objective measure of TFS encoding that could be used to detect changes in hearing before the onset of permanent threshold shifts, as well as measures of above-threshold hearing acuity that could be used to optimise hearing aid fitting.

Methods: We measured TFS sensitivity, a rapid test of speech in noise performance (QuickSIN) and self-reported speech and sound-quality measures (Speech, Spatial and Qualities of hearing – SSQ) behaviourally. We recorded brainstem responses to complex harmonic and inharmonic sounds as well as filtered versions that only contained unresolved harmonics. We performed cross-correlations between the stimulus waveforms and brainstem responses to generate a simple measure of brainstem encoding, and correlated these results with the behavioural measures of TFS sensitivity and speech-in-noise performance.

Results: The ability of the brainstem to encode the complex sounds accurately was significantly correlated with the TFS sensitivity measured behaviourally. In addition, trends were observed in the correlation between responses to the filtered ‘TFS’ sounds and self-reported speech and quality subscales of the SSQ.

Conclusions: Developing an objective measure of the ability of the auditory system to encode TFS is important for optimising the benefits that hearing devices can provide, especially for infants or adults who cannot respond behaviourally. With further development, similar measures could also be used to detect changes to functional hearing due to neuronal degredation before the onset of permanent threshold shifts.

Acknowledgements: Supported provided by the National Health and Medical Research Council of Australia. The Bionics Institute acknowledges the support it receives from the Victorian Government through its Operational Infrastructure Support Program.
8:00 AM  
Audiological rehabilitation is about unique human beings with real lives: trends in the examination of individual differences and auditory ecology  

Graham Naylor  
Eriksholm Research Centre, Oticon A/S, Snekersten, Denmark  

Stuart Gatehouse felt strongly that a true picture of the needs of people with hearing impairment, and of the value of attempts to fulfil their needs, requires that we look beyond averages, and regard variation as a source of information and opportunities.  

This talk will consider two areas in which Stuart carried out influential work ‘beyond averages’.  

Auditory Ecology is concerned with the range, types and importance of the communication environments within which people expect or are required to function.  

Individual Differences (or Candidature) is concerned with understanding how personal factors (psychoacoustical, cognitive, ecological, attitudinal, etc.) may affect the optimal choice of intervention to alleviate hearing problems.  

Stuart's work in these areas will be summarized, and some of the progress which has been made since will be described.  

8:45 AM  
On a meaningful increase in signal-to-noise ratio  

David McShefferty, William M. Whitmer and Michael A. Akeroyd  
MRC/CSO Institute of Hearing Research - Scottish Section  

How large does a change in speech in noise need to be before for it to be meaningful to someone? We here attempt to answer this question using objective and subjective methods. First, we measured the just noticeable difference (JND) in signal-to-noise ratio (SNR) to find the lower limits of perceptually relevant features (e.g., noise reduction). Second, we measured the minimum SNR change necessary to spur someone to seek out the intervention using different subjective-comparison paradigms.  

To measure an SNR JND, we used a variation on the classic level discrimination paradigm using equalised sentences in same-spectrum noise with various controls and roves to ensure that the task could only done by listening to SNR, not level per se. Averaged across participants, the SNR JND was 3 dB. This value was corroborated using different participants in a fixed-level task. JNDs were not correlated with hearing ability. To measure the subjective import of an increase in SNR, we presented paired examples of speech and noise: one at a reference SNR and the other at a variably higher SNR. In different experiments, participants were asked (1) to rate how many successive conversations they would tolerate given each example, (2) to rate the ease of listening of each example, (3) if they would be willing to go to the clinic for the given increase in SNR, and (4) if they would swap the reference SNR for the better SNR example (e.g., their current device for another). The results showed that the mean listening-ease ratings increased linearly with a change in SNR (experiments 1-2), but an SNR change of at least 6 dB was
necessary to motivate participants to seek intervention (experiments 3-4). To probe individual variance, a questionnaire of general and hearing health was also administered to participants in the latter experiments.

Overall, the results indicate not only the difference limen for SNR, but also how large a change in SNR is needed for it to be meaningful to someone. While an SNR increase less than 3 dB may have relevance to speech-recognition performance, it may not be enough of an SNR improvement to be reliably recognized and, furthermore, may be too little increase to motivate potential users. [Supported by intramural funding from the Medical Research Council (grant number U135097131) and the Chief Scientist Office of the Scottish Government.]

9:15 AM  

A classification algorithm to improve speech recognition in noise for hearing-impaired listeners

Eric W. Healy, Sarah E. Yoho, Yuxuan Wang and DeLiang Wang
1 Department of Speech and Hearing Science
2 Department of Computer Science and Engineering
3 Center for Cognitive and Brain Sciences, The Ohio State University, Columbus, OH

Despite considerable effort, monaural (single-microphone) algorithms capable of increasing the intelligibility of speech in noise have remained elusive. Successful development of such an algorithm is especially important for hearing-impaired (HI) listeners, given their particular difficulty in noisy backgrounds. In the current study, an algorithm based on binary masking was developed to separate speech from background noise. Unlike the ideal binary mask (IBM), which requires prior knowledge of the premixed signals, the masks used to segregate speech from noise in the current study were estimated by a deep neural network (DNN) classifier trained on speech not used during testing. Sentences were mixed with speech-shaped noise and with babble at various signal-to-noise ratios (SNRs). Testing using normal-hearing (NH) and HI listeners indicated that intelligibility increased following processing in all conditions. These increases were larger for HI listeners, for the modulated background, and for the least-favorable SNRs. They were also often substantial, allowing several HI listeners to improve intelligibility from scores near zero to values above 70 percent.

The above results were further extended to consonant recognition in noise, in order to examine the specific speech cues responsible for the observed performance improvements. Consonant recognition in speech-shaped noise or babble was examined in NH and HI listeners in three conditions: unprocessed, noise removed via the IBM, and noise removed via the DNN classification algorithm. The IBM demonstrated large performance improvements in noisy consonant recognition, averaging up to 45 percentage points. The algorithm also produced substantial gains, averaging up to 34 percentage points. These results also indicate that the algorithm is capable of improving recognition of isolated phones in noise, which require increased accuracy of bottom-up acoustic information. An information transmission analysis of cues associated with manner of articulation, place of articulation, and voicing indicated general similarity in the cues transmitted by the IBM versus the classification algorithm.

From the perspective of hearing aids, the monaural nature of the algorithm provides inherent convenience in implementation compared to microphone-array techniques. The classification-based framework shifts much of the workload to the training stage, and during the operational (test) stage, the algorithm involves only feature extraction and bi-
nary labeling using trained classifiers, both of which can be performed efficiently [Work supported by NIH.]

FRIDAY, AUGUST 15

SESSION FIVE
Real World Challenges of Training and Service Delivery
Moderator: Louise Hickson

11:15 AM Health behavior change in adults with hearing impairment

Gabrielle Saunders¹, Melissa Frederick¹, ShienPei Silverman¹, Lisbeth Dons Jensen² and Ariane Laplante-Lévesque²
¹ National Center for Rehabilitative Auditory Research
² Eriksholm Research Center

About 80% of adults do not seek treatment during the initial 5-10 years after they suspect they have a hearing loss. This is despite the positive outcomes that arise from use of hearing aids or other interventions. Reasons for these low rates of help seeking and intervention uptake are not well understood. There is consistent evidence showing that measured impairment and self-reported hearing difficulty are related to both help seeking and intervention uptake, while age, gender and living arrangement are not. Findings of relationships between hearing health care behaviors and socioeconomic status, and psychosocial/psychological factors such as motivations, attitudes, and expectations, are mixed.

The neglect of help seeking and intervention uptake is not unique to hearing, and the factors influencing hearing health care behaviors show considerable overlap with those associated with health-related behaviors for other chronic medical conditions. Health behavior theory can provide a basis for understanding health behaviors and, more importantly, for developing theoretically-based strategies to change those behaviors. This study examines the application of two models of health behavior change commonly used in health psychology - the Health Belief Model and the Transtheoretical Model - to adults seeking hearing help for the first time.

Participants were recruited within a few days of attending an initial hearing assessment. They completed questionnaires to measure hearing-related attitudes and beliefs using the Hearing Beliefs Questionnaire (targeting the Health Belief Model), and the University of Rhode Island Change Assessment (targeting the Transtheoretical Model). They also completed the Hearing Handicap Inventory, the Psychosocial Impact of Hearing Loss scale and a demographics questionnaire. Six months later they completed these same questionnaires once again, along with the International Outcome Inventory for Hearing Aids and the Psychosocial Impact of Assistive Devices Scale if they had acquired hearing aids in the intervening six months.

Data from over 100 participants have been collected. The relationships between attitudes, beliefs, audiometric and demographic variables will be presented, as will the interrelationships between the health behavior models used in this study. In addition, the association between baseline attitudes and later behaviors will be examined, as will determina-
tion of whether behavior change is accompanied by changes in beliefs and attitudes. These data provide insight into the applicability of two health behavior theories to help seeking and intervention uptake for hearing loss.

11:45 AM  A profiling system for the selection of hearing aids in the Netherlands

Wouter Dreschler  
Academic Medical Center, Amsterdam

In the Netherlands a project started to make the selection process of a hearing aid more transparent. A system is needed to support the selection of an adequate solution: a simple hearing aid when possible and a more complex aid when necessary. In our opinion, scientific literature is not strong enough for an evidence-based approach in such a system. An additional complication is that the hearing aid characteristics are only partly public domain, because some details are kept secret or are hidden by commercially oriented names and theories.

For this purpose we designed a preliminary model to match a user profile (Human Related Intended Use) to a hearing aid profile (Product Related Intended Use).

The Human Related Intended Use (HRIU) is based on a self-report inventory of the user’s disabilities, supported by COSI data covering environmental factors, auditory tasks and individual targets for rehabilitation. Objective hearing tests (tone and speech audiograms) are included in the system, but play a minor role. This approach yields an individual HRIU profile with scores on 6 dimensions: detection, speech in quiet, speech in noise, localization, focus, and noise tolerance.

We made a complete inventory of the hearing aid characteristics of 1800 commercially available hearing aids. For each of the six dimension described above, we selected the features and characteristics that were assumed to be relevant for that dimension by a panel of independent experts. The scores on relevant items yielded a higher score for the potential of a specific hearing aid to be helpful for the compensation of that particular dimension. This resulted in a 6-dimensional Product Related Intended Use profile (PRIU), representing objective criteria for the complexity for the technology available for each of the six dimensions.

The HRIU-profile determines the degree of complexity and/or sophistication of the hearing aid to be selected and the PRIU profile is helpful in finding appropriate candidates within the (usually large) selection of hearing aids available.

Post-fitting, a well-standardized evaluation procedure is used, including the same inventory of disabilities. The results show the improvements in the 6 dimensions of auditory functioning. This determines whether the hearing aid is adequate and can be bought. But also it provides well-standardized data to evaluate the basic assumptions and to improve the system based on practice-based evidence.

This approach can only work for high numbers of fittings. The national implementation will allow us to include >10,000 cases each month. Valuable data will become available and practice-based evidence will compensate for the problem that an evidence-based approach was not possible in the beginning. The system of hearing aid fitting will become more transparent, the effectiveness of specific features can be analyzed and the individual improvements can be related to expectations based on large-sample populations.
SESSION SIX
Noise as a Cause and Challenge for people with Hearing Loss
Moderator: Todd Ricketts

5:00 PM  Development of a therapeutic to protect the inner ear: from animal models to human trials

Colleen Le Prell
University of Florida

Abstract: Noise-induced hearing loss (NIHL) is a significant clinical, social, and economic issue. We now know that noise-induced free radical formation leads to cell death and hearing loss. This key finding has opened the door to novel interventions that reduce the effects of noise on the inner ear. Many laboratories have now demonstrated that free radical scavengers (“antioxidants”) reduce NIHL in animal subjects. Scientific data supporting the use of specific agents to prevent or reduce NIHL have led to human clinical trials. Completed clinical trials in our laboratory will be discussed. Agents that reduce NIHL in animal models have also successfully reduced hearing loss induced by aminoglycoside antibiotics; data will be briefly reviewed.

Learning Objectives: 1) Describe cell death and neural loss after noise trauma and ototoxic drugs; 2) Identify the role of oxidative stress in drug-induced and noise-induced hearing loss; and 3) Discuss the opportunities for novel therapeutic intervention.

Speaker Bio: Dr. Colleen Le Prell is an Associate Professor in the Department of Speech, Language, and Hearing Sciences at the University of Florida, where she also directs the Center for Hearing Research. She has received research funding from the National Institutes of Health, the Department of Defense, and several foundations, and she has led industry-sponsored contracts. Current research programs in her laboratory at the University of Florida include an effort to identify and prevent the progression of biochemical processes that lead to cell death in the inner ear, as well as collaborative translational research programs directed at prevention of noise-induced hearing loss. She has published more than 30 research articles in peer-reviewed journals, contributed chapters to 10 books, and was lead editor for a 2012 Springer Handbook of Auditory Research, titled Noise-Induced Hearing Loss: Scientific Advances. She is currently editing a book on the role of oxidative stress in ENT pathology.

Disclosure: Dr. Le Prell received funding from Sound Pharmaceuticals Inc., to study potential protection using SPI-1005. She is a co-inventor on patents owned by the University of Michigan (UM), and has served as a scientific advisor to the UM patent licensee, Hearing Health Sciences Inc.

5:30 PM  Predicting the speech intelligibility in fluctuating and reverberant background noise using the Speech Transmission Index

Jelmer van Schoonhoven, Koenraad S. Rhebergen and Wouter A. Dreschler

1 Department of Clinical and Experimental Audiology
Academic Medical Center, Amsterdam
2 Department of Otorhinolaryngology and Head & Neck Surgery
Rudolf Magnus Institute of Neuroscience, University Medical Center Utrecht
**Introduction:** The concept of the Modulation Transfer Function (MTF) in the field of room acoustics was introduced by Houtgast & Steeneken (1973). The MTF can be used to predict speech intelligibility in stationary noise and reverberation and can be expressed in one single value: the Speech Transmission Index (STI). The drawback of STI measurements using the classical, direct method that it is time-consuming and that it is not validated for fluctuating background noise (IEC60268-16, 2011). However, the MTF as a result of reverberation can also be calculated “indirectly” using impulse response measurements (based on Schroeder, 1978). Furthermore, when calculating the STI per time segment and applying the same time averaging as used by Rhebergen et al., 2006 (regarding the Extended Speech Intelligibility Index), the “Extended” Speech Transmission Index (ESTI) can be calculated in fluctuating noise. The aim of the current research is to obtain validation data using speech intelligibility experiments.

**Methods:** Monaural speech intelligibility experiments using headphones were conducted. The 50% Speech Reception Threshold (SRT, Plomp & Mimpen, 1979) was measured using sentences (Versfeld et al. 2000) and digit triplets (Smits et al. 2013). Spectrally matched stationary noise and 8 Hz interrupted noise were used as background noise. The SRTs were measured with reverberation times between 0 and 1.2 seconds. In the first phase of the experiments we measured SRTs in normal hearing listeners (work in progress).

**Results:** In stationary noise the effects of reverberation on the ESTI are small, which is in agreement with the results of Duquesnoy & Plomp (1980). For 8Hz interrupted noise, the ESTI values in the conditions without reverberation are somewhat lower, but for high reverberation times (T60 = 1.2 s) the ESTI values approach those for stationary noise. These main effects are found both for sentences and for digit triplets.

**Discussion and Conclusion:** Due to the gaps in the noise the SRT in interrupted noise is lower than in stationary noise, but this advantage disappears when adding reverberation. This effect is reflected in the ESTI values. Our preliminary results present possibilities to examine the ESTI model, but more experiments will be done in the near future in order to validate the model approach. We also aim to use a wider variety of background noises and to include hearing impaired subjects and children. A major goal is to use the ESTI regularly in the evaluation of listening conditions in classrooms.

**6:00 PM Spectrotemporal modulation sensitivity as a predictor of speech intelligibility in noise with hearing aids**

*Joshua G. W. Bernstein¹, Henrik Danielsson², Mathias Hållgren², Stefan Stenfelt², Jerker Rönnberg² and Thomas Lunner²*

¹ National Military Audiology and Speech Pathology Center
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Linkoping University, Linkoping, Sweden

The audiogram predicts less than a third of the variance in speech reception thresholds (SRTs) for hearing-impaired (HI) listeners properly fit with individualized frequency-dependent gain. The remaining variance is often attributed to a combination of suprathreshold distortion in the auditory pathway and non-auditory factors such as cognitive processing. Distinguishing between these factors requires a measure of suprathreshold auditory processing to account for the non-cognitive contributions. Preliminary results in 12 HI listeners identified a correlation between spectrotemporal modulation (STM) sensitivity and speech intelligibility in noise presented over headphones. The cur-
rent study assessed the effectiveness of STM sensitivity as a measure of suprathreshold auditory function to predict free-field SRTs in noise for a larger group of 47 HI listeners with hearing aids.

SRTs were measured for Hagerman sentences presented at 65 dB SPL in stationary speech-weighted noise or four-talker babble. Pre-recorded speech and masker stimuli were played through a small anechoic chamber equipped with a master hearing aid programmed with individualized gain. The output from an IEC711 Ear Simulator was played binaurally through insert earphones. Three processing algorithms were examined: linear gain, linear gain plus noise reduction, or fast-acting compressive gain.

STM stimuli consist of spectrally-rippled noise with spectral-peak frequencies that shift over time. STM with a 2-cycle/octave spectral-ripple density and a 4-Hz modulation rate was applied to a 2-kHz lowpass-filtered pink-noise carrier. Stimuli were presented over headphones at 80 dB SPL (±5-dB roving). The threshold modulation depth was estimated adaptively in a two-alternative forced-choice task.

STM sensitivity was strongly correlated ($R^2=0.48$) with the global SRT (i.e., the SRTs averaged across masker and processing conditions). The high-frequency pure-tone average (3-8 kHz) and age together accounted for 23% of the variance in global SRT. STM sensitivity accounted for an additional 28% of the variance in global SRT (total $R^2=0.51$) when combined with these two other metrics in a multiple-regression analysis. Correlations between STM sensitivity and SRTs for individual conditions were weaker for noise reduction than for the other algorithms, and marginally stronger for babble than for stationary noise.

The results are discussed in the context of previous work suggesting that STM sensitivity for low rates and low carrier frequencies is impaired by a reduced ability to use temporal fine-structure information to detect slowly shifting spectral peaks. STM detection is a fast, simple test of suprathreshold auditory function that accounts for a substantial proportion of variability in hearing-aid outcomes for speech perception in noise.

6:30 PM

**Is a non-auditory profile associated with bilateral directional processing benefit in challenging listening situations?**

*Gitte Keidser, Els Walravens and Jorge Mejia*

*National Acoustic Laboratories*

Recent years has seen an increasing interest in bilateral directional processing schemes that combine the directional input from both sides of the head to produce a highly directional beam. These schemes are known as beamformers. Laboratory data obtained in ideal conditions where target speech is arriving from directly in front of the listener and noise from surrounding sources, show that beamformers offer significant improvements in the signal-to-noise ratio (SNR), and hence speech intelligibility, over conventional directional microphones. Such results promise substantial improvements in hearing aid wearers’ ability to hear in noisy environments with this technology. However, noisy real-life environments are often dynamic; i.e., involving more than one talker and moving noise sources.

In a recent study conducted at the National Acoustic Laboratories that evaluated two different beamformers, it was found that audibility and thresholds explained less of large inter-individual differences measured in beamformer benefit as the noise got denser and when the target switched between several locations. In this study we explore to what extent the benefit obtained with beamformers in such challenging listening situations can be
explained by a non-auditory profile. Specifically, we obtained measurements of listening effort, switching and divided attention, working memory, executive function, processing speed, phonemic fluency, lexical access, and comprehension on the test participants. This paper presents results from correlational and multi-regression analyses to explain the impact non-auditory factors may have on the ability to utilise bilateral directional processing in complex real-life situations.

SATURDAY, AUGUST 16

SESSION SEVEN

Objective Modelling of Hearing Aid Functions

Moderator: Tim Trine

8:00 AM Understanding contemporary hearing aid algorithms and their relation to sensory factors and human performance

Birger Kollmeier
Universität Oldenburg & Cluster of Excellence Hearing4all, Oldenburg, Germany

Today, all major brands offer a range of fully digital devices that allow for a variety of novel processing schemes for improved rehabilitation of hearing impairment, including feedback control and management, automatic classification of the acoustic environment, multi-channel amplitude compression, speech enhancement and single-channel and multi-channel noise reduction. Recent Wireless Body Area Network (WBAN) technology allows for the transmission of audio and control signals across hearing devices attached to both ears, enabling true binaural signal processing. Thus, hearing aid technology has evolved rapidly in the last decades and is still a very active field of research and development. Although acoustic communication has improved by recent DSP, there is room for improvement, in particular in challenging acoustic conditions characterized by noise and reverberation and in tailoring the processing options to the needs of the individual listener.

The current talk will consider how much benefit the algorithms provide to the individual user in relation to his or her specific auditory speech communication problem. To do so, a better understanding of both the sensorineural and cognitive factors of hearing impairment has to be achieved. The knowledge about these factors can be condensed into “effective” processing models of the (impaired) auditory system and models of “aided performance prediction” to characterize the interaction between hearing aid processing and the patients “internal” processing as a way to assess the potential individual benefit for each user. The development and evaluation of such model-based processing, evaluation and fitting schemes is a key topic of our cluster of excellence “Hearing4all” that bundles more than 400 researchers in the Auditory Valley” R&D network in Oldenburg and Hannover. This consortium covers a broad range of disciplines and research topics all centered around finding better ways of helping the hard-of-hearing. Some of the current research lines and results from the cluster of excellence will be reviewed. Among them, elements of an assistive listening device will be introduced that both fits the requirements of near-to-normal listeners (i.e., providing benefit in noisy situations or other daily life
acoustical challenges using the concept of acoustically “transparent” earpieces) and, in addition, can be scaled up to a complete hearing aid for a more substantial hearing loss. The current prototype runs on the binaural, cable-connected master hearing aid (MHA) that includes earpieces allowing for approaching acoustic transparency. As an example, the evaluation of a binaural high-fidelity enhancement algorithm motivated by interaural magnification is described which was performed to assess its benefit for normal and near-to-normal listeners. [Supported by Deutsche Forschungsgemeinschaft (DFG). Key collaborators in the work reviewed here are – among others - Volker Hohmann, Stephan Ewert, Thomas Brand, Stephan Ernst, Tobias Neher, Tim Jürgens, Jörn Anemüller, Ania Warzybok and Marko Hiipakka.]

8:30 AM  A unified approach to predicting speech intelligibility and quality

James M. Kates and Kathryn H. Arehart
Department of Speech Language and Hearing Sciences
University of Colorado, Boulder, CO

The ability to predict speech intelligibility and speech quality is important for designing hearing-aid signal processing algorithms and in fitting those algorithms to hearing-impaired listeners. Intelligibility and quality indices are presented that are based on a shared model of the auditory periphery that incorporates the effects of hearing loss. The intelligibility index is the Hearing-Aid Speech Perception Index (HASPI), and the quality index is the Hearing-Aid Speech Quality Index (HASQI) version 2. HASPI compares the envelope and temporal fine structure outputs of the auditory model for a reference signal to the outputs of the model for the degraded signal under test. The auditory model for the reference signal is set for normal hearing, while the model for the degraded signal incorporates the peripheral hearing loss. HASQI compares the model outputs for the reference signal to the outputs for the degraded signal where both auditory models incorporate the hearing loss. HASQI comprises two terms: a nonlinear term that measures envelope and temporal fine structure modifications, and a linear term that measures long-term spectral changes. The quality prediction is formed by taking the product of the nonlinear and linear terms. The auditory model and the HASPI and HASQI implementations will be described, and the accuracy of the index predictions will be illustrated for several different intelligibility and quality experiments.

9:00 AM  Perceptual evaluation of noise reduction in hearing aids

Inge Brons, Rolph Houben and Wouter A. Dreschler
Clinical & Experimental Audiology, Academic Medical Center, Amsterdam
The Netherlands

Single-microphone noise reduction is a common feature in modern hearing aids. It should suppress unwanted background noise without affecting the target speech. Despite its broad application, there is only little knowledge on the implementation and effects of noise reduction in hearing aids. Clinicians who are responsible for prescribing and fitting the hearing aids have no insight in the consequences of activating noise reduction in a hearing aid. As a consequence, it remains uncertain if the appropriate hearing aid has been chosen and if the noise reduction has been fitted optimally for the user to compensate for reduced speech perception in noise.

In order to gain insight in the “black box” of hearing aid noise reduction, we recorded the noise-reduction output from different hearing aids. Acoustical analyses of these recordings provided information on noise-reduction parameters like the strength, frequency-
dependence, and dynamic characteristics of noise reduction, both for noise reduction in isolation and in combination with compression. Perceptual experiments with normal-hearing subjects and hearing-impaired subjects provided information on the effects of noise reduction on speech intelligibility in noise, perceived listening effort, noise annoyance, speech naturalness, and overall preference.

We will present an overview of this series of experiments on hearing-aid noise reduction and discuss the relation between acoustical and perceptual outcome measures. The implications of our results for the implementation and clinical application of noise reduction in hearing aids will be discussed.

9:30 AM  The influence of structure in binary mask estimation error on speech intelligibility

Abigail A. Kressner and Christopher J. Rozell
Georgia Institute of Technology

Listening to speech in the presence of background noise is a particularly demanding task for normal-hearing listeners and even more so for hearing-impaired listeners. Researchers have attempted to solve this problem from various angles, but traditional approaches fail to improve speech intelligibility. More recently however, researchers have demonstrated significant improvements in intelligibility for both normal-hearing and hearing-impaired listeners with the ideal binary mask. The ideal binary mask exploits a priori knowledge of the target speech and interfering masker signals to preserve only the time-frequency regions that are target-dominated. Although requiring a priori knowledge makes the ideal binary mask an impractical solution for most applications, the significant increase in intelligibility makes it a desirable benchmark. While this benchmark has been studied extensively in the literature and many researchers are developing algorithms to estimate the mask without a priori information, many questions still remain about the factors that influence the intelligibility of binary-masked speech. To date, researchers have used primarily uniformly random, uncorrelated mask errors and independently presented error types (i.e., false positives or “false alarms” and false negatives or “misses”) to characterize the influence of binary mask estimation errors on intelligibility. However, practical binary mask estimation algorithms will produce masks that contain errors of both types simultaneously and with nontrivial amounts of correlation among neighboring time-frequency units (i.e., structure). In this study, we first investigate how this structure in the errors influences the tolerance of false positives and false negatives, and then we investigate how both unstructured and structured error types interact. Previous research has suggested that false positives are significantly more detrimental to intelligibility than false negatives, but we demonstrate that false negatives can be just as detrimental to intelligibility if they contain structure. Furthermore, we show that the effects of false positives and false negatives are not independent. Even when errors are uncorrelated, relatively low levels of false negatives can be detrimental to intelligibility if the masks contain a non-zero level of false positives, and additionally, the detrimental effect of interacting errors is even more significant when the errors contain structure.
SESSION EIGHT

Bimodal Hearing for Children and Adults

Moderator: Jan Wouters

11:30 AM  Considerations in fitting and evaluating bimodal devices for pediatric CI recipients

Lisa Davidson
Washington University in St. Louis

Bimodal device use [cochlear implant (CI) + hearing aid (HA) at the non-implanted ear] is one avenue for bilateral stimulation of the auditory pathways for adults and children with severe to profound hearing loss requiring cochlear implantation. Studies have demonstrated that bimodal fittings improve perception in quiet and noise, localization, and sound quality compared with CI only fittings. Furthermore, there is evidence that some period of bimodal device use may have benefits for speech perception and language outcomes for children eventually receiving a second CI (bilateral CI use). The number of CI recipients using bimodal devices will likely increase as adults and children with more residual hearing are considered candidates for cochlear implantation. The need for a coordinated fitting between the CI and the HA becomes more relevant as bimodal device users present with greater degrees of residual hearing at the non-implanted ear. There is also a need for valid predictors of bimodal benefit. Predictors based on non-verbal, psychoacoustic performance (unaided thresholds and more) in the non-implanted ear would be especially useful for pediatric patients. Additionally, when evaluating benefit, there is a need for outcome measures that better reflect listening in real-world situations. This presentation will review the research literature and summarize the issues related to fitting both the CI and the HA for bimodal users. We will present data from our lab collected, which includes a variety of speech perception measures, to evaluate the effects of three different HA frequency responses (FR) on bimodal benefit in pediatric and adolescent bimodal device recipients. Bimodal fittings with three frequency responses were compared to the CI-alone condition. The three frequency responses are: 1) the traditional wide-band frequency response, 2) restricted high-frequency gain based on residual hearing and real-ear gain, and 3) activated non-linear frequency compression. We will also present recent pilot data that explore the relation between spectral modulation detection performance and bimodal speech perception benefit in pediatric bimodal recipients.

12:00 PM  Frequency dependent loudness balancing and AGC matching to optimize hearing aid benefits in bimodal CI users

L.C.E. Veugen\textsuperscript{1}, J. Chalupper\textsuperscript{2}, A.F.M. Snik\textsuperscript{3}, A.J. van Opstal\textsuperscript{1} and L.H.M. Mens\textsuperscript{3}
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\textsuperscript{3}Department of Otorhinolaryngology, Radboud University Nijmegen Medical Centre, The Netherlands

Despite the overall positive results of the combined use of a cochlear implant (CI) and a contralateral hearing aid (HA), large individual differences in bimodal benefit have been
reported. A possible reason for poor results is a mismatch in loudness between ears across the dynamic range and/or across frequencies.

At present, bimodal fitting guidelines recommend adjusting overall HA gain to equalize loudness of soft and loud speech in both ears, with limited adjustments of the frequency response on the basis of clarity judgments [1]. In the present study, we adjusted HA gain for soft and loud input signals in three frequency bands (0 - 500, 500 - 1000 and >1000 Hz) to match loudness with the CI (“three-band” balancing). Before balancing, HA gain was reduced where the hearing loss exceeded 120 dB HL. This procedure was compared to a simple balancing procedure using the International Speech Test Signal (ISTS) at 65 dB SPL (“broadband” balancing) and the HA manufacturer’s prescriptive rule. In a second phase of the study, HAs were engineered to match the fast and slow time constants of the dual-loop automatic gain control (AGC) in the CI processor (“AGC-matched HA”).

Fifteen CI users, all wearing an Advanced Bionics Harmony processor, were fitted with one and the same HA (Phonak Naida S IX UP), eliminating variability due to devices. In a four-visit cross-over design with four weeks between sessions “broadband” and “three-band” balancing was tested; in the second phase the “AGC-matched HA” was tested versus a HA with standard signal processing (“standard HA”). Bimodal benefit was assessed with subjective questionnaires and speech understanding in quiet and noise under several listening conditions.

Speech in noise results showed a bimodal benefit ranging from -0.2 to 1.8 dB for stationary noise (SØN±90) and -0.3 to 3.1 dB for ISTS noise (SØN±90, SØNCl, SØNHA). Three-band loudness balancing brought no additional benefit compared to broadband fitting. The AGC-matched HA improved bimodal benefit by an additional 0.6 to 1.8 dB compared to standard signal processing. Questionnaires showed no differences between fittings. However, increased listening comfort was found for the AGC-matched HAs compared to the other fittings, as indicated by the subjective preference test that was performed at the end of the study. After the study, nine out of fifteen subjects preferred the AGC-matched HA, one preferred the commercial HA and five subjects did not have a preference. [Study is financially supported by Advanced Bionics.]

Reference:
SATURDAY, AUGUST 16

SESSION NINE
Listening and Technologies in Real World Environments
Moderator: Ben Hornsby

5:00 PM  Robust wind noise detection
Justin A. Zakis and Christine M. Tan
Wolfson Dynamic Hearing

Wind noise is a major problem with hearing aids, with a major survey showing 58% satisfaction rate in wind compared with 61% for noisy situations and 91% for one-on-one conversation in quiet (Kochkin, 2010). Wind noise can exceed speech levels up to 1 and 6.3 kHz in 3 and 6-m/s wind, and saturate microphone circuits at 12 m/s (43 km/h) with input levels up to 116 dB SPL (Zakis, 2011). Therefore, wind noise has great potential to reduce speech understanding, listening comfort and/or sound quality, so aggressive wind-noise reduction (WNR) algorithms are required. It is crucial that aggressive WNR algorithms are controlled by reliable wind-noise detection (WND) algorithms. Otherwise, WNR may not be engaged when needed in wind, or may be inappropriately engaged in the absence of wind and unintentionally reduce the audibility and quality of speech. However, the authors are unaware of studies that compare the efficacy of WND algorithms in hearing aids. Since wind noise is due to localised turbulence in air flows, it tends to be less correlated between microphones than a propagating acoustic wave. Therefore, previous WND algorithms typically detect wind noise when level and/or phase differences between microphones become sufficiently large. However, large differences may also exist due to the microphone spacing, unmatched microphones, acoustic reflections, and/or near-field sound sources, while relatively small differences can result from saturating wind noise, all of which may lead to incorrect WND. The efficacy of two such previous WND algorithms was compared with the novel Chi-squared WND algorithm, which uses a statistical approach to detect wind noise. Behind-the-ear, hearing-aid microphone output signals were recorded with a range of wind and non-wind (speech, LTASS noise, tones) stimuli, and processed by MATLAB implementations of the WND algorithms. The novel WND algorithm was substantially more reliable than the previous WND algorithms at separating the wind and non-wind stimuli (the exact difference depends on hearing-aid shell design). Implications for the design and fitting of WND and WNR algorithms will be discussed.

References:

5:30 PM  Optimizing forward-masked thresholds with amplification
Marc Brennan, Walt Jesteadt and Ryan McCreery
Boys Town National Research Hospital

Previous research has demonstrated that participants with sensorineural hearing loss have poorer temporal resolution than participants with normal hearing. The poorer ability to perceive temporal cues could impact their ability to decode speech. These differences in
performance are most likely to occur when audibility is poor. Audibility with amplification is determined by the gain and compression speed settings. Specifically, audibility can be improved by increasing the gain provided and by using fast-acting compression. Our hypothesis is that the provision of increased gain and fast-acting compression is expected to best restore the normal ability to perceive temporal cues. We measured forward-masked thresholds on a group of adults with normal hearing and a group of aged-matched adults with hearing loss (n=12 per group). Adults with normal hearing were tested unaided. For the listeners with hearing loss we manipulated the gain by testing forward-masked thresholds with two different prescriptive targets. Specifically the output levels were set to the levels provided by the adult and pediatric versions of Desired Sensation Level Algorithm (DSL). Three different compression speeds were evaluated (slow compression: 150 ms attack time, 1500 ms release time; medium compression: 50 ms attack time, 150 ms release time; fast compression: 1 ms attack time, 5 ms release time). Data were collected using hearing-aid software. Both the masker and the signal were 2 kHz pure tones. The masker was fixed at 70 dB SPL. The data were analyzed to determine if forward-masked thresholds differed as a function of compression speed (slow, medium, fast), prescriptive method (adult, pediatric), signal delay and hearing status (normal hearing, hearing loss). For the participants with hearing loss, forward-masked thresholds were lower with faster compression speed and with the increased gain provided by the pediatric version of DSL. Forward-masked thresholds approximated the results observed in the listeners with normal hearing when a faster compression speed with the pediatric version of DSL was used. We conclude that temporal resolution, as measured using forward masking in adults with hearing loss, is dependent on compression speed and gain. Setting gain and compression speeds to improve temporal resolution could lead to improvements in speech understanding.

6:00 PM

Contribution of monaural and binaural cues to sound localization in listeners with unilateral conductive hearing loss (UCHL) and in total single sided deafness (SSD)

Martijn Agterberg1,2, Marc Van Wanrooij1,2, Myrthe Hol2, John Van Opstal1 and Ad Snik2
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Introduction: Sound localization in the horizontal plane requires neural processing of binaural difference cues in timing (ITD) and sound level (ILD). Spectral pinna cues are used for localization in the vertical plane. There is limited objective data about sound localization abilities in listeners who are fitted with auditory implants (middle ear implants and bone-conduction devices). Many clinical-studies demonstrate improved aided directional hearing after testing the localization abilities in a setup with a limited number of speakers and presenting stimuli which were fixed or varied over a small range. These studies actually demonstrate that ambiguous cues like the head-shadow effect and sound level can be used because they apply low variation in sound location, spectrum and level.

Methods: Listeners point a head-fixed laser in the perceived sound direction in a completely dark, sound-attenuated room. Horizontal and vertical head-movements are recorded with the magnetic search coil induced technique (Bremen et al., 2010). Stimuli consist of broadband (0.5-20 kHz), high-pass (3-20 kHz), low-pass (0.5-1.5 kHz) and narrowband (0.5 or 3 kHz) noise roved over a large range (45-65 dB SPL). Listeners with unilat-
eral conductive hearing loss (UCHL) and listeners with single sided deafness (SSD) were aided with a bone-conduction device on a percutaneous titanium screw.

**Results**: Listeners with SSD were able to use spectral cues for the localization of sounds in azimuth. In these listeners inter-subject variability could be explained to a large extent by the severity of high-frequency hearing loss of the hearing ear. A bone-conduction device did not deteriorate their monaural directional hearing abilities. Listeners with UCHL fitted with a bone-anchored hearing implant demonstrated improved localization abilities on the basis of restored use of ILDs and ITDs. We demonstrate that for listeners who lack binaural cues, sound level is not an adequate cue.

**Conclusions**: We demonstrate that several auditory implants provide successful use of ITDs and ILDs in listeners with UCHL and that these implants do not deteriorate directional hearing in SSD listeners without high-frequency hearing loss of their hearing ear, who are able to localize sounds on the basis of spectral pinna cues.

**The effect of amplitude compression on the ability to hear out individual lines in music**

Sara M. K. Madsen¹, Michael A. Stone¹, Martin McKinney², Kelly Fitz², and Brian C. J. Moore¹

¹Department of Experimental Psychology, University of Cambridge, UK
²Starkey Hearing Technologies, Eden Prairie, MN, USA

Trying to hear out one melodic line in a musical score is similar to trying to understand one talker in the presence of other talkers. This is often more difficult for hearing-impaired than for normal-hearing listeners. Hearing out individual lines is important in the appreciation of music and it is desirable that the ability to do this is not further reduced by the signal processing in hearing aids. Stone and Moore (2008) found that fast-acting amplitude compression made it harder to hear out speech in the presence of a competing talker for speech processed to retain mainly envelope cues. They argued that this could be partially explained by cross-modulation: the envelopes of the target and background became partially correlated after application of a rapidly varying gain and this hindered the perceptual segregation of the target from the background.

This study investigated the effect of multi-channel amplitude compression on the ability of hearing-impaired participants to hear out one musical instrument or voice from other instruments and explored whether cross-modulation impaired this ability. Ten participants with moderate hearing loss took part. They were asked to judge the relative clarity with which a specific instrument or voice could be heard in two successive stimuli. The two stimuli consisted of the same music excerpt but processed in two different ways. To indicate what instrument/voice to attend to, the excerpt played/sung by that particular instrument/singer in isolation was presented immediately before presentation of the pair of stimuli to be compared. A hearing aid simulator with a five-channel compression system was used to process the stimuli, using compression ratios and gains recommended by the CAM2 hearing aid fitting method for that individual. The attack/release times were 10/100 ms for the fast compression condition and 50/3000 ms for the slow compression condition. The individual instruments/voices were either compressed individually and then mixed (producing no cross-modulation) or were mixed and then compressed (producing cross-modulation). A condition using linear amplification was also used.

Clarity was generally rated higher for linear amplification than for either fast or slow compression, indicating that both fast and slow compression made the individual instru-
ments more difficult to hear out. There was no clear subjective effect of cross-modulation. [This work was supported by Starkey (USA).]

Reference:
Poster Program

Posters for Session A should be put up by 8:00 AM Thursday, August 14, and taken down after 10:00 PM Thursday, August 14, or before 7:00 AM Friday, August 15. Presenters should be at their posters from 10:00 AM – 11:30 A.M.; 8:30 PM – 10:00 P.M.

POSTER SESSION A
Thursday 10:00 AM – 11:30 AM

A1
The effects of amplification and visual speech cues on the speech reception threshold and slope of the intelligibility performance function
Ken W. Grant, Douglas S. Brungart and Danielle J. Zion
Walter Reed National Military Medical Center

The ability of hearing-impaired (HI) individuals to understand speech in noisy environments is influenced by two main peripheral auditory factors: an attenuation factor and a distortion factor. Clinically, these factors are estimated by the audiogram and the speech reception threshold (SRT) or, when compared to the average SRT of a group of normal-hearing listeners, the SNR loss. However, understanding speech in noise also requires intact and efficient central and cognitive processes to integrate (across time, frequency, and ears) and interpret speech cues against a lifetime of linguistic experience. Add in the fact that a good deal of speech processing in noise involves both auditory and visual cues and it becomes abundantly clear that difficulties experienced by HI patients cannot be predicted by the patient’s audiogram or by standard measures of SNR loss. In order to begin to assess the multitude of possible factors that lead to individual differences in speech understanding in noise among HI patients (especially elderly HI patients), hours of testing for central auditory and cognitive processing abilities would be necessary making this exploratory approach unfeasible. What is needed is a fast and reliable method for estimating the deficit in speech perception in noise due primarily to central auditory and cognitive processing. The proposed method for obtaining this information is to measure the slope of the intelligibility function (intelligibility as a function of SNR) with shallow slopes indicating relatively poor processing efficiency and steeper slopes indicating relatively better processing efficiency. In this study, we explore the feasibility of measuring the slope of the intelligibility function for individual normal hearing (NH) and HI subjects using a procedure that simultaneously provides a fast measure of SNR loss for any specified performance level. Similar to previous studies, we show that a patient’s SNR loss and audiogram are mostly unrelated. In addition, we show that a patient’s slope around a specified SRT is relatively independent of the patient’s SNR loss and of the audiogram, thus providing a new source of information about the patient’s speech processing efficiency. We also tested the degree to which individualized amplification and/or visual speech cues (lipreading) are able to improve the quality of speech processing in noise in HI patients. Finally, we tested whether the improvement due to amplification or visual cues approaches normal speech processing in noise.

A2
Improving speech understanding in adverse listening environments using distributed microphone processing
Tao Zhang, Bill Woods and Ivo Merks
Starkey Hearing Technologies

With the advent of wireless communication between hearing aids and smartphones, it is possible to leverage not only the hearing aid microphones but also smartphone microphones to greatly improve speech understanding in adverse and highly dynamic listening environments. Such microphone processing is often referred to as distributed microphone processing (DMP). Compared to traditional monaural or binaural microphone processing, DMP allows free movement of individual microphones relative to each other. Furthermore, its speech understanding performance may be minimally affected when a small
number of microphones drop out of processing if designed properly. However, DMP can introduce a significantly longer propagation delay compared to the existing hearing aid propagation delay. Acceptance of such a long delay by hearing impaired listeners in adverse listening environments is unclear. In addition, imperfections of wireless communication networks and variations in microphone frequency responses can significantly affect DMP performance. Finally, the development of DMP needs a comprehensive and realistic recording database which captures important and natural interactions among talkers, listeners, and interferers. In this presentation, we first demonstrate the potential benefits of DMP using a realistic recording database. Then, we review the relevant technical challenges and how they may affect the DMP performance and introduce research efforts directed to tackle these challenges. Finally, we provide an outlook on how DMP may evolve as the relevant technologies advance.

**A3**

The perception of wind noise and its effect on speech understanding with hearing aids

Justin A. Zakis and Christine M. Tan
Wolfson Dynamic Hearing

Wind noise is one of the least satisfactory areas of hearing-aid performance, with only 58% of surveyed respondents reporting some degree of satisfaction in wind (Kochkin, 2010). Wind noise can exceed speech levels up to 1 and 6.3 kHz at low wind speeds of 3 and 6 m/s, respectively, and cause microphone saturation and input levels of 116 dB SPL at 12 m/s (Zakis, 2011). Despite this, the authors are unaware of any studies on the perception of wind noise and its effect on speech understanding with hearing aids. Such studies are important for estimating: 1) When algorithms should reduce wind noise; 2) An appropriate gain reduction from wind-noise-reduction algorithms; and 3) Whether the above depend on the presence of speech and/or non-wind noise. The current, on-going study aims to address these research questions. A pair of behind-the-ear hearing aids mixed their live microphone signals with calibrated wind-noise recordings presented via their direct audio inputs (recordings made at the microphone output of the same hearing-aid model on a KEMAR with a wind machine). When binaurally fitted with these devices and ear molds, hearing-impaired participants (35-80 dB HL across frequency) were presented with BKB/A sentences in quiet and wind at 2, 3, 4 and 5 m/s. For each condition, speech understanding was measured, and the quality and loudness of speech and wind were rated. These 5 conditions were repeated with the addition of 8-talker babble to simulate an outdoor gathering. All 10 resulting conditions were repeated with stationary noise reduction turned on, since it is commonly used in clinical practice and can affect wind noise. We will report the results of Stage 1 with ADRO (Adaptive Dynamic Range Optimisation) amplification, and preliminary results for Stage 2 with WDRC (Wide Dynamic Range Compression) fitted with NAL-NL2. Implications for the design and fitting of wind-noise detection and reduction algorithms in hearing aids will be discussed.

**References:**


**A4**

Acoustic analysis of consonant confusions in the California Syllable Test (CaST)

E. William Yund1 and David L. Woods1,2
1 VA Medical Center, Martinez, CA
2 UC Davis

Many recent studies have focused on listeners’ The CaST measures the signal-to-noise ratios (SNRs) for 67% correct identification of 20 onset and 20 coda consonants in consonant-vowel-consonant (CVC) syllables presented in speech-spectrum noise (SSN). The stimulus set includes 40 exemplars of each initial and final consonant for each of three vowels (/a/, /i/, and /u/) and each of four talkers. CaST normative results for young-normal-hearing (YNH) and older-normal-hearing (ONH) listeners have been published [JASA 127: 1609-1623 (2010); JRRD 49: 1277-
1292 (2012)]. The purpose of the present study was to determine how effectively the spectral shapes of the unmasked portions of onset consonants accounts for the pattern of consonant confusions. The acoustic analysis includes: (1) separating each initial consonant (including any formant transition) from the remainder of the CVC, (2) computing time-normalized spectrograms for each exemplar by varying the time-spacing of the FFT spectral lines in proportion to the exemplar duration, (3) computing the average spectrogram for each consonant, for each vowel and talker, and (4) using the unmasked portion of the average spectrogram as a template to determine the best acoustic match for each consonant. The observed effects of SNR and vowel context on the confusion matrices of YNH and ONH listeners will be compared with the predictions derived from acoustic analysis. The long-term goal is to apply this acoustic analysis to clarify the effects of hearing loss and hearing aids on consonant identification in noise.

**A5**

**Detection and fusion in combined electric and acoustic stimulation**

*Christopher Fowler, Ashley Graham and Yang-soo Yoon*

*Department of Speech, Language, & Hearing Sciences, Texas Tech University Health Sciences Center, 3601 4th St, Lubbock, TX 79430*

Evidence has established that electric acoustic stimulation (EAS, a hearing aid or HA in one ear and a cochlear implant or CI in the opposite ear) provides considerable benefit in speech-in-noise tests, music perception, and gender discrimination. In EAS, patients must detect spectral and temporal cues independently processed by a HA and a CI and fuse them together. Hence, EAS benefit is associated with the efficiency of detecting and fusing spectro-temporal cues across two modalities. We evaluated how the detection ability of the HA influences fusion ability in EAS patients using systematic and comprehensive psychoacoustic approaches.

Two adult EAS subjects were post-lingually deafened and native speakers of American English. All subjects should be users of both a HA and CI full time for at least 6 months prior to testing. Three psychoacoustic measures were administrated: the amplitude modulation detection threshold (temporal processing), the spectral ripple detection threshold (broadband spectral processing), and frequency difference limens (narrowband spectral processing). These measures were used to evaluate how the detection ability of the HA is related to fusion ability in spectral and temporal information for greater EAS benefit. Consonant, vowel, and sentence recognition were measured as a function of signal-to-noise ratio. These measures were used to evaluate how EAS benefit in speech perception is related to detection ability with the HA in spectral and temporal processing. Using subject’s clinical devices and settings, each outcome measure were performed with a HA alone, a CI alone, and a HA+CI at the most comfortable level, directly facing the loudspeaker (0° azimuth) one meter away.

The results showed that better detection ability with the HA alone higher EAS benefit in speech perception. In addition, consonant and vowel confusions revealed that one subject with better detection ability in psychoacoustic measures has less confusion and received greater EAS benefit, compared to the other subject with a poorer detection ability with a HA. Complete outcomes will be fully presented and discussed.

We expect that results from the this study can provide evidence in efficiency criteria for high success in EAS use so that clinicians will have evidence-based practice in selecting ear for implantation or for the use of acoustic hearing. For current use of EAS, this study can provide an important rehabilitation framework by allowing researchers to develop an auditory training program which help improve detection and fusion ability in spectral and temporal cues, leading to further EAS benefit. [Work supported by The National Organization for Hearing Research Foundation]
Spatial release of cognitive load in normal-hearing and hearing-impaired listeners

Jing Xia, Nazanin Nooraei, Sridhar Kalluri and Brent Edwards
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Spatially separating interfering talkers from a target talker helps listeners better understand the target speech. Listeners with hearing loss, compared to normal-hearing listeners, generally show less benefit for speech intelligibility from increasing spatial separation. To better understand anecdotal evidence that listening is easier with greater spatial separation, we investigated whether increasing spatial separation between talkers also helps reduce the cognitive load of understanding speech in multi-talker environments. Furthermore, we investigated whether cognitive load and speech intelligibility depend differently on spatial separation, and whether hearing loss has any impact on the cognitive cost of using spatial hearing.

Performance on a visual-tracking secondary task served as a measure of the cognitive load imposed by the primary speech-reception task. We measured dual-task performance in conditions when target and interferers were distinguished by 1) both gender and spatial location, 2) gender only, 3) spatial location only, and 4) neither gender nor spatial location. By comparing the performance in the first two conditions, we assessed the effect of spatial separation when talkers could already be well segregated based on gender difference in voices. Comparing performance in the last two conditions revealed the effect of spatial separation when it was the primary cue for segregating talkers.

With small spatial separation between talkers (Experiment I), additional location cues reduced cognitive load of listening even though it did not provide further improvement in speech intelligibility over existing gender cues. This suggested that the role of spatial separation in releasing cognitive load may be independent of improving speech intelligibility. Compared to normal-hearing listeners, hearing-impaired listeners experienced a relatively small benefit of spatial separation. Experiment II used a large spatial separation between talkers, so that speech intelligibility was comparable in the location-only condition to that in the gender-only condition. This enabled direct comparison of the cognitive load of using location cue only with the cognitive load of using gender only to understand target speech. Comparable speech intelligibility was achieved with a lower cognitive load when only location cues were available to use compared to when only gender cues were available. Taken together, the two experiments suggest that comparable speech intelligibility can be achieved by exerting different amount of cognitive effort, depending on which segregation cues are used for understanding the target speech. Thus, a measure of cognitive load might provide another dimension besides intelligibility for assessing the benefit of spatial separation in multi-talker environments.

A real-world recording database for distributed microphone processing research

William S Woods, Ivo Merks, Buye Xu and Tao Zhang
Starkey Hearing Technologies, Eden Prairie, MN, USA

Wireless devices with microphones such as cell phones are ubiquitous. Such devices could be used to improve a listener’s experience in noisy environments if they could share their microphone signals for use in an array processing setup. This processing is often referred to as “distributed microphone processing”. While much research is being conducted on different aspects of such processing, no real-world database of recordings is currently available for testing developments from this research.

We will report on a recently-recorded database for use in testing distributed microphone processing. We positioned 24 microphones in various locations at a central 4-person table in a large room, and recorded the microphone outputs while 4 individuals (“target” talkers) both read from a list of sentences in a constrained way and also maintained a natural conversation for several minutes. This was done in the quiet and in the presence of 8-, 24-, and 56-talker babble. The babble comprised real talkers surrounding the central table at various distances. We also recorded without the 4 target talkers active in each
of these different conditions, and used a loudspeaker to measure impulse responses to the microphones from each target talker location and other positions in the room. The microphones were positioned at 12 behind-the-ear locations spread across all of the target talkers and a KE-MAR mannequin (placed at the target-talker table), 8 locations on the table top, and 4 “close-talk” locations, approximately 1 cm from each target talker’s mouth. It is thus expected that we have ground-truth information on various aspects of the actual speech of the target talkers, the non-target sources, and the transfer functions between the target talker locations and the microphones. Such information is essential for properly testing the microphone processing. Preliminary analyses of the recordings and of a video of the recording session indicate the capture of real-world effects such as conversational turn-taking, talker movements, and background-dependent Lombard speech, which can be difficult to accurately produce in simulated signals. These and other relevant characterizations of the recordings will be presented and discussed.

A8
Effects of dynamic-range compression on temporal modulation transfer functions and modulation depth discrimination of normal-hearing and hearing-impaired listeners
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Some of the challenges that hearing-aid listeners experience with speech perception in complex acoustic environments may originate from difficulties in temporal processing of sounds. Such difficulties might arise as consequence of a damaged auditory system (e.g. loss of hair cells) or as consequence of signal processing in the hearing-aid. These issues may compromise relevant information submitted to the retro-cochlear auditory system. Temporal processing is assumed to be related and crucial for robust speech perception in complex acoustical environments.

One measure of temporal processing is temporal amplitude modulation processing. The ability to detect amplitude modulation can be assessed from temporal modulation transfer functions (TMTFs), where the minimum modulation depth of a signal required to detect a modulation is shown as a function of modulation frequency. Data obtained in hearing-impaired (HI) listeners generally reflect wider TMTFs, possibly due to broader auditory filters, compared to normal-hearing (NH) listeners. Another measure to assess modulation processing is the discrimination of modulation depth at a given modulation frequency and reference modulation depth. It was hypothesized that HI listeners have lower modulation discrimination thresholds at intermediate sound pressure levels compared to NH listeners as consequence of reduced auditory peripheral compression.

Most studies compare modulation processing in NH and HI listeners using increased sound pressure levels for HI listeners to compensate for the loss of sensitivity. It remains, however, unclear to which extent the hearing-aid signal processing affects the performance of the HI listeners on the same task. The focus of the present study is to investigate the consequences of wide-dynamic range compression (WDRC, used to account for loudness recruitment) on amplitude modulation perception.

Temporal modulation transfer functions (TMTFs) and modulation depth discrimination thresholds were obtained using sinusoidal carriers at 1 and 4 kHz and modulation frequencies between 8 and 256 Hz for NH listeners and HI listeners with a mild to moderately-severe cochlear hearing loss. To estimate the impact of WDRC, TMTFs and modulation depth discrimination thresholds were measured for both groups of listeners with and without WDRC. The HI listeners were compensated for their hearing loss by a) linear amplification and b) WDRC. The NH listeners were tested with a) unprocessed stimuli and b) with WDRC.

It is hypothesized that after compensation for the hearing loss of HI listeners, the WDRC will influence the TMTFs and the modulation discrimination thresholds similarly for HI and NH listeners, allowing disentanglement of the effects of
broadened auditory filters and the perceptual consequences of distortions introduced by the WDRC.

**A9**

**Informational masking in complex real-world environments**

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The extent to which informational masking (IM) is involved in real-world listening is not very well understood. In literature, IM effects of more than 10 dB are reported, but the applied speech intelligibility experiments typically use simplified spatial configurations and speech corpuses with exaggerated confusions. However, Westermann (WASPAA 2013) considered a realistic cafeteria environment and could not find any significant involvement of IM in normal-hearing listeners, except when the target and maskers were both co-located and the same talker. The present study further investigates IM in such an environment, specifically considering the effect of hearing impairment and nearby distracting maskers.

Speech reception thresholds (SRTs) were measured with 16 normal hearing (NH) and 16 sensorineural hearing impaired (HI) listeners by presenting Bamford-Kowal-Bench (BKB) sentences in a simulated cafeteria environment ($T_{30} = 0.6$ s). The environment was reproduced by a 45 channel 3D loudspeaker array in an anechoic chamber. The target was simulated at 2 m and 4 m distance directly in front of the listener. Three different masker configurations were considered: (1) seven dialogues evenly distributed in the cafeteria (2) two monologues presented from a distance of 1 m, at either +/- 12.5° or +/- 62.5° in reference to the listener, and (3) a combination of (1) and (2). The contribution of IM was measured as the difference in SRT between maskers with different talkers than the target (real-life level of IM) and a vocoded version of each talker (minimum IM).

No significant IM was found for the seven dialogue masker alone, confirming the results of Westermann (2013). Once the nearby maskers were included substantial IM was observed for both NH and HI listeners. This results was consistent both with and without the seven dialogues, however in the latter, SRTs improved because of gap listening. These results indicate that the involved IM of nearby maskers relates to distractions (listener’s attention is captured) rather than target/masker confusions (listener’s cannot distinguish target and similar masker).

The present study shows that IM is less prevalent in real-life listening compared to measured effects in other listening tests. However, where Westermann (2013) only observed IM in idealized (unrealistic) configurations, significant IM was found for both NH and HI listeners when nearby maskers were introduced. These results suggest a distance dependant prioritization of sound sources in complex scenes and that especially the effect of nearby maskers is important when considering IM in real-life listening.

**A10**

**Measuring spatial auditory awareness in realistic multi-talker environments**

Tobias Weller$^{1,2,3}$, Jörg Buchholz$^{1,2,3}$ and Virginia Best$^{2,3}$

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Both hearing instruments and hearing impairment often reduce the ability of a listener to spatially analyze their surrounding environment. Problems include the internalization of sound sources, less accurate localization and reduced ability to make sense of an acoustically complex scene. Nevertheless there are very few studies that analyze spatial hearing in such realistic conditions.

Virtual acoustic environments (VAEs) have the potential to faithfully reproduce complex acoustic situations that occur in everyday life. In this study, a large room with several tables and
chairs (similar to a cafeteria) was first simulated using room acoustic simulation software and then reproduced using a loudspeaker array in an anechoic chamber. 56 different scenes in the virtual room were generated containing between two and seven concurrent talkers at different positions throughout the room.

Six normal hearing subjects listened to 30 seconds of each scene. During these 30 seconds subjects had to determine the total number of talkers, their sex and their position in the virtual room on a top-down drawing of the room on a sheet of paper.

The total number of concurrent talkers was always determined correctly if there were 3 or less talkers in the scene. The performance then dropped to 86 % correct for scenes with 4 talkers and then 66 %, 39 % and 8 % correct for scenes with 5, 6 and 7 talkers respectively. The positions of the talkers within the virtual room were estimated better for scenes with less talkers and scenes where talkers were spread more widely across the room. The distance between talker and listener was more often overestimated than underestimated.

We were able to show that experiments in VAEs can yield meaningful results. Moreover, the described methods and results will help in the development of more realistic listening tests that assess spatial hearing in hearing impaired and aided hearing impaired subjects.

A11

Applications of universal design for hearing in hearing aid users

Vinay Nagaraj1,2 and Arne Vik1

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Hearing loss affects millions of people globally and is one of the fastest growing sensory related chronic disabilities. Present day strategies for rehabilitation often include providing adaptive devices to individuals with hearing impairment that are targeted primarily at the person. Hearing aid users are exposed to a variety of acoustic signals in their everyday situation and one of the factors responsible for obtaining benefit from a hearing aid is the nature of the listening environment. A common issue that many hearing aid users complain is, difficulty understanding speech in noisy conditions. A mismatch between an individual with a hearing disability and the environment can create or aggravate barriers. Hence, continuing efforts are needed to promote a uniform listening environment that alleviates the communication difficulties experienced by most hearing aid users. The concept of universal design for hearing focuses on the design and composition of an environment that can be accessed and used by most people, both with and without hearing loss, to the greatest possible extent. The design of an environment that is flexible and meets the needs of a wide range of users can eliminate or minimize communication difficulties for a person with a hearing disability. In accordance with the concepts of universal design, the International Classification of Functioning, Disability and Health (ICF) system propagates a combination of biomedical and social paradigm to understand human abilities from a multidimensional viewpoint. Furthermore, the concept of a uniform approach is increasingly accepted within the clinical audiological setting in the rehabilitation of individuals with hearing loss. This serves as a catalyst for having an integrated hearing environment that promotes an auditory experience for individuals with a wide range of hearing abilities. The present study highlights the importance of considering universal design for hearing approach in the promotion of a hearing environment that is suitable for most people, especially in users of hearing aids.

A12

Towards measuring the Speech Transmission Index in fluctuating noise: Impulse response measurements

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**Introduction:** The concept of the Modulation Transfer Function (MTF) in the field of room acoustics was introduced by Houtgast & Steeneken (1973). The MTF can be used to predict speech intelligibility in stationary noise and reverberation and can be expressed in one single value: the Speech Transmission Index (STI). The drawback of STI measurements using the classical, direct method that it is time-consuming and that it is not validated for fluctuating background noise (IEC60268-16, 2011). However, the MTF as a result of reverberation can also be calculated “indirectly” using impulse response measurements (based on Schroeder, 1978). Furthermore, when calculating the STI per time segment and applying the same time averaging as used by Rhebergen et al., 2006 (regarding the Extended Speech Intelligibility Index), the “Extended” Speech Transmission Index (ESTI) can be calculated in fluctuating noise. A prerequisite of using the method described here is that the impulse response can be measured reliably in fluctuating noise. In the current study we investigated acoustical conditions under which the impulse response can be measured with sufficient precision to calculate the ESTI.

**Methods:** Impulse response measurements were done by playing, recording and deconvolution of an exponential sweep signal. Background noise of a single speaker was simulated by playing the ISTS (Holube et al. 2010) through a loudspeaker. Experiment 1 was conducted in a room with variable absorption, different levels of background noise, and a fixed sweep level. Experiment 2 was conducted in a room with fixed absorption and background noise level, but with different sweep levels. Besides the 2 experiments, simulations with 6 different types of fluctuating noise were done in order to extrapolate the experimental findings to other acoustical conditions.

**Results:** The experiments and simulations showed that a minimum broadband SNR of -5 dB in stationary noise was necessary to reliably estimate the ESTI in 90% or more of the cases (a deviation of 0.015 STI units was allowed, based on IEC60268-16, 2011). In fluctuating noise a SNR of +5 dB was needed to reach the 100% criterion.

**Discussion and conclusion:** The ISTS was chosen as a measurement signal due to its single talker characteristics. The study could be extended by using other noise types. We found that, when measuring the impulse response, a broadband SNR of +5 dB was necessary to accurately calculate the ESTI. This is sufficient to perform the measurements in regular surroundings like classrooms and office floors.

**Individualized measures of suprathreshold hearing deficit**

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Loudness is a manifestation of nonlinear suprathreshold perception that is altered when cochlear damage exists. Loudness perception can be a factor in hearing-aid acceptance and comfort (Shi et al., 2007), yet loudness measures are not typically used in hearing-aid fittings and evaluations. In this study, we evaluate the reliability of a current loudness measurement (categorical loudness scaling, CLS) and propose methods for using these data to develop individualized measures of suprathreshold hearing deficits.

A variety of loudness measures exist, but may require training in order to achieve reliable results. CLS measurements are efficient and provide ecologically-valid results, but are not widely accepted due to concerns with reliability and how to best apply loudness clinically. In this study, CLS was measured in 61 normal-hearing (NH) and 84 hearing-impaired (HI) listeners at 10 octave and interoctave frequencies (0.25-8 kHz). Test-retest reliability was assessed in the NH listeners, for whom correlations ranged from 0.92 to 0.98, depending on frequency. We refine the description of NH loudness perception by showing how growth of loudness systematically changes with audiometric threshold, even within the range of normal thresholds. We also describe a method for converting categorical loudness data (categorical units) into loudness level (phons). For a pure tone, the loudness level in phons is defined as the intensity (dB SPL) of an equally-loud 1 kHz tone. The CLS data, after conversion...
to phons, were plotted as equal-loudness contours (ELC) for NH and HI listeners. For NH listeners, ELCs were roughly equally spaced as level increased, shifting by about 10 dB for every 10-phon loudness level increment, demonstrating a pattern similar to the one described in the ISO standard. For HI listeners, the spacing between ELCs decreased as the degree of hearing loss increased, which is a manifestation of loudness recruitment.

Despite the orderly nature of ELCs in group data, individual HI listeners with matched thresholds can show ≈20 dB of variability in terms of loudness perception. These individual differences may explain why averaged loudness measurements can provide variable benefits in hearing-aid fittings. As a first step towards addressing this variability, we propose an approach that provides individualized estimates of suprathreshold hearing deficits from loudness data. We show how individualized measurements can be used to set level-dependent gain and applied to a novel hearing-aid signal-processing scheme (see poster by Rasetshwane et al., this meeting). [Work supported by NIH R01 DC2251, R01 DC8318, P30 DC4662.]

A14

Adults with mild hearing impairment: Learnings from a clinical database

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In recent years, the current trend to resort to Acquired hearing impairment is recognized by the World Health Organization as the third leading cause of disability, with a mild impairment being the most prevalent. Despite the mild degree of loss, such impairment can have significant negative effects and can represent clinical challenges in audiological practice.

Population questionnaires have shown that hearing care professionals are reluctant to fit hearing aids to adults with a mild hearing impairment and some professional literature suggests fitting hearing aids to this clinical population may not be viable. This presentation will report on current fitting practices with clients with mild hearing impairment compared to those with a greater degree of loss.

To collect data, participating hearing care clinics enabled a logging function in the hearing aid programming software, which then created a data store of all subsequent client fittings which were regularly transferred to a central server. The data, with all personal information except age and gender removed, logged the client audiogram, the selected hearing aid characteristics, hearing aid usage, as well as the clinician’s fitting and fine-tuning changes applied to the device.

This presentation will report on the resulting data from over 9,000 adults bilaterally fitted with hearing aids during 2012 and 2013 in three countries. Of these, over 1,500 had a four frequency average hearing loss (4FAHL) of between 25 and 40 dB HL in both ears. Their fitting data will be compared to the almost 5,000 adults with a 4FAHL of 40 to 69 dB HL in both ears. Data analysis of the differences in fitting approaches will be presented.

Enhanced understanding of the clinical challenge adults with mild hearing impairment present in audiological practices may go some way to improving outcomes for this clinical population.

A15

Hearing aid fine-tuning based on Dutch descriptions

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Since the introduction of the first ear-friendly hearing instrument, many changes have occurred in the amplification of sounds within a hearing aid. From the first transistors until today’s modern digital hearing aids, a substantial increase in system complexity is obtained. This increase however, turned hearing aid fitting to a complicated task. The process of fine-tuning is most...
frequently performed using trial-and-error. The hearing aid fitter uses his or her experience to adjust the parameters of the aid, based on complaints or remarks of the hearing aid user. Inexperienced fitters on the other hand cannot rely on their experience so a fitting assistant could be useful. All hearing aid manufacturers develop a fitting assistant based on their own technology, so advice in parameter adjustment between the different manufacturers is not interchangeable.

The aim of this study was to create an independent fitting assistant based on expert consensus. Two questions were asked. (1) what (Dutch) descriptions or terms do hearing impaired listeners nowadays use to describe their perception of specific hearing aid fitting problems? And (2) what is the expert consensus on how these complaints can be solved by hearing aid parameter adjustments?

Using a questionnaire, 112 hearing aid dispensers gave one or more descriptors that impaired listeners use to describe their reactions to specific hearing aid fitting problems. In total 387 different descriptors were identified.

Fifteen hearing aid fitting experts were selected and asked ‘How would you adjust the hearing aid if its user reports that the hearing aid sounds …?’ using the 40 most frequently mentioned descriptors in the blank position. In order to obtain the best possible solution for each descriptor, the responses from all experts were weighted. The solution with the highest weight value was considered to be the best (and first) possible solution for that descriptor.

Principal component analysis was performed to find the correlations between the different descriptors and determine the underlying fitting problems. Nine fitting problems could be determined within all descriptors, resulting in an expert-based, hearing aid manufacturer independent, fine-tuning fitting assistant for clinical use.

Many people who wear hearing aids have been complaining about the comfortability of their own voice and/or the mastication sound. These discomforts are posed by increasing the sound pressure in the low frequency when the ear canal is blocked by the hearing aid itself. It has been generally recognized that the increased sound pressure below 300 Hz indicates about 15 dB on average and about 30 dB at a maximum. The cause is often referred to as “occlusion effect”, that is one of the critical issues for hearing aid users. There has been an attempt to reduce the effect by installing a small pipe in the hearing aid and venting the internal air to the outside, however, CIC or ITE hearing aids are hard to make the vent which has sufficient internal diameter due to its small size. In order to resolve the occlusion-related problems, several systems which can reduce the increased sound pressure have been proposed. However, the amount of the occlusion reduction gain has obtained no more than about 20 dB, and that amount of the gain was insufficient to resolve severe cases.

The aim of this study is to develop an occlusion reduction system for the hearing aid, which can be regarded as an electrical embodiment of an acoustically transparent state in the ear canal completely blocked by the hearing aid. In this report, an occlusion reduction system using feedback active noise control (ANC) technique with two types of adaptive filter is proposed and simulated on a PC. The configuration of the proposed system is incorporated in an ITE hearing aid, and consists of a microphone (external), a receiver and digital signal processor for the general purpose of a hearing aid, and a microphone (internal) for the purpose of picking up an occluded sound in the canal. Other than the traditional Filtered-x LMS algorithm, the fast H-infinity filter (FHF) is employed as the algorithm of adaptive filter in order to improve the optimization for the non-stationary signal such as speech.

From the results of the simulation, the system with the FHF algorithm gives the gain of the reduction up to more than 30 dB, which indicates about 15 to 20 dB larger than the case of using the LMS algorithm. These results suggest that the system using the FHF algorithm reduces occlusion sufficiently even for people who have severe occlusion effect.

An occlusion reduction system for hearing aids based on the ANC technique with fast H-infinity filter

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Classification of pediatric hearing loss using behavioral measures of early prelingual auditory development

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Initial classification of pediatric hearing loss in young children is achieved by measurement of the auditory brainstem response (ABR). However, such measurements may be difficult or impossible to obtain in some instances because of the child’s compliance with the protocol, or because ABR instruments are unavailable. Without an initial classification of the child’s hearing loss, selection of appropriate early intervention options becomes more difficult.

This research addresses these issues through the use of behavioral measures of early prelingual auditory development (EPLAD) as an initial means of classifying the severity of pediatric hearing loss. EPLAD was assessed using the Chinese language version of the Infant-Toddler Meaningful Auditory Integration Scale (ITMAIS) (Zheng et al., 2009). The normal developmental trajectory for EPLAD has been established, allowing comparison of the child’s level of EPLAD with that of normally hearing children.

Tone-burst ABR measurements and ITMAIS scores from two samples of hearing impaired children (N = 139 and 133) under 5 years of age were included in the analyses. The first sample was used to develop rules for classifying severity of hearing loss based on ITMAIS scores, and the second sample was used to cross-validate these classification rules. PTAs were estimated from the better-ear ABR measurements in the first sample and used to classify subjects’ hearing loss as mild, moderate, severe, or profound. The distribution of ITMAIS scores and PTAs was used to define a criterion ITMAIS score to distinguish severe and profound losses, and a second criterion score to distinguish mild-moderate and severe losses. Separate sets of criterion scores were defined for children less than 3 years of age and for children over this age. Children in the cross-validation sample were classified in the same manner according to their PTAs, and according to their ITMAIS scores using the rules developed with the first sample. Classification rules were evaluated by calculating the Sensitivity, Specificity, and Accuracy for each rule in the cross-validation sample.

Results showed that EPLAD trajectories for each hearing loss group resembled the normal trajectory, but were shallower and did not reach the same levels. Classification Accuracy averaged 92%, with average Sensitivity of 94% and average Specificity of 77%. Reduced Sensitivity causes misclassification as a less severe hearing loss, while reduced Specificity causes misclassification as a more severe hearing loss. High Sensitivity, as exhibited in the current results, may be of greater clinical significance. Thus, the current results demonstrate the potential clinical utility of EPLAD measures for initial classification of the severity of pediatric hearing loss when ABR measures cannot be obtained.

Preference judgments in the field and in the laboratory

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Currently, there is a certain focus in the hearing-device research community on how more realistic laboratory tests should be constructed. Another question is how we should collect reliable data from the field.

In this study two hearing-aid gain settings were compared in two hearing-aid programs. Twenty participants with impaired hearing evaluated the settings with regard to preference, both during a two-week field trial period and in a subsequent laboratory test. The study was designed as a double-blind trial, where neither participants nor test leaders knew which setting was used in each program.

In the field, the following outcome measures were used:
- A diary where the participants logged preference for hearing-aid setting 1 or 2 in various listening environments.
A structured interview, where overall preference was investigated.

A questionnaire, answered by the participants together with the test leader, focusing on preference in a number of pre-defined and self-selected listening situations.

A hearing-aid log, where usage time and volume control settings could be studied for the two programs in a number of listening situations.

In the laboratory, the following preference outcome measure was used:

A paired-comparison test with ratings of the difference between the two hearing-aid programs. Preference, speech clarity, and comfort were compared for the two programs in a number of listening situations. The data can either be analyzed as “strict” paired comparisons or by including the rating data. Test-retest data were collected for the paired comparisons of preference.

Data collection is on-going. In the presentation, the various types of preference data will be compared. A particular focus will be on methodological issues such as time consumption, face validity, and comments from the participants and test leaders.

A novel speech intelligibility test is presented, in which individual test subjects can be evaluated in different conditions with varying difficulty, such that the test signal-to-noise ratio (SNR) is the same for all subjects. Thus, so-called SNR confounds can be avoided. Furthermore, the test allows selecting the test SNR according to considerations of auditory ecology. The SFS test is intended for comparisons, and thus the outcome measure is a pair of %-correct scores, e.g. for two hearing-aid settings. The SFS test uses HINT target sentences presented against two running speech maskers, and translation of the test to other languages should be straightforward. Test difficulty is varied by selecting among different combinations of target/masker spatial separation, female/male masker talkers, and sentence/word scoring.

In the present experiment carried out in a sound-treated listening room, a within-subject comparison was made between a standard prescribed hearing-aid setting with full audio bandwidth and a setting with low-pass filtering introduced at 2.5 kHz. 19 subjects were tested in their individual SFS condition, such that the best performers were tested in the most difficult SFS condition and vice versa. The results showed, as expected, that speech scores were higher with full bandwidth – by about 15% with high statistical significance. All %-correct scores were in the 29-91% range with the vast majority of data points within the preferred 40-80% range. Thus, floor and ceiling problems were not an issue, in spite of the fact that all listeners were tested at the same SNR of +3 dB.

The difference in difficulty among the four SFS test conditions was measured in an adaptive speech reception threshold (SRT) paradigm, and compared to the nominal 2.5-dB steps. Also, test-retest standard deviation was directly measured – to be 8.5%. This compares favorably to the test-retest properties of the standard HINT test. Apparently, spatial separation of target and maskers, speech maskers, and different scoring rules were introduced without any penalty on test-retest variance, as compared to the standard HINT test where steady-state noise is presented co-located with the target.

Finally, a comparison was made to a previous validation experiment conducted in an anechoic room. In that experiment, similar effects of the SFS conditions and test-retest properties were observed, whereas the overall test difficulty was found to be greater in the listening-room environment – by about 3 dB in terms of mean SRT. This is probably due to the reverberation present in the listening room.
Intelligibility improvement of noise reduction algorithms: Does the presentation of positive signal-to-noise ratios help?

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Speech-in-noise tests are required that can discriminate across settings of different hearing aid algorithms. For various reasons, it is an advantage for these tests to perform at positive signal-to-noise ratios (SNRs): First, several hearing aid algorithms (for example single microphone noise reduction algorithms) are most effective at positive SNRs. Second, most conversations in everyday life take place in noisy environments at positive SNRs, i.e., speech levels are higher than the noise level. In addition, for the same intelligibility hearing-impaired listeners need a higher SNR than normal-hearing listeners. So, testing of hearing aid algorithms with normal-hearing listeners should be done at hearing-impaired listeners’ settings with higher SNR values. Nevertheless, speech-in-noise tests like the Hearing in Noise Test and the German Oldenburg or Goettingen sentence tests simply present speech in a stationary background noise and typically yield negative speech recognition thresholds (SRTs, SNR of 50% intelligibility). Thus, the presented SNR range for a sensitive test is limited even for hearing-impaired listeners and for example varies between about -9 dB and about -1 dB for the Oldenburg sentence test. However, a recently investigated procedure can shift the SNR range towards fixed positive values and raises the difficulty of a speech-in-noise test by compressing the speech material in time. This test increases adaptively the speech rate in ten possible steps with a 1up-1down method. Eleven hearing-impaired listeners conducted the test and their individual speech rate was determined, where listeners reached approximately SRT at 1 or 5 dB SNR. Then, intelligibility of the individually time-compressed sentences was measured with and without two different single microphone noise reduction algorithms at positive SNRs. One algorithm was a Wiener filter, which applied apriori knowledge of the noise. A realistic algorithm used minimum recursive averaging for the noise estimation and spectral subtraction. An objective measure confirmed an SNR improvement after noise reduction for time-compressed speech. Speech intelligibility tests showed improved scores with the Wiener filter. No improvement was measured with the realistic algorithm which confirms earlier studies. In summary, time-compressed speech presented at positive SNRs can show the improvement after single microphone noise reduction processing. Also, it can help to present comparable SNRs for different listeners in a speech-in-noise test.

Common part modeling of acoustic feedback paths in open-fitting hearing aids

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The number of hearing impaired persons supplied with open-fitting hearing aids has been steadily increasing in recent years. Although open-fitting hearing aids largely alleviate problems related to the occlusion effect, they suffer from an increased risk of acoustic feedback. Therefore, robust and fast-adapting acoustic feedback cancellation (AFC) algorithms are required. The most promising approach for AFC is the use of an adaptive filter to model the acoustic feedback path between the hearing aid receiver and microphone. It is known that the convergence speed and the computational complexity of the adaptive filter are determined by the number of adaptive parameters. In order to increase the convergence speed, the adaptive filter can be decomposed into a convolution of two filters: a fixed filter accounting for parts common in various feedback paths, e.g., microphone and receiver characteristics and individual ear canal properties, and a shorter variable filter enabling to track fast changes. Recently, we proposed different estimation procedure to estimate a common pole-zero filter and variable all-zero filters from a set of acoustic feedback paths [1,2].

The focus of the present contribution is twofold. Using acoustic feedback paths measured on a
dummy head with adjustable ear canals; first, the estimation procedures presented in [1] and [2] are compared to each other in terms of modeling accuracy and maximum stable gain. Results show that the estimation procedure in [2] outperforms the previously proposed method in [1], yielding improvements in modeling accuracy of up to 7dB. Second, we present results evaluating the perceived quality of the proposed decomposition for both estimation procedures. For a variety of common part and variable part settings the speech quality is evaluated using both objective measures (PESQ) and a listening experiment (MUSHRA).

References:

A22

Comparison of different forms of frequency lowering in different forms of frequency lowering in digital hearing aids for people with dead regions in the cochlea
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When high-frequency hearing loss is associated with extensive dead regions (DRs) in the cochlea, i.e. regions of the basilar membrane with non-functioning or very poorly functioning inner hair cells and/or neurons, high-frequency amplification via a hearing aid often fails to provide benefit for speech intelligibility. Frequency-lowering hearing aids, which transform high-frequency sounds into lower-frequency sounds, may be used to provide information from high frequencies to people with high-frequency DRs. However, there have been few studies of the effectiveness of frequency-lowering hearing aids for people with confirmed extensive DRs. Here, two frequency-lowering prototypes based on a commercially available digital hearing aid (Phonak Exélia Art P) were evaluated. One of the prototype hearing aids used frequency transposition, i.e. frequency components in a high-frequency band were recoded by decreasing the frequency of each by a constant amount in Hz. The other prototype hearing aid used frequency compression, i.e. the downward frequency shift of components in a high-frequency band increased with increasing frequency above the lower edge of the band. Ten subjects with steeply-sloping hearing losses and extensive DRs took part in a single-blind, three-period, three-condition cross-over design trial whose aim was to evaluate the effectiveness of frequency-lowering hearing aids for speech perception. Conditions were: (1) Conventional amplification over the widest frequency range possible with no frequency lowering; (2) Frequency transposition; (3) Frequency compression. The edge frequency \( f_e \) of the DRs ranged between 0.5 and 2 kHz, and was below 0.75 kHz for five of the subjects, and below 1 kHz for seven of the subjects. The two frequency-lowering schemes were fitted taking into account the characteristics of the DRs of the subjects, which led to frequency lowering at frequencies below those used in previous research and in current clinical practice. Outcome measures were: 1) Identification of consonants in nonsense vowel-consonant-vowel (VCV) combinations; 2) Detection of word-final \([s]\) and \([z]\); 3) Speech reception threshold in noise (SRT); 4) Responses on the Speech, Spatial and Qualities Questionnaire (SSQ). Frequency lowering did not provide a significant benefit for any of the outcome measures relative to the control condition. However, the use of frequency lowering may provide some practical advantages, such as reduced problems with acoustic feedback and lower output level requirements.

A23

On electroacoustic assessment of remote microphone devices/systems
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Hearing Assistive Devices/Systems (HADS) encompass a broad array of instruments that facilitate enhanced listening in challenging environments. Remote microphones (RMs) are a sub-
category within HADS, which wirelessly transmit the desired acoustic signal to the hearing aid (HA). Several RM systems are currently available on the market, each with its own set of transmission protocols, and signal processing algorithms for coding/decoding, noise reduction etc. Fitting and verification of RMs, and benchmarking the relative performance of different RM devices in varied acoustic environments are of significant interest to Audiologists and RM developers. Current AAA guidelines for RM performance assessment are based on the concept of transparency, which does not allow comparison of two RMs that are transparent but are potentially perceived in a differently by the RM user. The present study therefore investigates the application of instrumental speech intelligibility and quality metrics for characterizing the RM performance.

In this ongoing project, electroacoustic measures of RM performance are obtained in a sound-treated booth and a reverberation chamber (RT60 = 0.76 s). In both these environments, two mannequins are placed 1 m apart, with one mannequin with built-in mouth simulator served as the “talker” and the other with ear mold simulators served as the “listener”. A separate noise-generating speaker is placed at 90° azimuth relative to the “talker” mannequin. Four RM systems are tested, two of which utilize the newer digital transmission protocols, while the other two rely on analog frequency modulation (FM) transmission. For each RM, the transmitter microphone is placed near the mouth simulator and the receiver is coupled to a HA through the direct audio input. The HA itself is programmed to match the prescriptive targets for the “N4” standard audiogram and the same HA is used for interfacing with all four receivers. After transparency is verified with each RM, the HA output for speech-in-noise stimuli is recorded for different noise types (stationary and multi-talker babble) and SNRs (measured at the HA microphone location). The HA recordings are then used to extract a set of speech intelligibility and quality metrics including: short-term objective intelligibility (STOI), hearing aid speech perception index (HASPI), hearing aid speech quality index (HASQI), Perception Model-Quality (PEMO-Q), and speech-to-reverberation modulation ratio (SRMR-HA). Preliminary results show considerable variability in the metrics across the four RMs. Clinical implications and applications of these results will be discussed.

Characteristics of regular users of a smartphone hearing app

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Smartphones, when used with earphones, have the ability to function as amplification devices. Ambient sounds are picked up by the phone’s microphone, processed on the device, and output via earphones, all in real time. The apps that facilitate this function -- hearing apps -- are becoming increasingly popular and numerous. Recent research indicates that some apps can be as effective as hearing aids. Despite their prevalence, is currently unknown what types of individuals seek out and benefit from such apps. Here we begin to examine this question using our free hearing app for Apple’s iOS platform, EarMachine. In addition to using the app for amplification, users can also submit their age, gender, and self-reported hearing status to our servers. In previous work, we reported the demographics of all users who have downloaded EarMachine. However, that analysis included one-time users, and therefore does not answer the question of who are regular users. Here we combine the data from questionnaires with usage statistics (such as session duration and frequency) to identify the characteristics of regular hearing app users. Our initial analysis indicates that the peak age of regular users is closely matched to the average age of first time hearing aid purchasers. However, the age distribution of regular app users has a considerable skew toward the younger ages. Further, for each user we computed an estimated audiogram by identifying audiograms of the listeners in the NHANES database who had similar questionnaire answers. As the severity of the estimated audiogram worsened, listeners were more likely to become regular app users. Overall, it appears that regular hearing app users have a similar, but slightly younger, profile to that of first-time hearing aid purchasers. This profile might indicate
that hearing apps provide individuals a risk-free way to evaluate whether they would benefit from amplification. [Work supported by NIDCD grant R44 DC013093.]

Word recognition in sentence context: Individual differences and working memory
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Purpose: The goals of this study were a) to develop and pilot a test of the contribution of sentence context to word recognition, b) to determine the association between context-use and working memory. (The long term goal of this work is to develop a clinically viable test of sentence context use to aid in planning and outcome assessment of Aural Rehabilitation).

Method: Subject were eight young hearing adults. Word recognition was measured, as a function of Signal-to-Noise Ratio, using meaningful and nonsense sentences, each 4-words long, presented in sets of 10. Working memory was measured in terms percent words recalled during a reading span test. Both tests were administered using custom software.

Results: Group mean performance vs. SNR functions showed the expected benefit of context. At 50% recognition, the meaningful sentence context was equivalent to a 3dB reduction of noise level.

When data were collapsed across four SNRs, and compared in terms of the k-factor (the ratio of the logarithms of the two error probabilities), the result was a k-factor of 2.2. (Note that the k-factor is mathematically identical to the proficiency factor of Articulation Index theory). The value of 2.2 indicates that in these materials, and these listeners, the sentence context effect was equivalent to roughly doubling the independent sources of information about word identity. This value is similar to those reported in the literature.

There were large individual differences in noise-masked word recognition regardless of sentence type. In fact the range of scores was virtually identical (at around 30%pts). And the correlation between the two was very strong (p = .0003).

There were large individual differences in reading-span, but reading span did not predict performance on either type of sentence. It did, however, predict the difference between the two word-recognitions scores, as evidence by a significant correlation between reading-span and k-factor (p = .02). Listeners with poorer reading-span scores tended to have the higher k-factors. In other words they showed increased use of, or dependence on, sentence context.

Conclusion: These results strongly suggest that individual differences of working memory can influence the benefit derived from sentence context. Actual performance in meaningful sentences, however, appears to depend much more on individual noise susceptibility.

Auditory event-related potentials outcomes with ReadMyQuips auditory training program
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The goal of this study was to document effectiveness of ReadMyQuips (RMQ), a structured auditory training program, using electrophysiological and behavioral outcome measures. Participants were adults with hearing loss in the mild to moderate range who are first-time hearing aid users. RMQ is an audio-visual tutorial designed to improve speech perception in everyday noisy listening environments. As the individual makes progress in therapy, the training program uses a hierarchical approach to make listening more difficult by reducing the signal-to-noise ratio. Participants receive immediate feedback and are rewarded with points for successful listening. A fundamental cognitive ability linked to listening in noise is selective attention. We hypothesize that training using RMQ will lead to enhanced auditory selective abilities. An auditory selective attention paradigm is being used to collect event-related potentials (ERP) data. Selective attention refers
to the ability to suppress irrelevant information and focus on relevant signals in the environment, a cognitive skill of tremendous importance for everyday living and learning. Behavioral outcomes include signal detection measures collected during performance of the selective attention task.

Participants were divided into two groups: (1) experimental and (2) control. Participants in the experimental group completed the RMQ program after 4 weeks of amplification use. Participants in the control group received listening training using audiobooks after the same period of hearing aid use. ERP and behavioral data were collected before hearing aid fitting, after 4 weeks of hearing aid use and after 4 consecutive weeks of training. The auditory selective attention paradigm utilizes an oddball paradigm allowing us to collect ERP data to track changes in P3b and P3a cortical evoked potentials to targets and distractors, respectively. After 4 weeks of amplification use, signal detection measures obtained while participants performed the selective attention task did not show a statistically significant change. However, correlation statistics showed that an increase in P3b was associated with improvement in data from pretest to after amplification use. After training this correlation remained in the experimental group, but not in the control group, suggesting that the RMQ training program targeted specific attentional mechanisms. The poster will include results from the two groups and also outcomes after training.

**A27**

The effects of WDRC and reverberation on syllable identification

Paul N. Reinhard1, Pamela E. Souza1,2, Nirmal Kumar Srinivasan2,3 and Frederick J. Gallun3,4

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2 Knowles Hearing Center, Evanston, IL
3 National Center for Rehabilitative Auditory Research, Portland, OR
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Reverberation is a common feature of everyday listening environments; however, the interaction between hearing aid processing and reverberation has not been well studied. A previous acoustic analysis indicates that while fast WDRC does preserve the modulation profile of reverberant speech, it also normalizes the modulation profile across all speech sounds making all syllables more similar to one another (Srinivasan et al., 2014), potentially increasing the chance of perceptual confusion. Conversely, some products use fast-acting WDRC to compensate for reverberation. This study investigates the effect of compression speed for speech intelligibility in reverberant environments.

Participants are older adults with mild-moderate sensorineural hearing loss. Test materials include 16 nonsense syllables (/aCa/ format) processed in two stages. First, stimuli are distorted using virtual acoustic methods to simulate 5 reverberation times (0.0, 0.5, 1.0, 2.0, 4.0s). The acoustic simulator uses an image model to produce binaural room impulse responses that are reasonable physical and perceptual approximations of those measured in real rooms. These reverberation times reflect the full range of realistic listening environments. Second, each token is processed by a WDRC simulator using 4 release times (12, 90, 800, 1500ms). Similarly, these release times reflect the range of processing values found in commercial hearing aids. Stimuli were presented monaurally with individual frequency shaping to provide audibility.

Preliminary results indicate that intelligibility decreases as reverberation time increases. Additionally, initial data suggest that slower release times are better for intelligibility than faster time constants across the range of reverberation times tested; however, the effect is not equivalent for all levels of reverberation. We will discuss how the effects of release time interact with reverberation. [Work supported by NIH R01 DC60014.]

**A28**

Evaluation of a hearing-aid signal-processing strategy for restoration of cross-channel suppression

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Boys Town National Research Hospital
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This study tested a signal-processing strategy for restoration of cross-channel suppression in hearing-impaired listeners. This novel strategy was recently proposed by Rasetshwane et al. [IEEE Trans. Biomedical Eng. 61, 64-75 (2014)]. The strategy unifies instantaneous compression with cross-channel interaction in its calculation of level-dependent gain. The gain at each frequency is dependent on the instantaneous level of frequency components across the entire audible range of frequencies, in a manner that mimics cochlear two-tone suppression. Suppression presumably plays an important role in the coding of speech and other complex stimuli, and has been suggested to result in the enhancement of spectral contrast of complex sounds, such as vowels. The prescription of gain for this new strategy is based on individualized measurements of loudness using categorical loudness scaling and is intended to restore normal loudness growth. Fifty-three adults with hearing loss (HL) and an additional 30 adults with normal hearing (NH) participated in this study, with data from the NH group used as prescriptive targets for restoring normal loudness growth. Word recognition in quiet and in speech-shaped noise was tested in participants with HL using an adaptive tracking procedure that determined the 30% and 70% points on a psychometric function. Word recognition for the suppression strategy was compared to word recognition for a compression-only strategy obtained by disabling the cross-channel suppression. Comparison was also made to a conventional eight-channel, nonlinear wide dynamic-range compression strategy fit using a method that is currently used in the clinic (Desired Sensation Level algorithm). Processing-strategy preference when listening to sentences and music in quiet was also evaluated using a two-alternative forced-choice task. Word recognition for the strategy with suppression was better than that for the conventional approach for both quiet tests and in one of the noise tests. Equivalent word-recognition performance among the three strategies was observed in the other noise test. Although variable, participants preferred the processing strategy with suppression over the other two strategies when listening to music. Preference for a strategy that uses instantaneous compression is encouraging, considering that previ-ous work has suggested that instantaneous compression has deleterious effects on sound quality. These findings suggest that restoring normal suppression and using individual loudness data to achieve normal growth of loudness may improve speech recognition and user satisfaction. [Work supported by the NIH, R01 DC2251, R01 DC8318, and P30 DC4662.]

A29

Knowledge, attitudes, behaviors and noise exposure of baristas

Alyssa Pursley, Gabrielle Saunders, Susan Griest and Serena Dann

National Center for Rehabilitative Auditory Research (NCRAR)

Objectives: Occupational noise-induced hearing loss is a known hazard in many professions, e.g. building construction, mining and farming. As such, workers are informed and knowledgeable about the need to protect their hearing while at work. There are, however, hospitality industry professions, in which workers may be exposed to dangerously high levels of noise, but worker awareness of possible hazardous noise exposure is low. One such group of individuals is classified by the Bureau of Labor Statistics (2010) as ‘Counter Attendants, Cafeteria, Food Concession, and Coffee Shop.’ The position of baristas is one job that falls into this classification. Baristas are individuals who prepare or serve specialty coffee or other beverages, and serve food such as baked goods or sandwiches to patrons. In 2010 there were an estimated 439,000 such employees, with additional job openings projected to increase by 267,800 by 2022 (O-Net Online, 2010). Baristas are exposed to noise sources from coffee grinders, espresso machines, as well as ambient noise from music and conversation. Ambient noise levels can vary considerably depending on the size and volume of the café, the floor, wall and ceiling surface coverings, the type of furnishings and the level of background music. There are no published data on typical sound levels in coffee shops, but sound levels in restaurants range from 65 dBA (low range) to 90 dBA (high range) across 8 studies (Berger et al., 2013). For machinery specific to baristas, the NoiseNavigator™ database specifies that house-
hold coffee grinders (which tend to be smaller than those used in coffee shops) range in level from 80-95 dBA. These data suggest that baristas may be exposed to potentially damaging noise levels during their work day, but this has not been informally examined. The objective of this study is to measure baristas’ daily noise exposure and to assess their knowledge and attitudes toward hearing and hearing conservation.

Research Plan: This study will involve examining the daily sound pressure levels to which baristas working in coffee shops in the Portland metro area are exposed. In addition to noise exposure levels, knowledge, attitudes and behaviors regarding hearing, noise and use of hearing protection will be assessed. The perceived ambient noise levels in the work environment and the perceived barriers to use of hearing protection at work will also be included in the assessment.

Methods: Up to three baristas from up to ten coffee shops in the Portland metro area will participate. All participants will complete the Knowledge, Attitudes and Behaviors questionnaire (KAB) of Saunders et al. (2013), and a structured interview will be used to document perceptions of noise in the work environment and perceived barriers to use of hearing protection and/or decreasing noise in the coffee shop. Daily noise exposure will be measured using a personal dosimeter. Participants will wear the dosimeter for the duration of three different work shifts. The data will provide information about typical daily noise doses and about attitudes toward noise and hearing. By understanding these issues, it will be possible to understand the needs and address hearing conservation among these employees.

Findings to Date: None, data collection will begin in June 2014. Results will be analyzed, interpreted, and discussed beginning in July/August 2014.

Relevance to VA’s Mission: Occupational noise exposure is a hazard of many jobs that Veterans will encounter on leaving the military. Indeed, in terms of baristas, at least one major coffee shop chain has employment opportunities listed on Veteran employment websites. Awareness of potential noise damage is thus important for Veterans and VA.

MeSH Terms: Hearing Loss, Noise-Induced; Occupational Noise; Health behavior; Health Education; Hearing Protective Devices

References:

A30

Testing the maximum equivalent pressure output and gain before feedback of a light-based contact hearing device
Sunil Puria1,2, Rodney Perkins1 and Peter Santa Maria3
1 EarLens Corporation
2 Stanford University
3 University of California, San Francisco
The EarLens system is a non-surgical investigational hearing device consisting of two components: a tympanic contact actuator (TCA), which is a light-activated balanced-armature transducer that drives the middle ear through direct contact with the umbo; and a behind-the-ear unit (BTE) that encodes amplified sound into pulses of light that are emitted via a light source in the ear canal to wirelessly drive and power the TCA. In comparison to conventional acoustic hearing aids, this approach is designed to provide higher levels of output and functional gain over a broad frequency range; and has a significantly higher maximum gain before feedback (GBF).
The maximum equivalent pressure output (MEPO) of the TCA provides an important guideline for setting BTE light output levels and fitting of the device to a subject’s hearing profile. MEPO varies across subjects, and is in part a function of the distance between the light source and TCA. It was designed to exceed 100 dB SPL at many frequencies. Because MEPO could not be measured in subjects, and given that no suitable artificial middle ear model exists for testing the system, we performed our measurements on fresh human cadaver temporal-bone specimens.

To calculate MEPO and GBF, we measured ear-canal pressure ($P_{EC}$) within 2–3 mm of the eardrum, using a probe-tube microphone; and stapes velocity ($V_{ST}$), using a Polytec HLV-1000 laser Doppler vibrometer; for: 1) the baseline case of sound-driven stimulation (unaided), and 2) light-driven cases using a BTE and custom-shaped TCA.

The baseline sound-driven measurements (e.g., around 0.1 (mm/s)/Pa near 1 kHz) are consistent with previous reports (Rosowski et al., 2007). The overall average MEPO (N=4) varies from 117 to 127 dB SPL in the 0.7 to 10 kHz range, with the peak occurring at 7.6 kHz. Below 0.7 kHz, it varies from 83 to 117 dB SPL. In terms of the average GBF, a broad minimum of about 10 dB occurs in the 1–4 kHz range. Above about 4 kHz, it rises and the overall average reaches 42 dB at 7.6 kHz. Above this, a sharp decline in GBF occurs at around 11 kHz. From 0.2 to 1 kHz, GBF decreases linearly from about 40 dB as the frequency increases.

In summary, the direct-contact photonic hearing device provides high output and high gain margins, which may offer a feasible way of providing broad-spectrum amplification appropriate to treat listeners with mild-to-severe hearing impairment.

Remote online screening of functional hearing ability in occupational medicine

Jennifer Hart¹, Steven Verdooner¹, Sigfrid Soli¹,²
¹ Hearing Test Systems, LLC
² House Clinic, Los Angeles, USA
provide a statistical summary of the screening results to date. The potential for use of this technology in clinical medicine for long-term outcome assessment following hearing aid and/or cochlear implant interventions will also be addressed.

Speech recognition with minimal segregation cues
Niels Henrik Pontoppidan, Marianna Vatti, Lars Bramsløw and Renskke Hietkamp
Eriksholm Research Centre, Rørtangvej 20, DK-3070

Hearing-impaired listeners repeatedly comment that they find competing voices situations difficult. While they recognize and understand one voice, they cannot segregate and understand the exact same voice in competition with another. This study aims to replicate this behaviour by minimizing the segregation cues for normal-hearing listeners and investigate how cues contribute with different weight to segregation and recognition. With noise gated by binary masks, Wang et al measured speech recognition of a single voice, and found that the normal-hearing listeners achieved above 90% word correct scores. However, what happens when the binary mask contains information from two competing voices? This study extends Wang et al. and measures speech recognition for both noise gated with single-voice binary masks and noise gated with competing-voices binary masks. We hypothesise that listeners require more information than the binary mask provides to segregate the two streams, and thus that speech intelligibility will be severely reduced with competing-voices binary mask.

Preliminary results with normal-hearing listeners show that the presence of another voice in the binary mask pattern reduces speech recognition to a third, in fact making the competing-voices condition almost impossible. This suggests that, while the single voice ideal binary mask is sufficient to facilitate speech recognition, the competing-voices binary mask does not carry sufficient cues to facilitate segregation. Furthermore, this suggests that the computational goal for competing-voices speech enhancement is to enable the listener to internally reconstruct individual binary mask representations of each voice. Future research is therefore needed to understand how different listeners, including hearing impaired, utilize segregation cues.

Reference:

Evaluation of the ITU-T P.563 standard as an objective enhanced speech quality metric for hearing aid users
João F. Santos¹, Vijay Parsa², Susan Scollie² and Tiago H. Falk¹
¹ INRS-EMT, University of Quebec, Montreal, QC, Canada
² University of Western Ontario, Electrical and Computer Eng., London, ON, Canada

P.563 is an ITU-T standardized speech quality metric for narrow-band telephony applications. The metric is single-sided, i.e., it estimates speech quality without needing a clean reference signal, and is based on several internal parameters that measure the effect of six dominant distortion classes (background noise, signal interruptions, signal-correlated noise, speech robotization, and unnatural male/female speech) on speech quality. During the design of speech enhancement algorithms for hearing aids (HA), speech quality and intelligibility are usually assessed via subjective tests. These tests are laborious and time-consuming, which hinders their use during algorithm development. Currently, there are no standardized speech quality metrics tailored to hearing impaired listeners, but recent results have shown that P.563 can potentially be used as a speech intelligibility metric for noise-vocoded speech in cochlear implants.

In this study, we investigate the performance of P.563 and its internal parameters as a tool for assessing speech quality for hearing aid users. Two speech quality datasets were used. The first database explores the effects of frequency lowering (more specifically, Nonlinear Frequency Compression, NFC). Quality ratings for different
talkers and NFC strategies were obtained with 11 hearing impaired listeners with severe to profound hearing loss. In the second database, the impact of speech enhancement on perceived speech quality was investigated in noise-only, reverberation-only, and noise-plus-reverberation conditions. Twenty-two adult HA users rated the quality of sentences processed using different noise reduction, adaptive direction microphone, and speech enhancement configurations. Subjective results were then compared to P.563 and its internal parameters by computing their correlation and prediction error. Results show that while the overall mean opinion score computed by the metric has poor performance with enhanced speech, several internal parameters present higher correlations, suggesting that an improved mapping from these parameters to the subjective quality ratings of HA users can be designed.

Ng et al. (2013) showed that noise reduction in hearing aids reduced the adverse effect of irrelevant competing speech on memory for heard speech for hearing aid users. However, the effect was found only for individuals with high working memory capacity. The main aim of the study was to investigate whether the effect of noise reduction on memory for audible speech would generalize across individuals with a range of cognitive abilities if the memory task was simplified by reducing the number of to-be-remembered items. A further aim was to investigate whether the effect was generalizable from native-language competing speech to foreign-language competing speech. Twenty-six experienced hearing aid users with symmetrical moderate to moderately-severe sensorineural hearing loss were tested. The free recall memory task used in Ng et al. (2013) was modified in the present study so that the test became less cognitively demanding. In this task, target sentence lists were presented in a competing speech background, either in the listeners’ native language or in an unfamiliar foreign language. All speech stimuli were presented at favorable individualized signal-to-noise ratios. Working memory capacity was measured using a reading span test. As in Ng et al. (2013), binary masking noise reduction algorithm (Boldt et al., 2008; Wang et al., 2009) was used. Results showed that noise reduction improved memory for words heard in competing speech. This effect was found irrespective of working memory capacity. When the test is less demanding, it appears that the impact of individual differences in working memory capacity on task performance becomes less important. Recall performance was more disrupted by the native-language than the foreign-language competing speech. When noise reduction was applied, the effect of the native-language competing speech was reduced to that of the foreign-language competing speech. One possible mechanism is that noise reduction facilitated segregation of the target native speech from native speech babble. [A companion study will be presented by Lunner et al., having in essence identical methods, but in Danish.]

**Noise reduction improves memory for target speech heard in competing speech**

Elaine Ng, Mary Rudner, Thomas Lunner and Jerker Rönberg

1 Linnaeus Centre HEAD, Swedish Institute for Disability Research, Department of Behavioural Sciences and Learning, Linköping University, Sweden

2 Eriksholm Research Centre, Oticon A/S, Snekersten, Denmark

Ng et al. (2013) showed that noise reduction in hearing aids reduced the adverse effect of irrelevant competing speech on memory for heard speech for hearing aid users. However, the effect was found only for individuals with high working memory capacity. The main aim of the study was to investigate whether the effect of noise reduction on memory for audible speech would generalize across individuals with a range of cognitive abilities if the memory task was simplified by reducing the number of to-be-remembered items. A further aim was to investigate whether the effect was generalizable from native-language competing speech to foreign-language competing speech. Twenty-six experienced hearing aid users with symmetrical moderate to moderately-severe sensorineural hearing loss were tested. The free recall memory task used in Ng et al. (2013) was modified in the present study so that the test became less cognitively demanding. In this task, target sentence lists were presented in a competing speech background, either in the listeners’ native language or in an unfamiliar foreign language. All speech stimuli were presented at favorable individualized signal-to-noise ratios. Working memory capacity was measured using a reading span test. As in Ng et al. (2013), binary masking noise reduction algorithm (Boldt et al., 2008; Wang et al., 2009) was used. Results showed that noise reduction improved memory for words heard in competing speech. This effect was found irrespective of working memory capacity. When the test is less demanding, it appears that the impact of individual differences in working memory capacity on task performance becomes less important. Recall performance was more disrupted by the native-language than the foreign-language competing speech. When noise reduction was applied, the effect of the native-language competing speech was reduced to that of the foreign-language competing speech. One possible mechanism is that noise reduction facilitated segregation of the target native speech from native speech babble. [A companion study will be presented by Lunner et al., having in essence identical methods, but in Danish.]

**References:**


**The validation of the Chinese version of the Spatial Hearing Questionnaire**

Hua Ou, Bei Wen and Richard S. Tyler

1 University of Iowa

2 Sichuan Provincial People’s Hospital, China

The validation of the Chinese version of the Spatial Hearing Questionnaire
**Objectives:** Very few questionnaires address how to measure spatial hearing ability in complex listening situations. The purpose of the study was to validate the Chinese version of the Spatial Hearing Questionnaire (C-SHQ) among Chinese hearing-impaired participants.

**Methods:** This was a cross-sectional study. The C-SHQ was obtained through the process of translation and back-translation of the original English version following the World Health Organization guidelines. The C-SHQ was administered to measure self-perceived spatial hearing ability for 91 hearing-impaired patients seen at the Department of Otolaryngology Clinic of Sichuan Provincial People’s Hospital between October, 2013 and May, 2014 at Sichuan, China. The severity of hearing loss ranged from mild to severe in either one or two ears. The types of losses included sensorineural, conductive, and/or mixed loss. The hearing symmetry was defined as an interaural differences in hearing level of less than 15 dB across the frequencies 0.5, 1, 2, and 4 kHz. 63 out of 91 patients were found to have symmetrical hearing loss for the study. A factor analysis and reliability tests were performed and compared to the results from the original SHQ (Tyler et al., 2009).

**Results:** The average performance score was 80.6% (SD = 20.4%) for all participants. Participants with symmetrical hearing loss indicated higher self-perceived ability compared to those with asymmetrical hearing loss, which suggested good construct validity. The internal consistency reliability was high (Cronbach’s α = 0.987). The range of the item-total correlations was from 0.75 to 0.92. The factor analysis revealed scores loaded on three similar factors compared to the original SHQ.

**Conclusions:** Results indicated that the C-SHQ shares similar psychometric characteristics with the original English version of the SHQ. The C-SHQ is a reliable and valid questionnaire which is suitable for both research and clinical settings to measure spatial hearing abilities in Chinese population.

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**Who is using hearing aids? Analysis of epidemiological and customer pure-tone threshold data from hearing aid users**

*Theresa Nuesse, Petra von Gablenz, Inga Holube*  
*Institute of Hearing Technology und Audiology, Jade University of Applied Sciences, Oldenburg, Germany*

During the epidemiological study HÖRSTAT, 1,900 adults from the northwest of Germany were tested for hearing impairment between 2010-2012. For this purpose pure-tone audiograms, speech recognition in noise and subjective ratings were recorded. Using this data, Holube and von Gablenz (2013) found that 4.6% of the German population wears hearing devices. As expected the percentage of hearing aid uptake increases with increasing hearing loss. For example within the participants with mild hearing loss 10% use hearing aids, whereas this rate increases up to 85 % for those with profound hearing loss. Nevertheless the daily wearing time of hearing aids is less strongly associated with the degree of hearing loss than with the satisfaction with hearing aids. The distribution of hearing losses in the group of hearing aid users in HÖRSTAT, classified according to the criteria of WHO (World Health Organization), are very similar to customer-data acquired by a German hearing aid dispenser. For this analysis data from 21 shops was investigated over a period of approximately 15 years. Both HÖRSTAT and customer data show that almost half of the hearing aid users have a moderate hearing loss. Considering the hearing thresholds of customers at the date of their current hearing aid uptake, data showed less severe hearing loss the later the uptake took place. Furthermore a comparison of the customer database with the age-typical shape of hearing thresholds, which can be expected in healthy, otologically normal adults according to ISO 7029 was made. It seems that hearing aid uptake takes place when the hearing loss exceeds that of the peer group. Furthermore the data of customers who visited the hearing care professional for a first hearing test was investigated. About 85% of them had at least a mild hearing loss according to the WHO criteria. Additionally, a reduction of their hearing losses within the same age can be observed over time. A similar trend in age-related prevalence of hearing loss could be supposed by HÖRSTAT-
Finally the gender differences of customers were examined. It was found that females tend to have lower thresholds at higher frequencies than males within the same age group. However at lower frequencies, approximately below 2 kHz, the effect is the other way round and males perform better.

Reference:

### A37

**Hearing aid outcomes and specific features of the hearing aid**

**Peter Nordqvist**  
**Research Institute Hearing Bridge, Sweden**

The Swedish Quality Register within Hearing Rehabilitation collects information both from private and the county organized dispensers. The register includes subjective and objective information collected from the patient six months after the rehabilitation.

The subjective information consists of a self-reporting questionnaire. The measures in the questionnaire comprise the international outcome inventory for hearing aids (IOI-HA; Cox & Alexander, 2002), a set of questions regarding satisfaction with specific features of the hearing aid, and clinical service (information, participation, and treatment). The information is collected in an identical way for all patients in order to minimize any bias effects.

The objective information is transferred directly from the journal system and consists of audiogram, manufacture, and hearing aid model.

Each year the register collects data from about 45,000 prescriptions. The data is structured on a national level, a county level, a dispenser level and an individual level.

A hearing rehabilitation contains several elements, from the first contact with the audiologist to the final fitting and the acclimatization period. Each element contributes with a certain amount to the overall benefit that is reported from the patient six months after the rehabilitation. However, very little is known about the relative importance between the elements. Are some elements more important than others?

The IOI-HA questionnaire is effective and short and consequently does not include detailed questions regarding specific features of the hearing aid. How important are specific features to the overall benefit? This is essential information when designing and developing signal processing algorithms for hearing aids.

From this work in progress preliminary results will be presented with focus on correlation between IOI-HA score and specific questions regarding features in the hearing aid, e.g. feedback suppression and the handling of uncomfortably loud sound.

The magnitude of benefit differences between units will also be discussed and presented.

### A38

**Predicting self-reported hearing aid outcomes from the Perceptually Robust English Sentence Test - Open Set (PRESTO)**

**Christi W. Miller, Erin Stewart and Samantha Coppa**  
**University of Washington**

The ability to understand speech in noise is one of the most common complaints from hearing aid (HA) users. It would be advantageous to patients and practitioners to predict how a listener performs in his or her own noisy listening environments to support the selection and fitting of HA parameters. Often there is a weak correlation between self-reported HA outcomes and clinical measures of speech perception in noise. One possible reason for the discrepancy could be the limited representation of real world environments in currently used measures of speech understanding in noise. For example, listeners encounter many types of voices throughout the day, yet clinical speech tests only use one voice for all sentences (e.g., QuickSIN; Killion et al., 2004). We hypothesize that a speech-in-noise test using multiple voices would better predict how a person performs in their own listening environments. A recently developed test, called the Perceptually Robust English Sentence Test - Open Set (PRESTO; Gilbert et al., 2013), uses sentences of
various length, syntactic structure, talker gender and dialect. The aim of this study was to characterize how well the PRESTO predicted self-reported HA outcomes. The PRESTO predictions were also compared to predictions from the QuickSIN. Seven adult HA users have been recruited so far. The fittings of the right and left HAs were quantified using real-ear-measures with a 65 dB SPL speech signal. The match to a NAL-NL2 target and the SII were calculated. The PRESTO and QuickSIN sentences were presented at 65 dB SPL in the free field. The PRESTO uses 6-talker babble as the background noise, which was presented at SNRs of 0, +3, +6, and +9 dB. The results at +6 dB SNR were analyzed for this presentation, as they showed the most variability in PRESTO performance from the data collected to date. The QuickSIN presents sentences with a background of 4-talker babble, presented at SNRs of +25 to 0 dB in 5 dB steps. Three measures of aided self-report outcomes were completed. Results from all questionnaires showed statistically stronger correlations to the PRESTO, than to the QuickSIN. Clinical implications are discussed.

A compendium of hearing questionnaires

Jack Holman, Kay Foreman and Michael A Akeroyd
MRC/CSO Institute of Hearing Research - Scottish Section, Glasgow, United Kingdom

As part of wider work on outcome measures for hearing interventions, we have collated available, published hearing questionnaires. We excluded those wholly devoted to tinnitus, children, or cochlear implants. The search was part incremental, part systematic: we believe that we have included every major questionnaire, most of the minor ones, and a fair fraction of those obscure. In total we found 178 questionnaires, of which 109 can be classified as primary (e.g., SSQ) while 69 are variants (e.g. SSQ-B, SSQ-C, and SSQ-12). The primary questionnaires had, in total, 3614 items. The median number of items per questionnaire was 20; the maximum was 158.

This poster reports various summaries of the items (e.g., classified by whether they related to hearing per se, the effects of hearing aids, or the repercussions or a hearing loss). Many questionnaires had common themes; for instance, including some form of “How much difficulty do you have with your hearing” or “how annoyed does your partner get at your hearing loss”. It can be argued that these themes essentially represent what hearing scientists and clinicians have thought important enough about hearing to be worth asking. But it is also clear that the choice of items is an indirect result of the genealogy of the questionnaires, in that many questionnaires take some of their items from an earlier questionnaire, which took them from an earlier one, and so on (e.g., parts of the SSQ were based on the AIAD, the PHAP, and Noble et al’s (1995) unnamed questionnaire). We conclude with recommendations for future questionnaire designs. [Work supported by the Medical Research Council and the Chief Scientist Office, Scotland.]
Posters for Session B should be put up by 8:00 AM Friday, August 15, and taken down after 10:00 PM Friday, August 15 or before 7:00 AM Saturday, August 16. Presenters should be at their posters from 9:45 AM – 11:15 AM; 8:30 PM - 10:00 PM.

POSTER SESSION B

Friday 9:45 AM – 11:15 AM

B1
Development of Mandarin spoken language after pediatric hearing aid fitting
Sigfrid D. Soli1, Gang Li2, Yun Zeng2
1 House Clinic, Los Angeles, USA
2 West China Hospital of Sichuan University, Chengdu, PRC

The purpose of this study was to evaluate early spoken language development in young Mandarin-speaking children, as measured by receptive and expressive vocabulary growth rates, during the first 48 months after they were fit with hearing aids. Growth rates are compared with those of normally hearing children.

Receptive and expressive vocabularies were measured with the Simplified Short Form (SSF) version of the Mandarin Communicative Development Inventory (MCDI) in a sample of 92 pediatric hearing aid recipients at baseline, 3, 6, 12, 24, 36 and 48 months after fitting. Fitting ages ranged from 1-5 years. Scores were expressed in terms of normal equivalent ages, allowing normalized vocabulary growth rates to be determined.

Vocabulary growth during the first 24 months after fitting reached levels similar to that of normally hearing children at 16 months of age. Comparisons with vocabulary growth with that of normally hearing children 16-30 months of age showed that the children fit with hearing aids at 1-2 years of age had the most rapid growth rates while children fit at 2-3 years of age had slightly less rapid growth rates. Children fit with hearing aids after 3 years of age had the smallest average growth rates. However, the initial normal equivalent ages in the oldest groups were higher at baseline than those for the youngest age groups.

The SSF version of the MCDI is suitable for assessment of Mandarin language development during the first 48 months after hearing aid fitting. Effects of fitting age and duration of hearing aid use can be expressed using normalized vocabulary growth rates, which allow direct comparisons with language development trajectories for children with normal hearing. The clearest evidence of early language development following early intervention with hearing aids occurs, on average, during the first 24 months after hearing aid fitting, regardless of fitting age. These findings can be used to define milestones for early language development after hearing impaired children are fit with hearing aids.

B2
The effects of noise reduction on forward masked thresholds
Jordan Rock1, Christi Miller1, Marc Brennan2 and Justin Zakis3
1 University of Washington
2 Boys Town National Research Hospital
3 Wolfson Dynamic Hearing Pty Ltd

The most common complaint among hearing aid users is difficulty understanding speech in background noise. Research has shown that noise reduction (NR) algorithms can improve listening effort and comfort, but improvements in speech perception have not been demonstrated consistently. Understanding the mechanisms responsible for these perceptual effects can help clinicians and researchers choose NR settings to achieve maximum benefit. Because listeners use temporal cues to decode speech, understanding the effects of noise reduction on measures of temporal resolution could lead to improvements in speech understanding with noise reduction. One possible explanation is that NR algorithms reduce the forward masking of a target signal by a masker that immediately precedes it. The aim of this study was to determine whether NR algorithms improve forward masked thresholds by reducing the level of the masker. We will present preliminary data (at least five normal- and five impaired-hearing listeners) from this on-going study. For-
ward masked thresholds were measured using a 2 alternative forced choice paradigm. The target stimulus was a 20 ms pure tone at 2000 Hz preceded by narrowband noise (masker). The delay time between the masker and the signal was 10 ms. The masker was the width of one critical bandwidth (200 Hz) and 4 seconds in duration. All stimuli were processed by a MATLAB implementation of commercial hearing-aid algorithms, which consist of a time-domain signal path with its gain-frequency response controlled by frequency-domain NR and amplification algorithms. The strength and time constants of the NR gain were manipulated. The activation of the NR algorithm in response to the forward masking stimuli was confirmed in each condition. Each subject had their stimuli amplified in accordance with NAL-RP prescription gain formulas and presented via headphones. Linear gain allows the isolation of NR effects from those due to other types of processing (e.g., compression). Thresholds with NR activated were compared to baseline thresholds with noise reduction deactivated. It was expected that there would be a greater improvement in forward masked thresholds as the NR strength increases and time constants decrease.

B3

Cochlear amplification and the cytoarchitecture of the organ of Corti

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The thousands of outer hair cells (OHCs) in the cochlea, found within the organ of Corti (OoC) situated along the length of the basilar membrane (BM), are known to detect and change shape in response to sound-induced displacements, although precisely how the OHCs work together to provide the high sensitivity and frequency selectivity of mammalian hearing remains only partially understood. An interlaced arrangement of Y-shaped structures along the length and width of the OoC, each formed by an OHC, phalangeal process (PhP), and Deiters’ cell (DC), is thought to pump mechanical energy onto the BM in order to amplify the travelling wave. Experiments to confirm this hypothesis are difficult to perform, however, and remain lacking, so we instead used two computational models to test the hypothesis that the Y-shaped structures in the OoC form the fundamental building blocks for cochlear amplification. An improved fundamental understanding of cochlear amplification could lead to improved amplification strategies for hearing devices.

For the first approach, a multi-scale 3D finite element (FE) model of the mouse cochlea with realistic OoC cytoarchitecture was constructed using COMSOL Multiphysics software. Linearized equations representing the fluid in the cochlear scalae were coupled with a structural orthotropic material for the BM and solved in the frequency domain, using Euler-Bernoulli beams to represent the PhPs and DCs and to reduce the computational expense while still modeling the OoC in a physically realistic way. For the second approach, we used a more efficient asymptotic model that allowed direct calculations of the power in each cross section of the cochlea. The fluid pressure and velocity distribution in the scala vestibuli were examined, and the power input from the outer hair cells and power loss due to viscous damping in the fluid were distinguished from the total power change along the cochlea.

Both models showed amplification of BM motion consistent with physiological measurements in the mouse cochlea. The influence of OoC material properties on the traveling wave and amplification was explored using the FE model, and comparisons with experimental results allowed us to estimate the material properties of the OoC’s individual structures. The asymptotic model showed that the amount of power provided by the OHCs is close to the viscous power loss, such that the total power is barely positive near the area of peak vibration on the BM. This may explain why it has been so difficult to experimentally measure the cochlear amplifier.
Objective evaluation of adaptive noise reduction algorithms in the presence of noise and reverberation

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Noise and reverberation are pervasive within listening environments, necessitating additional signal processing for effective amplification. Adaptive Noise Reduction (ANR) techniques play an important role in noise management. While there is some evidence for the efficacy of ANR algorithms, no clear clinical guidelines exist for objective ANR evaluation. The objectives of this study are two-fold: (a) to benchmark the performance of ANR algorithms in noisy and reverberant environments using speech intelligibility, quality, and loudness metrics, and (b) to validate a procedure for simulating reverberation within a test chamber for clinical verification of ANR performance in reverberation.

Premium BTE hearing aids (HA) from different manufacturers were fitted to match the same prescriptive targets for the “N4” standard audiograma, and placed inside a desktop anechoic test chamber. The ISTS signal was played back either alone (i.e. quiet condition), or mixed with either the speech-shaped stationary noise or multi-talker babble at 0 dB or 5 dB SNRs. Aided recordings were obtained in a 2cc coupler, with the ANR algorithm on or off. The Speech Intelligibility Index (SII), Hearing Aid Speech Quality Index (HASQI), and Loudness based on the Moore-Glasberg auditory model, were measured for each recording. In the second set of experiments, the same BTEs were attached to a 2-cc coupler and placed at the centre of a speaker array in a reverberation chamber, with configurable reverberation times (RT60) of 0.76 sec or 0.97 sec. In each RT60 configuration, the ISTS was played back from the 0° azimuth speaker and aided recordings were made within the coupler. The SII, HASQI, and loudness metrics were calculated. To simulate reverberation in the test box, the reverberation chamber impulse response was measured at the location of the BTE microphones in both RT60 configurations, and used to filter the speech stimuli. The filtered speech stimuli were played back in the test box and a separate set of HA recordings and instrumental metrics were collected. Preliminary data analyses show that: (a) ANR algorithms do not affect speech intelligibility in quiet; (b) there are performance differences across ANRs in noise; (c) reverberation affects the ANR algorithm performance; and (d) metrics computed in real and simulated reverberation environments are similar. Clinical implications of these results will be discussed.

Reference:


Design of a competing voices test

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Competing voices are part of the everyday challenges for a hearing aid user. In order to test the performance of hearing aids in this user scenario, a new type of speech test has to be developed.

Compared to traditional tests, the competing voices test has two or more targets that are all important to follow and in the simplified case no masker. By definition, since the energy levels are equal in the two targets, the signal-noise-ratio (SNR) is then zero. A good competing voices test should not only be ecological and resemble a typical daily situation; it should also be accurate and repeatable without learning effects. Tests of this type have been reported in the literature (Mackersie, Prida, & Stiles, 2001), (Helfer, Chevalier, & Freyman, 2010) but no particular test has been put to common use. Suitable tests using continuous speech are also being investigated (Xia, Hafter, Kalluri, & Edwards, 2012), (MacPherson & Akeroyd, 2014).

In the poster, we present a range of potential solutions, using existing or new speech material, and using various scoring techniques. Some of the proposed tests have been qualitatively evaluated with hearing-impaired listeners for their eco-
logical validity (e.g. is this encountered in real life) and for their difficulty (listening effort) and precision. The suggested models are discussed and directions for future work are proposed.

References:

**B6**

Binaural advantage and spatial release from masking with different masking noises

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'HiST speech audiometry' is a speech test in Norwegian that can be used in sound field audiometry with a standard surround sound DVD equipment. The speech test contains lists of 20 three-word utterances and each of the subsequent utterance is reduced in level until it becomes unintelligible. Thresholds can be calculated from the number of words recognized in each list that is similar to the QuickSIN test. The five surround speakers are arranged in the following five azimuth positions: 0°, ±45° and ±135°. It has been observed that tests administered with speech from the center speaker (0° azimuth) and uncorrelated stationary masking noise from the remaining four speakers have hardly resulted in a spatial release from masking. Measurements comparing speech and noise localized in front speaker have shown only 1.6 dB release from masking. Such a small difference limits the efficacy of the method for evaluating listeners' performance with or without hearing aids and there is a need to identify another type of masking noise.

The current study will be performed by audiology students in our department, where in, they alter between the role as a tester and subject. The following 5 sounds will be evaluated as masking noise: Stationary speech noise, ICRA noise (International Collegium of Rehabilitative Audiology, track 5 - male 3 band speech modulated), reverse speech from the same presenter as used in the speech test material, ISTS (International Speech Test Signal) and 4 concurrent speakers (2 male and 2 female).

For all masking noises, measurements with test speech and 4 uncorrelated masking noises located together in center channel is compared with measurements performed with masking noise presented in the other 4 channels. Spatial release from masking will be evaluated for all of these measurements.

To evaluate the binaural advantage for these situations, measurements will be performed twice-one series measured with binaural listening and another with monaural listening. To achieve monaural listening condition, stationary speech noise is delivered to one ear through an E-A-R Tone 3A insert phone with foam tip, with sufficient levels to mask the ear. Binaural advantage is calculated as the difference between the monaural and binaural measurements.

The results for signal-to-noise ratios, spatial release from masking and binaural advantage will be presented. The results of the present study have important implications on the type of masking noise to be selected for the successive version of HiST speech audiometry.

**B7**

Hearing aid users’ self-adjustment of amplification in noisy listening conditions

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Hearing aid fitting formulas are designed to maximize audibility and comfort in favorable listening conditions. However, many daily situations of active hearing aid users are noisy. It is well known that people with hearing loss have particular difficulty in understanding speech in noisy listening conditions, but it is still not clear why this occurs or how hearing aid signal processing should be configured to improve performance in noise. In this study ten subjects with varying degrees of sensorineural hearing loss listened to speech in recorded real-world noise. The subjects listened via 9-channel WDRC hearing aid-like processing implemented on an iPod Touch with headphones. Gain, compression, and MPO in all 9 channels, as well as overall frequency response, were adjusted in real time by the subjects using two simple controllers (Ear Machine) on the iPod. Subjects listened in a variety of SNRs ranging from -5 to +25 dB and adjusted the controllers in each SNR to the hearing aid parameters that they felt allowed them to understand speech best. The stimulus was the female talker from the Connected Speech Test that was played in the presence of the background noise recordings made in a living room and three noisy restaurants. The listeners also made the same self-adjustments in the field at the local restaurants from which the recordings were made. Results show that the listeners set their preferred gain and frequency response settings similar to NAL-NL2 fitting targets in quiet living room settings. In noise, however, self-adjusted parameters deviated systematically from NAL-NL2 targets. On average, subjects reduced gain incrementally as SNR decreased from 10 to 0 dB, and reduced it substantially for the lowest SNR. Substantial inter-subject variability was observed. [Work supported by NIDCD R01 DC 13267.]

**Beyond speech intelligibility testing: A memory test for assessment of signal processing interventions in ecologically valid listening situations**

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Working memory (WM) is important for online language processing in a dialogue. We use it to store, to inhibit or ignore what is not relevant, and to attend to things selectively. According to the Ease of Language Understanding (ELU) model, a poor representation of a signal in the neural pathways will lead to activations of the (effortful) WM system. We argue that hearing impaired persons must rely much more on effortful WM resources to understand what has been said, compared to normal hearing persons who can rely more on effortless highly over-learned automatic speech recognition systems. Therefore tests which probe the degree of load on WM resources may be useful for evaluating the benefit of hearing aids.

Performance of hearing aid signal processing is often assessed by speech intelligibility in noise tests. Usually such tests are most sensitive at a signal-to-noise ratio (SNR) below 0 dB. However, a recent study by Smeds et al. (2012) showed that the SNRs in ecological listening situations (e.g. kitchen, babble, and car) were typically well above 0 dB SNR. That is, SNRs where the speech intelligibility in noise tests are insensitive. Therefore new measures are needed that can show eventual benefits of hearing aid processing in the +5-15 dB range.

Cognitive Spare Capacity (CSC) refers to the residual capacity after successful speech perception. In a recent study by Ng et al. (2013), they defined the residual capacity to be number of words recalled after successful listening to a number of HINT sentences. In this presentation we describe a recent study with 25 hearing impaired test subjects. Close to 100% correct speech intelligibility in a four talker babble noise required around +7-9 dB SNR. At that SNR, a hearing aid noise reduction scheme improved memory recall by about 10%. Thus, this kind of memory recall test is a possible candidate for assessment of hearing aid functionality in ecologically relevant (positive) SNRs.

A companion study will be presented in a separate poster by Ng et al., having in essence identical methods, but in Swedish.
B9

Preliminary results of a survey of experiences in listening to music via hearing aids
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An internet-based survey was conducted of experiences in listening to music via hearing aids. Preliminary analyses of the results are presented here. Valid responses were obtained from over 500 participants. Responses were analyzed according to the self-reported degree of hearing loss: Mild, Moderate, Severe, and Profound. 36% of participants reported problems with acoustic feedback when listening to music, and the prevalence of such problems did not vary significantly with degree of hearing loss. A substantial number of participants reported problems such as “warbling effects” or “when (the organ) music STOPS, aids howl”, that are probably associated with feedback cancellation. In response to the question “When listening to music via radio, TV or stereo system, do you find your hearing aids to be helpful”, 64% of participants responded “Yes, a lot” or “yes, a little”, but 13% responded “No, a bit worse” or “No, a lot worse”. In response to the question “When listening to live music, do you find your hearing aids to be helpful”, 53% of participants responded “Yes, a lot” or “Yes, a little”, but 17% responded “No, a bit worse” or “No, a lot worse”. The difference in responses for reproduced and live music probably reflects the higher sound levels that are typical of the latter. For live music, hearing aids tended to be more helpful for those with severe or profound hearing loss. While 61% of participants agreed with the statement that hearing aids “Make the music louder”, only 28% agreed with the statement that hearing aids “Help hear soft passages without the louder parts being too loud”. This indicates that the automatic gain control systems in the hearing aids were not performing as well as would be desired. 32% of participants reported that hearing aids “sometimes make the music seem distorted”, perhaps reflecting clipping or overload of the circuitry resulting from the high levels of live music. Only 21% of participants agreed with the statement that hearing aids “improve the tone quality of music” while 29% agreed with the statement that hearing aids “worsen the tone quality of music”. Judgements of tone quality tended to be more positive for participants with mild hearing loss.

Overall, the results point to a need to improve feedback cancellation systems and automatic gain control systems in hearing aids, and to increase the dynamic range of hearing aids to avoid distortion for high input levels. [Work was supported by Starkey (USA).]

B10

Diagnostics, first fit, and fine tuning in a Self-Fit hearing aid
Monique Boymans, Wouter A. Dreschler
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The Self-Fit hearing aid is a personal amplification device controllable through an iOS based mobile app, manageable entirely by the user without assistance from a (hearing) health care professional or the need for special equipment. This hearing aid contains an onboard tone generator for in-situ user-controlled, automated audiometry (2-Tone test), and other relevant tests (ACALOS). It contains also an onboard prescriptive algorithm to determine the initial hearing aid settings (First-Fit) and it contains a well-structured interactive audio-visual method to optimize and individualize the initial fitting by Paired Comparisons streamed through the hearing device (Final-Fit).

A self-fit hearing aid can be used in the “developing” world or in countries with large distances between the hearing impaired and the hearing health care professional. However, the fine-tuning paradigm implemented may also be applicable in the “developed” world to optimise the first fit by a well-structured set of paired comparisons for user-driven fine tuning.

20 Subjects were involved in this study: 11 new hearing aid users and 9 experienced hearing aid users with mild hearing losses. They fitted one hearing aid by themselves using the Self-fit App for i-Pad. The frequencies that were used only were: 750, 1500 and 3000 Hz.

The following results will be shown:
Relationship between temporal processing ability and performance on a time-compressed speech test in younger and older listeners

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As most measures of temporal processing ability decline with age, and some are also correlated with speech understanding in degraded listening conditions, it seems likely that temporal processing ability may help account for differences in speech understanding between younger and older listeners in cases where performance cannot be entirely explained by the audiogram. Available data suggest that temporal processing is particularly important for understanding speech-in-noise and rapid speech. We sought to determine whether and to what extent aging, hearing sensitivity, and temporal processing ability, impact the identification of time-compressed speech in quiet and in noise. Listeners varying in age across a 50-year range with nominally normal hearing were presented with time-compressed digit strings at three uniform compression ratios (2:1, 3:1, and 5:1) in quiet and in a steady-state speech-shaped background noise (SNR +5). Listeners identified the final four digits of the seven-digit strings. In additional conditions, the total duration of the compressed stimuli was expanded by the insertion of periodic gaps. To assess temporal processing ability in these same listeners, measures of monaural gap discrimination were obtained with four stimulus types. Listeners were asked to discriminate differences in gap duration between pairs of (1) brief (4-ms) 2-kHz tone bursts; (2) rising-frequency chirps (Dau et al., 1999); (3) time-reversed chirps; and (4) “noise chirps” which had the same frequency spectrum as the rising chirp, but with a randomized phase spectrum. As expected, increased time compression together with the presence of background noise impaired performance for all listeners on the speech task. The effects of time compression were somewhat mitigated through the introduction of periodic gaps. Although significant relationships with were observed between speech identification and both age and hearing loss, especially at the highest compression rates, performance on the gap discrimination task accounted for more of the variability in performance among listeners in all listening conditions. Results show that aging itself, separate from age-associated threshold elevation, impacts speech understanding in challenging listening conditions. Results further indicate that listeners’ temporal processing ability accounts for more of the variability in speech identification performance than either age or hearing loss. [Work supported by VA RR&D.]

Cognitive and hierarchical Bayesian models for subjective preference data

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Subjective preference data are an important source of information for the subjective evaluation of hearing aids. Unfortunately, due to the ordinal nature of preference judgments, standard statistical tools such as general linear models (e.g., standard analysis of variance) are inad-
Incorporating spatial information into optimal binaural noise suppression design for hearing aids
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The basic idea of noise suppression is to obtain an estimate of the reference signal. Several existing binaural noise suppression algorithms have been proposed to extend the basic design to situations involving binaural hearing aids by exploiting the extra degrees of freedom brought by the multiple microphones. Some of the existing algorithms have been formulated to strike a balance between noise suppression and speech distortion. However, the existing algorithms don’t fully take advantage of the spatial locations of interferences even though it is possible to estimate them in practice.

In this work, we revisit the binaural noise suppression problem by incorporating the a priori spatial information into consideration. We formulate the algorithm design aiming at striking an appropriate balance between two main design objectives: noise suppression and speech distortion. In addition, we propose a low-complexity iterative approach to solve the proposed formulation. The proposed method has a similar computational complexity to most conventional binaural noise suppression algorithms. Numerical simulation shows that the proposed algorithm can achieve better performance than the existing algorithms in terms of both design objectives under reverberant conditions using imperfect spatial information. The improvement becomes much more significant when voice activity detection errors increase. Moreover, the proposed algorithm converges quickly, often achieving a close-to-optimal performance within 5~10 iterations.
Acoustic simulation using 3D modeling of the Development of a super-directional system, past, present and future

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It is well established that modern directional microphones in hearing aids provide listeners with an improved speech understanding in noise. Despite this, the impact of directional microphones in real life conditions is limited. In fact, hearing in noise remains one of the biggest problems for hearing aid users. Fortunately, recent developments in super-directional technology, at least in laboratory settings, promise to deliver significant benefits to hearing aid users. Experiments suggest large improvements in speech understanding in noise and significant preference for highly directional systems. This advantage is often extrapolated to suggest equal advantages in real world listening situations. However, it is increasingly apparent that hearing in noise entails various complex tasks for the listener. Consequently, super-directional technology may be advantageous in some situations but may also have some limitations in its usage. Here we present a discussion of super-directional microphone technology based on several studies. In our research we have examined various factors that influence benefit such as beamwidth design, adaptation speed, preservation of spatial cues, vent sizes, acoustic scene, and reverberation. Our evidence appears to be confounded by individual characteristics of the listener such as age, hearing loss, personality traits, and cognition. All these factors combined will guide our discussions and thoughts about future research and development of super-directional systems.

Audibility and spectral resolution as predictors of speech recognition with nonlinear frequency compression

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Nonlinear frequency compression (NFC) has been implemented in hearing aids to extend the upper bandwidth that is audible to listeners with hearing loss. However, improvements in speech recognition with NFC are highly variable across individuals and across studies. Age, degree of hearing loss, and stimulus factors have all been identified as contributing to the variability in speech recognition with NFC. Recent evidence suggests that speech recognition with NFC may depend on how much the processing affects audibility. For listeners with mild to moderate high-frequency hearing loss, speech recognition was consistently improved when the NFC settings were optimized to provide the broadest audible bandwidth (McCreery, et al. 2014). However, listeners with greater degrees of hearing loss may have greater negative impacts from spectral distortion, particularly for signals that are frequency compressed. The purpose of this study was to evaluate speech audibility and spectral resolution as predictors of word recognition under conditions with conventional hearing-aid processing and NFC in adults with moderate to severe hearing loss. Monosyllabic word recognition and measures of listener preference were evaluated for 24 adults using hearing aids with and without NFC. Spectral-ripple discrimination was completed with and without NFC as a measure of spectral resolution. Overall, the mean differences between conventional processing and NFC were not significant; however, significant individual variability was observed. To explore the factors that might determine the amount of benefit listeners experience from NFC, a linear regression model with aided SII and spectral-ripple discrimination threshold (ripples/octave) as predictors of speech recognition accounted for more than half of the variability in aided word recognition in both conventional processing and NFC conditions. These results support the hypothesis that both audibility and spectral resolu-
tion may influence aided speech recognition in adults with moderate to severe sensorineural hearing loss. Clinical assessment of spectral resolution may help to predict individual variability in benefit from frequency lowering.

Reference:

Domains of sound-tolerance underlying acceptable noise level
Carol Mackersie, Stephanie Baxter, Megan Lane
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Purpose: The purpose was to determine the contributions of four sound tolerance domains (loudness, annoyance, distraction, communication interference) to the acceptable noise level measure (ANL).

Methods: Acceptable noise levels of thirty young adults (18-31 years) were measured using the original ANL babble recordings. Speech levels were adjusted to listeners’ comfortable loudness levels. Background noise was added to the speech signal and adjusted to determine the highest acceptable level “while listening to the speech”.

Paired-comparison tests were administered to determine which domain most strongly influenced the participants’ judgments of noise acceptability. Each domain was paired with all other domains. Participants were also asked to complete absolute ratings of loudness, annoyance, distraction, and interference using a 100-point scale. Paired comparisons and absolute ratings were obtained with the noise level adjusted to 3 dB above the maximum acceptable background noise level.

These sound-tolerance domains were defined for the participants as follows:

Loudness: “I don’t mind the way it sounds and it doesn’t interfere with what I want to hear. It is just too loud. It makes me want to get away from it.”

Annoyance: “The noise bothers me. I don’t like the sound, regardless of how loud it is or how much it interferes with my listening.”

Distraction: “I don’t mind the sound and the loudness doesn’t bother me. But it takes my attention away from what I am doing. It is hard to concentrate.”

Communication Interference: “The noise makes it hard for me to follow what the person is saying. It interferes with what I am trying to hear.”

Results: Similar patterns were observed for the paired-comparisons and absolute ratings of the sound-tolerance domains. Two distinct sound-tolerance domain patterns were revealed by a k-means cluster analysis. One-third of the participants weighted annoyance as a stronger determinant of noise acceptability than interference. The remaining two-thirds weighted communication interference as a stronger determinant than annoyance. The ANL of the “annoyance-weighted” group was significantly higher (5.80) than the ANL for the “interference-weighted” group (0.80).

There was a significant correlation between ANL and absolute ratings of annoyance, but not the other domains; higher ANLs were associated with higher ratings annoyance.

Conclusion: Distinct sound tolerance domain profiles suggest that young normal-hearing listeners differ in the criteria used for ANL judgments. Those with higher ratings of annoyance were less tolerant of noise (higher ANLs).

Comparison of auditory brainstem responses to round and oval window stimulation in animal experiments
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² Keimyung University
³ University of Iowa

To ensure the safety and efficacy of implantable hearing aids, animal experiments are an essential
developmental procedure, in particular, auditory brainstem responses (ABRs) can be used to verify the objective effectiveness of implantable hearing aids.

In the case of implant middle ear hearing aids (IMEHDs), which have been developed and commercialized since the 1980s, the input acoustic energy is applied to an oval window. However, recent research has been focusing on the use of a round window. The performance and effectiveness of oval window stimulus-type IMEHDs have already been verified based on the stapes footplate response according to ASTM F2504-05. Plus, the same method has been used to confirm the performance and effectiveness of round window stimulus-type IMEHDs, as it was assumed that the biological response to round window stimulation would be the same as that to oval window stimulation. New study results were also recently published on this topic in the Journal of Hearing Research.

Accordingly, to confirm the abovementioned results, this study measured and compared the ABRs generated when applying the same vibration stimuli to an oval window and round window. The ABRs were measured using a TDT system 3 (TDT, USA), while the vibration stimuli were applied to a round window and oval window in 4 guinea pigs using a piezo-electric transducer with a proper contact tip. A paired t-test was used to determine any differences between the ABR amplitudes when applying the stimulation to an oval window and round window. The paired t-test revealed a significant difference between the ABR amplitudes generated by the round and oval window stimulation (t=10.08, α<.0001). Therefore, the results confirmed that the biological response to round window stimulation was not the same as that to oval window stimulation. [Work supported by the Korean Government (NRF-2012R1A1A2006475) and (Korean Ministry of Health & Welfare-A092106).]

**B18**

Long-term effects of non-linear frequency compression on performance of music perception

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Several studies have indicated that speech perception is improved when using a Non-Linear Frequency Compression (NLFC) algorithm (see a summary in Hopkins et al. 2014).

General research on frequency lowering hearing instruments and music perception is however limited and this is troublesome as listening to music is an integral part of people’s daily lives and provides a medium for human interaction (Cross 2006).

Some research in this area has used the Music Perception Test (MPT) (Uys 2011) which focuses on how subjects perceive basic music components like rhythm, timbre, pitch and melody. The studies showed (Uys 2011, Uys et al. 2012, Uys et al. 2013) that the perception could be improved when subjects got access to high-frequency components by using NLFC.

The current study looks at the long-term effect of a NLFC scheme on music perception, something that could not be evaluated in the original Uys studies. This was done by inviting a subgroup of the subjects described by Uys et al. for a retest of the MPT after three years of using an NLFC system. Additionally, speech perception in quiet and noise was measured using the HINT to investigate the long-term effect of NLFC on speech perception. The initial results indicate a trend towards improved performance in music as well as speech perception with NLFC activated.

**References:**


Development of a music based mobile auditory training game
Francis Kuk and Petri Korhonen
Widex ORCA

Auditory training has been demonstrated to improve neural encoding and to produce measurable benefits when listening to speech sounds presented in noise. The convenience and relatively low cost of computer assisted training programs have made it easier for a clinician to employ auditory training as a component of comprehensive auditory rehabilitation. However, depending on the design of the training program, the trainee may find the training tedious and unexciting. More entertaining training programs have been shown to reduce the dropout rates and to improve effectiveness auditory training.

We have developed a computer-assisted auditory training program, which uses a game format to challenge the trainee’s auditory, attention, memory, and problem solving skills. The program was implemented as an application that runs on a mobile tablet or a smartphone (iOS). Unlike the typical auditory training programs that use speech as the source material, the current program uses melodies played by various musical instruments as stimuli. One of the potential merits of using music as stimuli is that musical activities are often enjoyable. Also, the use of music renders the training program language independent. The game-like aspect of the training was selected to engage the listeners’ problem solving skills, increase the attention towards the training, and to promote repetition by challenging the trainee to improve their own performance.

Research has demonstrated that music training can induce functional and structural changes in the auditory system, which primes the brain for processing musical sounds. Interestingly, plethora of evidence shows that musical training can also improve auditory skills in non-musical domains, such as speech, language, and emotion. Importantly, musical training has been shown to enhance the ability to hear speech in noise (Parbery-Clark et al. 2009). The training materials in the current program were selected in an attempt to specifically target improving auditory processing of pitch, timbre, and timing. These three aspects of sound have been shown to share processing mechanisms in the subcortical structures of the brain by both music and speech. The program adjusts the acoustic characteristics of the music stimuli (pitch, timing, timbre) iteratively for individual auditory training need based on the trainee’s skill level.

The rationale used in the development of this mobile auditory training game application will be presented. Additionally, preliminary results of a pilot study investigating the efficacy of the program in improving auditory skills will be discussed.

B20

Technical analysis of an OAE hearing measurement probe
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A hearing measurement probe is an acoustic system for measuring otoacoustic emissions (OAEs). In this presentation we shall outline the many pros and cons of such probes. The typical probe has built-in sound sources (receivers), and a microphone. There is only one such commercially available system, the Etymotic ER10C. This venerable probe system has been widely used clinically for hearing screening, measuring OAEs, middle ear impedance/reflectance, and diagnostics for the several decades. We have experimentally investigated the properties of this OAE probe. There are two main parts in our study, investigation of a) the microphone and b) the receiver. Beginning with a detailed description of the probe, we carried out crosstalk measurement to investigate the microphone characteristics and study electrical input impedance measurements of the receiver with various loads to characterize sensitivity of the system's output. We believe that this study will not only provide a fundamental and operational understanding the ER10C sys-
Factors associated with satisfaction in hearing aid users

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Background and Objectives: This investigation evaluated the level of subjective satisfaction in hearing aid users by adopting Korean version of International Outcome Inventory for Hearing Aids (K-IOI-HA) and Hearing Handicap Inventory for the Elderly (K-HHIE).

Materials and Methods: The K-IOI-HA and K-HHIE are subjective evaluation methods for hearing aid users. One hundred and eighty-seven subjects participated in this study and completed the K-IOI-HA and K-HHIE at pre-fitting, 1 month after fitting and three to six months after fitting. The satisfaction score of each survey was analyzed according to the patient-related and hearing aids-related factors.

Results: Pre-hearing aid disability evaluated by K-HHIE was significantly associated with the pure tone average (PTA) and age (p < 0.05). After using the hearing aids, the K-IOI-HA score showed a significant correlation with the word recognition score and the PTA (p < 0.05). The results of 1 month HHIE revealed that subjective disability significantly improved according to the word recognition score (p < 0.05).

Conclusion: Word recognition score is the most important factor for satisfaction in hearing aid users. [This work was supported by a grant from the Korean Health Technology R&D Project, Ministry of Health & Welfare (HI12C1460) and Seoul R&D Program (SS100022), Republic of Korea.]

A design of the signal processing chip for fully-implantable middle ear hearing device

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Recently, the population of hearing impaired people is increasing continuously and more people suffer from their hearing problems. According to this problem, the various types of hearing aid are being rapidly developed. Especially, the fully-implantable middle ear hearing device (F-IMEHD) is actively studied for sensorineural hearing loss people. The basic F-IMEHD system consists of implantable microphone, signal processor, and vibration transducer. A Korea search team has already been developed with implantable microphone and vibration transducer and also signal processor is being developed. The signal processor design must consider the input and output characteristics such as microphone output level and resonance of vibration transducer. In addition, signal processor has accomplished the small size and low power consumption for implantation.

In this paper, we design and implement the small size and low power consumption signal processing chip for the F-IMEHD system. The designed chip consists of analog to digital converter (ADC), multi-channel wide dynamic range compression (WDRC), and output stage. Traditionally most hearing aids have been designed using the Σ-Δ ADC. However, Σ-Δ ADC is difficult to implement in a small size and low power consumption. In order to achieve the small size and low power consumption, we adopt the successive approximation register (SAR) ADC. The designed SAR ADC has 16 bit data resolution and operates on the 32 kHz sample rate. The multi-channel WDRC is applied 4-channel system to concern the frequency resonance of transducer. The frequency channels are separated to use 64 point FFT method. The proposed algorithm is calculated the cosine factor...
for frequency analysis and signal reconstruction is used to summation. The output stage is used class D amplifier which is widely used as power amplifier for hearing aid. The output stage is composed of a pair of inverter and it is adopted the fixed taper buffer strategy.

The designed signal processing chip is implemented in 0.18 $\mu$m mixed-signal CMOS technology and die size of 2.1 $\times$ 2.1 mm. As a computer simulation results, designed chip consumes 180 $\mu$W with a 1.8 V supply voltage.

[Work supported by a grant of the Korea Healthcare Technology R&D Project, Ministry of Health & Welfare (A092106) and was also supported by the National Research Foundation of Korea (NRF) grant funded by the Korea government (MSIP) (No.2013R1A2A1A09015 677).]

**B23**

**Patient-specific approach to frequency compression**

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Frequency lowering features in hearing aids are aimed at hearing loss compensation for those patients whose high frequency loss is too severe to obtain audibility from standard amplification methods. Currently available hearing aid frequency lowering features employ a large variety of signal processing methods, parameterization approaches and default feature settings. These differences between currently available hearing aid features prevent the use of a universal frequency lowering fitting rationale. Unlike other frequency lowering fitting rationales, which either switch the feature off by default or set the parameters at minimum value, an individually optimized approach will be introduced, with the goal of indicating the estimated benefit of the feature for each individual patient. Default parameter values based on empirical studies, known factors influencing a successful fitting and the specifics of the hearing aid system will be presented.

**B24**

**The role of memory in hearing aid outcome measures**

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Many hearing-aid outcome measures rely on listener’s memory for sound quality. In paired comparisons, the listener must hold quality of one listening experience in memory while comparing to another listening experience, which reduces reliability (Purdy & Pavlovich, 1992). Retrospective questionnaires of hearing aid benefit and satisfaction ask the respondent to recall multiple instances of varied acoustic environments over an extended period of time of several weeks to months. Little is known about the time-course of auditory memory for sound quality. Studies of echoic memory show the auditory image may fade within 2 seconds to several minutes (Deutsch, 1975; Winkler & Cowan, 2005), while memory for qualities, such as those that are associated with voice recognition, may last for up to 24 hours (Yarmey & Matthys, 1992). For this experiment, we will compare listeners’ evaluations during, soon after (10 seconds, 1 minute, 5 minutes), and long after (1 hour) listening conditions are presented in the lab.

Participants (10 participants, at least 6 months HA experience, moderate to moderately-severe bilateral SNHL, age 18 yrs or older) will be asked to complete speech-in-noise testing at a variety of signal-to-noise ratios (0, +5, and +10 dB SNR), at three different presentation levels (50, 65, and 80 dB SPL to represent a range of conversational levels; cf Pearson, Bennett, Fidell, 1977), and with two different types of background noise (competing speech babble and non-speech steady-state noise). During each block of testing (at each SNR, level, and type of background noise), participants will complete a subjective assessment of their listening experience, including ratings of intelligibility of the target speech; difficulty of the listening task, loudness of speech and noise, annoyance of noise and
quality of speech. For each assessment, participants will be given a pair of descriptors and be asked to mark a point on a continuum between the two descriptors.

Pilot data show that test-retest reliability decreases as a function of time after the listening condition is presented. Implications for hearing aid outcome measures will be discussed.

References:

B25

Comparison of single-channel noise reduction schemes: Can hearing impaired listeners tell the difference?

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Nine different single-channel noise reduction algorithms were evaluated in subjective listening tests. Male and female speech samples (short sentences) added with party noise and traffic noise at 0 dB and 10 dB SNR were used as input signals. 18 normal hearing and 18 hearing impaired subjects participated at two test sites (nine of each group per site). Assessments were done employing a simplified MUSHRA measurement method. Four sound quality dimensions of the processed signals and the unprocessed signal were assessed: (1) overall quality, (2) amount of speech distortions, (3) intrusiveness of the background noise and (4) listening effort. The results revealed significant differences between the rating behaviors of the two subject groups: While normal hearing subjects clearly differentiated between the different noise reduction algorithms, hearing impaired subjects rated all processed signals very similarly and basically only differentiated between processed and unprocessed signals and between the two input SNR conditions. This applies to all tested quality dimensions. Moreover, speech distortion rating differences between processed and unprocessed signals had opposite signs in the two subject groups: As expected, normal hearing subjects rated the perceived speech distortion of the unprocessed signals minimal and clearly lower than the distortions of all processed signals, which generally include more or less processing artifacts that distort the speech. In contrast, hearing impaired subjects showed the opposite rating behavior, i.e. they rated the speech distortions of the unprocessed, noisier signals as more severe than the distortions of the processed signals. Since all measurement conditions, including experimenter and instructions, were the same for both subject groups and the described differences between subject groups were consistent across both test sites, possible bias effects do not seem likely to explain the observed differences. Hence, the results suggest that hearing impaired listeners either have a different understanding of the term “speech distortion” than normal hearing listeners have, or that it is harder for them to distinguish between additive noise and speech distortions due to an altered perception.

B26

Hearing loss and speech-processing related fatigue

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Fatigue is common among individuals with a variety of chronic health conditions and has significant negative effects on quality of life. Recent evidence suggests that adults and children with hearing loss (HL) may also be at increased risk for fatigue and energy deficits. It is commonly assumed that the increased mental effort required by persons with HL to detect and process auditory stimuli, over time, leads to the subjective experience of fatigue. While intuitive, empirical
work examining this linkage is limited. In this paper we discuss recent work that systematically examines this assumption.

In one study, subjective measures of effort and fatigue were obtained following a cognitively demanding speech task. Coordinate Response Measure (Bolia et al., 2000) messages were presented in noise. Study participants were younger adults (18-43 years; mean = 26.1 years) with and without HL. Participants listened for target numbers while ignoring other numbers and pressed a button when targets occurred, providing a measure of attention and processing speed, and then recalled the Call sign and Color associated with the target number. Task difficulty/effort was varied for normal hearing participants (n=12) by changing the signal-to-noise ratio (SNR; -2, -4, and -6 dB). For participants with HL (n=12) task difficulty was varied by completing the task with and without hearing aids (-2 dB SNR, n=11; 0 dB SNR, n=1).

Results suggest relationships between subjective effort and changes in fatigue (fatigability) differ in persons with and without HL. Although mental effort systematically varied with task difficulty for both groups; changes in fatigue varied with task difficulty only for persons with HL- with aided listening reducing fatigability despite the fact that speech understanding ability was only minimally improved (~7 percentage points).

In addition, ongoing work examining the effects of SNR on speech processing-related fatigability in older (60-80 years) adults with hearing loss will also be discussed. This work uses a similar, but more cognitively complex, speech task than our first experiment and focuses on the relationship between SNR (+2, 0, -2, and -4 dB SNRs) and fatigue when listening unaided. Preliminary results confirm that sustained speech-processing demands can be fatiguening for this group. However, in contrast to the unaided/aided fatigue effects observed in our first study, preliminary results suggest that when listening unaided, variations in SNR have a limited impact on fatigue in this group- even though understanding varies widely with SNR (~40 percentage points). Implications of these findings will be discussed. [Research supported, in part, by grants from the NIH (R21 DC012865-01A1, NIDCD Short Term Research Traineeship-T35). IES #R324A110266, the

ASHFoundation, the Dan Maddox Foundation and Oticon, Inc.]

B27

Listening effort and speech intelligibility in reverberation and noise

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Subjectively rated listening effort and speech intelligibility (Oldenburg sentence test) were determined in different adverse listening situations incorporating noise and reverberation. The situations were selected for similar values of the speech transmission index (STI) derived with noise as well as reverberation only, and combinations of noise and reverberation. For this purpose, noise was mixed with speech at different signal-to-noise ratios and the signals were convolved with exponentially decaying white noise or with impulse responses from real rooms with selected reverberation times. 21 normal-hearing and 20 hearing-impaired listeners participated. The presentation level was about medium loud for normal-hearing listeners and individually adjusted to the same loudness perception for hearing-impaired listeners. The STIs were set to a reasonable intelligibility range for both subject groups. In general, listening effort decreased and speech intelligibility increased with increasing STI. Mostly, same STI conditions resulted in similar subjective ratings and in similar speech intelligibility scores. Nevertheless, normal-hearing listeners rated lower listening effort and reached higher speech intelligibility scores in reverberation only than in noise only at the same STI. In contrast, hearing-impaired listeners experienced higher listening effort and achieved lower speech intelligibility for reverberation derived with real impulse responses than for reverberation derived
with white noise or for noise interference. The agreement between listening tests and STI can be improved by incorporating frequency-dependent signal properties as well as hearing loss. Since speech intelligibility shows a ceiling effect at medium STI values while listening effort is not yet negligible, both measurement methods taken together can capture the impact of different degrees of reverberation and noise over a wide range of conditions.

**B28**

**Dynamic spatial acoustic scenarios in multichannel loudspeaker systems for hearing aid evaluation**

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With increasing complexity of hearing aids the reproduction of realistic and complex acoustic environments in the laboratory gains more and more attention. This contribution presents a method of creating virtual realistic acoustic environments for this purpose. The system simulates moving sound sources with a first-order image source model in higher-order ambisonics (HOA). Diffuse background noises and late reverberation is added in first-order ambisonics (FOA). By combining the simulation of sources with recordings from real environments a high degree of plausibility can be reached with comparably low computational effort, allowing real-time processing and interactive listening tasks, including listener movement in the acoustic scene. From the scene description, both multichannel loudspeaker output and binaural signals can be rendered.

Several listening scenarios relevant to hearing aid research were defined and rendered with the described simulation method. The scenarios comprise in-door communication scenarios, in-door passive listening scenarios, and out-door environments, including public transport. Case studies using these scenarios are presented.

Furthermore, the general suitability of HOA for applications in hearing aid research, and its limitations, is shown in a simulation study. The data show that the minimally required number of reproduction loudspeakers depends on the class of hearing aid algorithm involved in the evaluation, as well as on the required frequency range. [Work funded by DFG FOR1732.]

**B29**

**Binaural wind noise cancellation algorithms in hearing aids using HRTF cues**

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With an increasing number of active hearing aid (HA) users in recent years, opportunity to communicate by speech has increased even in a situation at outdoor. HA users, however, often suffer wind noise which degrades a desired speech in windy situation. According to the survey of MarkeTrak (Kochkin, 2010), wind noise still continue to be major cause of dissatisfaction for HA users in listening environment.

Several algorithms with various approaches have been proposed to suppress and detect wind noise. In a single microphone approach in wind noise cancellation, it was proposed that wind noise is detected from a distribution of spectrum and then signal of low frequency bands are appropriately reduced using the ratio of a total power and low frequency bands of a monitored spectral shape for a long term (Kates, 2007). On the other hand, it is known that the cross-correlation of front and rear microphones was utilized to detect the wind noise in two microphones approach (Pertersen, 2004). Although the background noise due to the wind can be attenuated by using these approaches, a perceived directionality of desired signal is also simultaneously degraded, i.e. the localization cue is partially lost, because these are managed to attenuate only low frequency bands. The inability to correctly localize sounds puts the hearing aid user at disadvantage. The sooner the user can localize a speech signal, the sooner the user can begin to pay attention to that direction. The advantage of preserving the spatial separation between the desired signal and the interfering signals leads to large improvement in speech intelli-
In this study, we propose the binaural wind noise cancellation (BWNC) algorithm with preservation of localization cues. The proposed BWNC features comparing with existing algorithms are as follows; (1) The BWNC reconstructs the segregated desired signal as a binaural form, not bilateral. (2) The BWNC utilizes IPD and ILD cues of each frequency bin to detect a wind noise using the reference direction of arrival map, which can be estimated from appropriate HRTFs. In order to assess the proposed method an objective evaluation for perceived direction was performed by "Perceptually Motivated Measurement of Sound Width and Location" (Brookes et al., 2005) compared to bilateral wind noise cancellation. In our simulation experiments, wind noise could be detected with high accuracy by using proposed method while maintaining the localization cue of the desired signal.

B30
Contributions of spectral details of the direct and reverberant components to sound externalization
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In reverberant environments both the direct and the reverberant component of the binaural room impulse response (BRIR) are thought to contribute to the externalization of sound sources. The head-related transfer function and the reverberation contribute to the spectral details of the BRIR. These spectral details can be determined by measuring the BRIR, which virtually can create externalized sound sources when applied to signals and presented over headphones. This study investigated how reductions of the spectral details in the BRIR affect the degree of perceived externalization of virtual sound sources. The reduction of the spectral details was realized by smoothing the magnitude responses of either the direct or the reverberant component of the individual BRIRs. The smoothing was realized by averaging the magnitude responses of either the direct or the reverberant component of the BRIRs with gammatone-shaped filters. This was done for various bandwidths of the gammatone-shaped filters to achieve different degrees of smoothing. The individual BRIRs were recorded at a distance of 1.5 meters from the test subjects for azimuth angles of 0 and 50 degrees. The recordings were carried out in a reverberant listening room designed in accordance with the IEC 268-13 standard. Both the unmodified and modified BRIRs were then convolved with band-pass filtered noise (50-6000 Hz) and presented over headphones. To provide a visual reference the test subjects were seated in front of the loudspeakers placed at the measurement positions. The test subjects were asked to remain still and judge the perceived position of the sound source between the loudspeaker and their head. The data shows that reductions in spectral details of the direct component obtained with gammatone filters narrower than one equivalent rectangular bandwidth (ERB) do not affect sound externalization. However, reductions with filter bandwidths larger than one ERB lead to a gradual internalization of the virtual sound sources. Furthermore, reductions in spectral details of the reverberant component did not affect sound externalization to the same degree as in the case of the direct component. Finally, a larger reduction in externalization was observed at 0 degree azimuth than at 50 degrees azimuth.

B31
A newly-developed transducer for an occlusion reduction system optimized by the Filtered-x LMS technique
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The occlusion effect is one of the crucial issues for hearing aid users. It is caused by undesirable increase of sound pressure of the own voice in the ear canal occluded by the hearing aid itself. The amount of the increase is generally indicated from about 20 to 30 dB below 500 Hz. In order to reduce the increased sound pressure, several
studies have been conducted based on the active noise cancellation theory and classical feedback control theory by using additional microphone to monitor the sound pressure in the ear canal. However, it can be said that the amount of the reduction is insufficient to resolve complaints of users with severe occlusion effects. One of the reasons why it is hard to obtain the sufficient reduction is that there is large variation in a phase response of the transfer function from the receiver to the monitor microphone which may often cause oscillation to disturb the system to provide sufficient feedback gain for reduction. As another reason, it is difficult to design a highly-performed compensation filter because the procedure for designing of the filter had been liable to heuristic approaches.

In this study, in order to obtain sufficient amounts of the reduction, the occlusion reduction system is improved from two kinds of technical aspects. Firstly, a newly-developed transducer which has two kinds of unconventional structures is proposed as follows; (1) A smaller hole for pressure equalization of the microphone is employed to minimize the phase variation. (2) The housings of the receiver and the monitor microphone are integrated in order to reduce the variation of the phase response compared with separate receiver and microphone. Secondly, an automatically-optimized method for the design of the compensational filter is proposed based on the Filtered-x NLMS technique.

Evaluation of the proposed system consists of the newly-developed transducer and the compensation filter designed with an adaptive filtering approach shows that the system gives the amount of the reduction of 25 dB at around 300 Hz. Therefore, these results suggest that the proposed system reduces occlusion sufficiently even for hearing aid users who have severe occlusion effect.

Anecdotal evidence suggests that the current battery changing process presents challenges for hearing aid users, particularly those with strength and dexterity limitations. However, reliable experimental evidence on the type and magnitude of those challenges is lacking.

During the first phase of this study, an experiment was designed and run to quantify challenges inherent in loading and unloading size 312 and size 10a batteries in existing hearing aids. Sessions were video recorded and reviewed to find problematic strategies commonly used by participants. The experiment found that the smaller 10a batteries were more difficult for participants, regardless of age. Seven types of loading and unloading strategies were extracted from the video data. It was concluded that hearing aid users would benefit from a battery that does not need to be removed from a tray, and that new battery housing designs should accommodate one or more of the observed strategy types. To accommodate this need, we created three prototypes consisting of a hearing aid and a removable module that can have either a 312 battery or an integrated rechargeable battery. The module designs that we created are compatible with both battery types.

In the second phase of the study, participants were recorded while loading and unloading the three candidate devices and, as a control, a ReSound Alera hearing aid with a 312 battery. Time required to load and unload the candidate devices was much shorter than the time required to load or unload the 312 battery, and was not dependent on participant age. Strategies for loading the candidate devices also improved, in that fewer attempts were needed to load prototype battery modules than 312 batteries with ReSound Aleras.

These results demonstrate that the prototypes under investigation all require less time and effort (as measured by number of attempts) to load and unload than do existing batteries and battery door designs. Based on this information, all three designs can be recommended, with the caveat that further work must be done to develop color, texture, and other ergonomic features of the new device.
Assessing the effects of hearing aid provision using an audio-visual ‘online’ measure of speech-in-noise processing

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Conventional speech audiometry is often insensitive to the effects of hearing aid (HA) provision on higher-level (e.g. cognitive) processes. Recently, Wendt (2013) developed an eyetracking paradigm that allows estimating how quickly a participant can grasp the meaning of complex audio-visual stimuli (the 'processing speed'). In this paradigm, an acoustic sentence-in-noise stimulus is presented concurrently with two similar pictures, only one of which depicts the meaning of the sentence correctly. Wendt found that hearing-impaired listeners with HA experience fixated the correct picture more quickly than hearing-impaired listeners without HA experience.

The aim of the current study was to explore the effects of various HA conditions on the processing speed of both novice and experienced HA users (N = 15 per group). Firstly, we compared the processing speed achievable with NAL-R gain prescription to that achievable with gains prescribed in accordance with a 'spectral shaping' approach after Humes (2007). This approach ensures good audibility across the stimulus frequency range for each individual listener, thereby increasing the influence of cognitive factors on speech-in-noise performance. Secondly, we assessed the effects of single-channel noise reduction (NR). Several recent studies have demonstrated an influence of cognitive factors on listeners’ response to NR processing (e.g. Neher et al, 2013), leading us to explore the effects of this type of HA signal processing on listeners’ processing speed here. Thirdly, we investigated if any observable changes in processing speed depend on individual differences in HA use, hearing loss, age, or cognitive ability. Finally, we also assessed the test-retest reliability of our processing speed estimates.

We will present and discuss our results with a focus on the differences in processing speed between the two groups of listeners and the different HA conditions. [Funded by the DFG Cluster of Excellence EXC 1077/1 “Hearing4all”.

References:

Objective assessment of speech in noise abilities and the effect of amplification in children with hearing loss

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Children with mild to moderate hearing loss (MMHL) represent an estimated 11-15% of school-age children in the U.S. (Bess & Humes, 2008). Although there is a paucity of research in this population, it has been consistently shown that children with MMHL require a more favorable signal-to-noise ratio (SNR) for optimal speech understanding (Crandell, 1993). Despite this, they spend the majority of their time in classrooms that exceed recommended noise levels (Walinder et al., 2007), which may contribute to increased fatigue and academic difficulties. This study was designed to 1) evaluate neural efficiency of speech sound detection and discrimination in children with MMHL compared to children with normal hearing; and, 2) examine the effects of amplification on detection and discrimination of speech in noise in children with MMHL.

The P1-N1-P2 complex (Naatanen & Picton, 1987) and the P300 response (Sutton et al., 1965) are cortical auditory-evoked potentials (CAEPs) that have been used to assess objectively a listener’s central auditory speech detection and ability to actively discriminate speech contrasts, respec-
tively. When stimuli are presented in noise, normal hearing listeners show changes in CAEPs dependent on the SNR of the stimuli (McCullagh et al., 2012). These responses are also affected by the listener’s degree of hearing loss (Oates et al., 2002) and the signal processing of the hearing aid (Korczak et al., 2005); however, the effects of hearing loss and amplification on CAEPs using speech stimuli in noise are not well understood.

A passive oddball paradigm with two consonant-vowel syllables presented at 75 dB SPL embedded in multi-talker babble (+10 dB SNR) was used to measure the temporal N1 response and the central-parietal P300 response in 6-12 year old children with normal hearing (n=26) and children with MMHL (n=15). To examine the effect of amplification, children with MMHL were tested with and without personal hearing aids.

Latency, amplitude and morphology of the N1 and P300 responses were examined. Preliminary data suggest that children with MMHL show less well-defined N1 and P300 responses when compared to children with normal hearing. Amplification increased N1 and P300 amplitudes, but not to the level observed in children with normal hearing. These results suggest that children with MMHL show poorer neural efficiency of speech in noise and that the use of personal hearing aids only partially improves neural markers of detection and discrimination.

References:

B35
Preferred aided listening levels for music in the sound field
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In audiology literature, there is a dearth of research involving music listening through hearing aids. Existing reports generally focus on the electroacoustic limitations of modern hearing aids and perception of sound quality with a variety of hearing aid processing schemes. In the present study, we documented participants’ preferred aided listening levels (PLLS) for music presented in the sound field.

Ten adults with hearing loss were fitted bilaterally with six different hearing aids, each from a different hearing aid manufacturer. Each pair of hearing aids was programmed to a set of default music listening parameters. Participants were given manual control of presentation level for three music samples, each representing a different music genre. The data show that intra-individual preferences were consistent across hearing aid types but inter-individual differences were as high as 25 dB. Further analyses aimed to characterize relationships between participant and prescriptive factors that may contribute to the observed PLLs. The novel learning provided by this study suggests that aided PLLs for music are more affected by individual preference than the prescriptive characteristics of the hearing aid. In this regard, the ability for a patient to manually adjust music level or hearing aid output is an important consideration for music listening.
**Spatial Audio Quality Prediction using a Binaural Auditory Model**

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This work aims at developing a binaural quality model to be applied for the prediction of perceivable spatial audio quality. A double-ended (reference signal-based) approach is applied. The estimated perceivable spatial differences between two audio signals are defined as spatial audio quality.

To extract spatial properties from the audio signals, a recent binaural auditory model (Dietz et al., 2011) is employed as a front-end of the spatial quality model. The auditory model is able to provide perceptually relevant binaural features for the reference and test audio signal. Binaural features used for the quality model are interaural time differences, level differences and a feature related to the interaural coherence. The back-end has the purpose to derive intermediate measures from the binaural features for specific spatial quality subcategories (e.g. envelopment, localization) and to compare these intermediate measures between reference and test signal. Multivariate adaptive regression splines (MARS) are used to combine the intermediate measures to the final spatial quality measure.

A database containing binaural audio signals with various spatial quality differences and corresponding subjective quality assessments was generated for the development and evaluation of the model. Spatial quality differences between the signals were introduced by different binaural hearing aid algorithms and by artificial signal manipulations.

Since purely binaurally perceivable differences between audio signals represent rather special cases of audio quality, the spatial quality model should also be applicable in combination with a monaural quality model to build an overall quality model. Therefore, the spatial quality measure developed in this work was combined with the monaural quality measure for linear/spectral distortions proposed by Moore & Tan (2004) and another monaural quality measure mainly accounting for nonlinear distortions (Huber & Kollmeier, 2006) via multivariate adaptive regression splines.

First evaluation results show that different types of spatial quality differences can be predicted fairly well. These results are promising and support the approach to apply a binaural auditory model as a front-end for a binaural quality model.

Next steps include the extension of the database and further analysis of the relation between monaural, binaural and overall quality. Eventually, an adaption of the front-end to account for individual hearing impairments is planned.

**Non-linear frequency compression: exploring the role of auditory training—Preliminary findings**

Ann-Marie Dickinson, Richard Baker and Kevin Munro

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**Aim:** To evaluate the role of auditory training in adaption to non-linear frequency compression (NLFC).

**Design:** A between group design compared changes over 4 weeks in two groups of NLFC users; one who completed home-based, high-frequency focussed, auditory training and one who completed sham training (listening to a talking book). Detection and recognition thresholds, word recognition test scores (including reaction time measures), speech in noise recognition and
listening self-efficacy were used to assess outcome.

**Study sample:** Twenty-six experienced adult hearing-aid users completed the study (N=13 in each group). Allocation to group was pseudo-randomised to ensure confounding variables such as hearing loss, age and cognition were matched.

**Results:** Participants who completed the auditory training showed significant improvements in SRT post-training compared to the control group (mean 1.9 dB, SE 0.9, p=0.046). Effect size was large (d=0.8).

**Conclusions:** Preliminary conclusions are that auditory training significantly improves speech recognition in noise. This suggests that auditory training may be beneficial for adults using NLFC devices.

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**B38**

Structural similarity as a metric for analysis of the quality of acoustic signals

*Leonard E Cornelisse*

*System Performance Science, Unitron*

Structural Similarity is an analysis technique that has been used for accessing the quality of a compressed picture (digital image) as compared to the original uncompressed image. A metric similar to the Structural Similarity (SSIM) index has also been used to access the neural activity of auditory models (Michael R. Wirtzfeld and Ian C. Bruce, ARO, 2013). The purpose of this project is to explore Structural Similarity as a metric for sound quality, when applied to analysis of spectrograms of recorded audio signals.

The calculation formula for SSIM is:

$$SSIM = \frac{(2u_xu_y)(2\sigma_{xy})}{(u_x^2+u_y^2)(\sigma_x^2+\sigma_y^2)}$$

where:

- $u_x$ the average of x
- $u_y$ the average of y
- $\sigma_x^2$ the variance of x
- $\sigma_y^2$ the variance of y
- $\sigma_{xy}$ the covariance of xy
- And x & y are the frequency dependent levels of the two spectrograms being compared.

A mixed speech plus noise signal, at various SNRs, was recorded simultaneously, in both un-aided and aided conditions (test system to be described in separate poster at IHCON), and different signal processing strategies. The noise was located at +/-45° front and back. Spectral analysis of the recorded .wav files was performed using a standard signal analysis package (SpectraPlus).

The SSIM was calculated point-by-point, between two spectrograms, to observe the impact of the change in the comparison condition against a reference condition. Simple correlation between the comparison and reference spectrograms was also calculated.

Results will be presented for repeatability of measurement (multiple recordings), impact of SNR on SSIM, sensitivity to between ear differences, and sensitivity to signal processing.

The results suggest that the SSIM can be used to judge relative magnitude of the difference or similarity of a comparison to a reference recorded audio signal. When several comparison recordings are analyzed relative to a single reference condition, then the differences in the SSIM are indicative of audio quality relative to the reference. That is, a poorer SSIM indicates that the comparison signal is less similar to the reference signal than a comparison signal with a higher SSIM.

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**B39**

Education of audiologists to maximize hearing aid outcomes

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**Introduction:** Hearing aids are the major tool available for rehabilitating people with mild to severe hearing loss. The outcomes achieved depend on many factors, including the severity of hearing loss, the quality of the hearing aid’s
sound processing, the adjustment of the hearing aid parameters to the needs and preferences of the user, previous hearing aid experience and user’s attitude. In addition, it is obvious that hearing aids can have no effect if they are not used, as is the case with about 75% of people who would benefit, in developed countries such as Australia and USA. Evidence supports that many hearing aids in 2014 give improved benefits compared with hearing aids in 2004. Whilst there is a need for counselling with some clients, this is not true for the majority of people with mild to moderate loss. There is a need for audiologists to evaluate recent developments in hearing aid technologies, pre-fitting and post-fitting measures, and rehabilitative techniques.

**Goals:** This study investigated the postgraduate education programs available to meet the needs of audiologists in 2014. The aim was to ascertain the time spent and the goals for audiologist education on hearing aid technology, fitting, and rehabilitation, including cost and duration of the course, and the main learning outcomes.

**Methods:** A quantitative analysis was performed on web-published curricula for 50 English speaking audiology courses (Master’s degree and Doctor of Audiology), across three continents. Qualitative analysis of open-ended questionnaires were used with Australian lead educators.

**Results:** The course curriculae often combined auditory rehabilitation with hearing aid studies. Audiology courses varied widely in the time spent on hearing aid related units, especially on the technical dimension of auditory rehabilitation. The time spent was expressed inconsistently across courses, ranging from “unspecifed”, to 10% of the course. Tuition fees ranged from $0 to $100,000. Core activities and practicum hours were diverse. Programs were also influenced by health funding models in each country.

**Conclusions and recommendations:** The data suggest that most audiologists spend part of their time fitting hearing aids but hearing aid studies make up a relatively small component of most courses. This is confusing for a prospective student interested in hearing aid technologies or in auditory rehabilitation, from the psychosocial and counselling perspective.
We simulated a natural complex auditory environment in the laboratory that aims to capture the experience of listeners in multi-talker environments. An important difference from conventional laboratory simulations is that listeners were asked to process a continuous flow of speech information in such a way as to give a measure of how well they got linguistic meaning as well as recognize speech. Subjects heard multiple talkers from different directions, each uttering different streams of natural speech (stories). Questions derived from the stories were presented sequentially on a screen, immediately after the occurrence of the relevant information in the story.

One consequence of using naturalistic listening scenarios is that they involve multiple sources of variability including, not only differences in performance across test conditions of primary interest, but also variations in difficulty across questions (item variability), as well as variations in performance across subjects due to different listening strategies (e.g., different allocation of attention across streams). To properly account for these multiple sources of variability, we formulated a hierarchical Bayesian model that included differences in attentional strategies and item variability. We show an application of the model to an experiment examining the effect of spatial separation between talkers on the ability of normal-hearing listeners to selectively attend to a primary talker while deriving, if possible, information from a secondary or less important talker. Based on subjects’ performance with different spatial separation between two talkers, the model supported the evidence of two spatial bands, an exclusive band that favors larger spatial separation between talkers to isolate primary speech from interference, and an inclusive band that allows better monitoring of non-primary stimuli when they are located closer to the primary.

We also show new results from hearing-impaired listeners, looking at the benefit of aiding when the subject had to focus and understand the meaning of speech from a target talker, not just recognize speech, despite interference from another talker. Subjects also completed a battery of cognitive tests. The results indicate that the benefit of wearing hearing aids for understanding the target talker is influenced by hearing loss and cognitive abilities.
speech communication in that noise environment. The accuracy of this estimate depends on the relationship between intelligibility measures obtained using a stationary energetic masker and intelligibility in the real-world noise environment, which may be non-stationary and may be comprised of both energetic and informational maskers.

This research examined the effects of informational maskers on the accuracy of these predictions. Calibrated recordings from 75 real-world noise environments were classified in four categories: mechanical/equipment noise, multi-talker babble, music, and clear speech. All recordings were made at locations in large cities. A sample of 24 individuals with normal hearing and 5 hearing impaired individuals participated in the study. ESII-intelligibility functions were obtained for each individual using HINT sentence lists and a stationary energetic masker (the HINT reference noise masker). These functions were used to determine ESII values that would yield 25%, 50% and 75% intelligibility for each subject. Percent word intelligibility for HINT sentence lists was then measured at the identified ESII values for each level of intelligibility. These measures were obtained in three of the four categories of noise environments, excluding the music noise environment. Scatterplots comparing predicted and obtained intelligibility were produced for each noise environment.

Results showed that measured intelligibility was accurately predicted for mechanical/equipment noise, a non-stationary energetic masker. Intelligibility predictions were slightly less accurate for multi-talker babble, a mixture of non-stationary energetic and information masker, and were overpredicted by about 10%. Intelligibility predictions were even less accurate when clear speech was used as a non-stationary informational masker, and were overpredicted by up to 20%, depending on the target intelligibility.

These findings suggest that ESII-based intelligibility predictions may be best suited for use with energetic maskers, which is consistent with the rationale for the ESII. Clear, intelligible speech carries linguistic information that may make it more difficult for individuals to segregate target speech from speech interference, suggesting that energetic and informational masking engage different perceptual mechanisms.

C3
Unveiling the source of language delay for children with hearing aids
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Children must acquire a variety of language and literacy skills in order to succeed in academic endeavors and social relationships. These skills fall along a continuum from ones largely dependent upon detailed phonological systems to ones more morphosyntactic in nature. Hearing loss can present challenges to the acquisition of all those skills, for two reasons: (1) sensory information used to form linguistic representations is degraded; and (2) opportunities to interact with the ambient language are restricted, so experience is diminished. The first of these problems would be expected to most strongly influence language and literacy skills dependent on phonological structure. The second should affect all skills. Understanding how each of the two challenges influences acquisition across this range of skills for children with hearing aids (HAs) should help in the design of intervention programs. The goal of the current investigation was to explore these relationships.

Sixty-eight children (49 with NH and 19 with HAs) were tested with 16 language and literacy measures that were predicted to range between ones strongly dependent on having a detailed phonological system and those strongly morphosyntactic in nature. Two-component factor analysis confirmed the predicted designations of tasks as either more phonological or morphosyntactic in nature. When three-component factor analysis was performed, a third factor was uncovered that appeared lexical in nature. Several measures loaded heavily on this factor, suggesting a third category of language and literacy skills. This category of skills likely depends on acoustic structure that is not as fine-grained as acoustic cues to phonetic categories, but rather is more global in structure. Subsequently, when Cohen’s $d$s were computed to index effect sizes of the differences between groups (NH vs. HA), it was found that
children with HAs performed most differently from children with NH on measures that were largely phonological in nature; they performed least differently on measures that were mostly morphosyntactic in nature, and immediately on measures that loaded heavily on the lexical factor.

In summary, outcomes of this analysis indicate that children with HAs experience language deficits largely arising from the degraded sensory information available to them, which interferes with acquisition of detailed phonological systems, and to a lesser extent, lexical skills. These children apparently acquired morphosyntactic skills more easily, meaning that sufficient experience was available. These outcomes suggest that intervention approaches should be designed to remediate phonological and lexical abilities and facilitate the acquisition of morphosyntactic skills as compensatory mechanisms.

### Relating hearing-aid gain settings to clinical outcome measures

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The setting of hearing-aid gains is a key component of the hearing-aid fitting process. Although much literature has been devoted to the development, merits, and limitations of gain-prescription formulas, open questions remain regarding the actual impact of deviations from recommended gains on patient satisfaction with the aids. In this study, we examined (a) deviations between clinician-adjusted real-ear aided gains (REAGs) and recommended gains (according to NAL-NL1, NAL-NL2, and Starkey’s proprietary formula, eStat), and (b) statistical relationships between these deviations and answers to clinical hearing-aid outcome questionnaires (SSQ-B, COSI, and IOI-HA) in 68 subjects with mild to severe high-frequency hearing loss. Gain was adjusted for subjects’ preference relative to an NAL-NL1 starting point at the first visit, and again one month later. Three questions were of particular interest in this context: (1) How are clinician-adjusted gains distributed around existing prescription formulas? (2) Do deviations, when present, differ between the initial fitting and one month later? (3) Do subjective outcome measures co-vary in a statistically, and clinically, significant manner with deviations from recommended gains? Results showed a relatively wide (+/-10 dB) distribution of clinician-adjusted gains around the mean. Although NAL-NL1 targets were used as a starting point for gain settings, after adjustment by the clinician, gains were very close to eStat targets on average. In addition, statistically significant relationships between deviations from gain targets and SSQ-B scores were identified. Clinical implications of these findings, limitations of this study, and new avenues for exploring statistical relationships between hearing-aid gain settings and clinical outcome measures will be discussed.

### Directional benefit measures in diverse background noises

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It has been a long-standing assumption that directional microphones provide a benefit when a target speaker is in the frontal looking direction and noise surrounds the listener. As a consequence, modern hearing aids include directional microphone schemes that automatically increase directional response as the estimated level of surrounding noise increases. Despite this, evidence about the subjective advantage gained by directional hearing aids in real world listening situations remains elusive. We report how binaural real world recordings from 30 different environments were used to examine the correlation between signal-to-noise (SNR) benefit and perceptual advantage from four microphones modes; omni, cardioid, cardioid + 5dB SNR (ideal super-
directional), and binaural beamformer, all presented under headphones. The SNR benefit from the different microphone modes was estimated using speech reception threshold in noise measures (SRTn), and these measures were compared to subjective measures of benefit which included acceptable background noise level, sound quality attributes ratings, and overall microphone mode preference. The study included a group of participants with 4 frequency average hearing levels (4FAHL) ranging from normal hearing through to severe hearing loss. The various tests were administered in the laboratory during 6 appointments over a 4 a week period. Our findings will be presented and discussed.

"You can lead a horse to water...": Initiating and supporting change on the older adult's journey to hearing health care.

Heather Holliday, Lorienne Jenstad, Garnet Grosjean and Barbara Purves
University of British Columbia

Age-related hearing loss (ARHL) is a health problem with wide-spread implications. It is the third most common chronic condition in older adults in Canada and the United States (CASLPA, 2005). Despite its common occurrence, wide-spread consequences and effective management options, ARHL is left untreated by the majority of older adults who are affected by it. Understanding the reasons for low uptake of management options can inform how hearing health might be supported.

The term “hearing health change” is used here and is meant to encompass any step that a person takes, or any change that he or she makes that propels the individual towards improved ability to communicate. This could include scheduling a hearing test, talking to a doctor, or pursuing amplification. Resistance to taking these steps towards hearing health change may be due to many factors such as perceptions of susceptibility, benefits, and barriers; self-efficacy and outcome expectations; and, lack of access to appropriate and trusted information (Cox et al., 2005; Egger et al., 1999; Hickson & Scarinci, 2007; Winsor, 2011).

In the present study, an information-sharing presentation, Hearing Health in Older Adults, was designed with the seniors’ advocacy group Council of Senior Citizens’ Organizations of British Columbia (COSCO). The presentation combined participatory action learning, peer teaching, peer learning, and narrative case studies to promote the health literacy of older adults about hearing health.

The purpose of this investigation was two-fold. The first aim was to evaluate the presentation. The second, broader aim was to explore hearing health change from the perspective of older adults. Four minimally-led focus group discussions were held following the COSCO presentation Hearing Health in Older Adults. The data from these discussions were analyzed using the inductive techniques of qualitative description and thematic analysis. The dialogue that ensued was varied, yet had several common threads. Five central themes emerged under the overarching theme of initiating and supporting change: Recognizing and Admitting hearing loss; Understanding the Options; Sharing Stories and Experiences; Barriers and Facilitators; and Peer Teaching. These themes will be described in detail and discussed in relation to models of health behaviour change.

A hierarchical model for the analysis of intra-individual variability

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Starkey Hearing Technologies

When analyzing and reporting results of studies involving multiple participants, researchers are often faced with two choices: average data across participants to estimate group-level effects or treat each participant separately. Neither of these two options is entirely satisfactory: averaging data across participants ignores interindividual differences and treating each participant as a separate entity ignores commonalities across participants. Hierarchical (i.e., multi-level) Bayesian models (HBM$s$) provide a principled solution to this conundrum$^{1,2}$. In a HBM, estimates of performance or treatment effect for a given participant are informed by data from other participants.
The pulling of information across participants is automatically adjusted, such that when less contribution is needed (because individual data are not sufficient to produce precise estimates) or warranted (because individual data are abundant), less pulling is applied. The result is that, in most cases, estimates of performance or of treatment effects computed using HBMs are both more precise and more accurate than estimates of performance obtained using more traditional methods. We illustrate an application of HBMs to the analysis of audiological data concerning the treatment effects of directional microphones on speech recognition in noise.

References:

C8
An auditory model-based approach to dynamic compression
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Sensorineural hearing loss typically results in elevated thresholds and steepened loudness growth with level significantly conditioned by a loss or dysfunction of outer hair cells (OHC). In hearing aids, amplification and dynamic compression aim at restoring audibility and at widening the limited dynamic range. However, success usually shows large inter-individual variability when considering, e.g., speech perception. In particular, hearing impaired listeners generally still have considerable problems in complex acoustic communication situations.

Physiologically motivated approaches to dynamic compression may help by directly replacing damaged or absent OHC function.

An auditory-model based, fast-acting dynamic compression algorithm (MDC2) is introduced which aims at restoring OHC functionality. In the compression algorithm instantaneous gains are derived from a comparison of normal-hearing and hearing-impaired basilar-membrane input-output function. Frequency analysis and resynthesis is realized using a gammatone filterbank implementation [Hohmann, Acta Acustica 88, 433 (2002)]. Off-frequency component suppression is achieved using an instantaneous frequency estimate. The algorithm is fitted by estimating low-level gain loss (OHC loss) from adaptive categorical loudness scaling data and audiometric thresholds based on Ewert and Grimm [in: proc. ISAAR (2012), 393] and Jürgens et al. [Hear. Res. 270, 177 (2011)]. In a revised version of the compressor, a new stage prior to resynthesis was introduced, that applies a gain correction based on the amount of filter widening expected in the individual impaired system.

In a technical evaluation, the suggested compression algorithm and its different stages were compared to a conventional multi-band dynamic compressor with well-established gain prescription (NAL-NL1). Although the more complex model-based instantaneous compression comes along with slightly higher distortion components, other measures reveal its potential to restore perception for a wide range of input signals related to, e.g., the resulting signal-bandwidth dependent gain.

The considered algorithms were implemented for real-time operation using the “Master Hearing Aid” signal processing platform [Grimm et al., Acta Acustica 92 (2006)]. Aided speech reception thresholds were measured for 30 hearing impaired subjects with moderate sensorineural hearing loss in a variety of spatial conditions. Moreover, after testing the system in a real world environment, the participants were asked to fill in a questionnaire including, for example, listening effort. As far as completed, aided speech reception thresholds in stationary and fluctuating noise were comparable to conventional compression strategies. [This work was funded by the BMBF]
Does susceptibility to proactive interference predict hearing aid benefit?
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Proactive Interference refers to an effect whereby previously learned information interferes with the capacity to learn new information. It has been suggested that performance on complex tests of working memory may be heavily influenced by susceptibility to proactive interference (Kane and Engle, 2000). There is a wealth of evidence linking performance in complex working memory tasks to speech-in-noise recognition by listeners with normal hearing and those with a hearing loss in both aided and unaided conditions (see Akeroyd, 2008 for a review). Previous research has shown that susceptibility to proactive interference can be used to predict speech-in-noise recognition in young listeners with normal hearing (Ellis and Rönnberg, in press). We will report results that show whether this finding can be replicated in older adults with hearing loss and whether susceptibility to proactive interference can be used to predict the degree to which hearing aid use improves speech-in-noise recognition in these listeners. Potential clinical implications relating to the use of tests of proactive interference in the hearing aid fitting and rehabilitation processes will be discussed.

References:

Musician and non-musician preferences for hearing aid gain and compression settings
Kristen D’Onofrio, Erin Picou and Todd Ricketts
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At an initial hearing aid fitting, level dependent gain and output settings are commonly determined from prescriptive targets based on validated methods. Because these prescriptive methods were developed primarily to maximize speech understanding while also considering comfort, it is unclear whether they are appropriate for other important sounds, such as music. Further, it is uncertain whether preferences for hearing aid settings differ for individuals with little to no musical training compared to individuals with considerable musical training. The purpose of the present study was to evaluate potential group differences between musicians and non-musicians in their self-adjusted gain and compression settings for both music and speech stimuli. In addition, speech recognition, sound quality, and strength of preference for the self-adjusted (SA) settings and the original prescriptive (NAL-NL2) settings were compared. Musician and non-musician matched pairs with varying degrees of symmetrical high-frequency sensorineural hearing loss were initially fitted to NAL-NL2 prescriptive targets via probe microphone techniques. All participants made adjustments to the hearing aid settings for a speech stimulus and two music stimuli in an effort to achieve their own SA settings. Stimuli were presented three times each in a counterbalanced blocked order, and all adjustments were made in real-time using a manufacturer-specific computer-based signal processing tool which allowed for changes to overall gain, frequency shape, and compression ratio. Speech recognition in quiet and noise, sound quality, and strength of preference ratings using paired comparisons were then assessed at both the NAL-NL2 settings and the participants’ SA settings. Data collection is ongoing, but preliminary findings from 6 musicians and 2 non-musicians show that both groups demonstrated a relatively consistent increase of approximately 3-
4 dB around 1500 Hz for all stimuli. Additionally, these data suggest that musicians may desire a similar frequency response for at least some styles of music, but a different frequency response for speech, while non-musicians may desire a similar frequency response for both music and speech. However, paired comparison data thus far show that listeners do not consistently show preference for their SA settings. Possible implications for hearing aid fittings of musicians will be discussed.

Subjective evaluation of interaural coherence preservation in MWF-based noise reduction algorithms for binaural hearing aids

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Noise reduction algorithms in hearing aids are crucial to improve speech understanding in background noise for hearing impaired persons. For binaural hearing aids, algorithms that exploit the microphone signals from both the left and the right hearing aid are considered to be promising techniques for noise reduction, because in addition to spectral information spatial information can be exploited. For the binaural Multi-channel Wiener Filter (MWF), it has been theoretically and experimentally shown that in case of a single speech source this technique preserves both the so-called Interaural Transfer Function (ITF), comprising the Interaural Level Difference and the Interaural Time Difference, and the Interaural Coherence (IC) of the speech component. On the other hand, the binaural MWF typically distorts the ITF and the IC of the noise component, such that both speech and noise components are perceived as coming from the speech direction and hence the binaural hearing advantage can not be exploited by the auditory system.

For diffuse noise fields, whose spatial characteristics can be properly described by the Interaural Coherence, two extensions of the binaural MWF have been proposed that aim to achieve binaural cue preservation for both the speech and the noise components: the MWF-N, mixing the binaural output signals with a portion of the input signals, and the MWF-IC, incorporating an IC preservation term into the MWF cost function. Since for both binaural MWF extensions a trade-off between IC preservation and output SNR exists, it has been proposed in [2] to control the amount of IC preservation based on the IC discrimination abilities of the human auditory system.

To subjectively evaluate the influence of the perceptually optimized trade-off between IC preservation and noise reduction, in this contribution we present the performance of the considered binaural algorithms (MWF, MWF-IC and MWF-N) with respect to speech intelligibility (SRT) and spatial impression for normal hearing subjects.

References:


Predicting benefit based on hearing loss for adults using non-linear frequency compression—Preliminary findings

Ann-Marie Dickinson, Richard Baker, and Kevin Munro
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Aim: To identify which severities of hearing loss, if any, gain benefit from nonlinear frequency compression (NLFC).

Design: An ABA design was used to compare outcome with NLFC disabled (A) and enabled (B). Outcome was measured at 4 test sessions held at 4 week intervals (4 weeks disabled → 4 weeks enabled → 4 weeks enabled → 4 weeks disabled). Detection and recognition thresholds, word recognition test scores (including reaction time measures), speech in noise recognition and
self-report measures were used to assess outcome.

**Study sample:** Twenty-one experienced adult hearing-aid users completed the study. High-frequency hearing thresholds ranged from mild to profound.

**Results:** Participants showed improved detection and recognition of /s/ and /ʃ/ when NLFC was enabled. Speech reception thresholds (SRT) showed a small but significant improvement of 0.74 dB (SD 1.1) when NLFC was enabled. When participants were split into mild-moderate or severe-profound groups only the severe-profound group showed a significant improvement in SRT when NLFC was enabled.

**Conclusions:** Preliminary conclusions are that NLFC significantly improves speech recognition in noise but this is dependent on hearing thresholds.

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**C13**

**Localization & speech reception in noise with a Microphone and Receiver in the Canal hearing-aid**

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**Purpose:** Natural hearing through better localization and speech reception in noise.

**History:** Microphone placement in hearing instruments has been a study of interest for a considerable time. With the initiation of in-the-ear type instruments, hearing instrument designers could now take advantage of the acoustical properties of the outer ear (Preves, 1976). In this way it would be possible to utilize the effects of the concha’s high frequency focusing power.

**Design:** Though advantages of in-the-ear microphone placement have been known for decades, a RIC, with its behind-the-ear microphone placement does not benefit from the pinna’s high frequency sound focusing ability. ExSilent, however, has designed a new type of RIC device called the MaRiC (Microphone and Receiver in the Canal). Unlike a traditional RIC, the MaRiC incorporates a small canal worn module that contains both the microphone and receiver. Since the microphone portion of the MaRiC sits down deep into the meatus, sound may be focused by the auricle to where the microphone is positioned. The MaRiC and its over-the-ear component make up the ExSilent Ytango (Hearing Review, April 2011).

**Focus:** The MaRiC is specifically beneficial for those hearing aid users, who frequently complain about the “unnatural sound” from regular and directional RIC instruments. The MaRiC preserves the natural sound field.

**Results:** Fitting experiences and research results of measurements with a significant group of hearing-impaired subjects will be presented. The MaRiC will be compared to a traditional RIC, focusing on the positive consequences regarding directivity and localization, but even more speech reception in noise issues while using the MaRiC. Besides this, more insight will be given into successful feedback cancellation, even for open fittings.

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**C14**

**Tuning of auditory attention: the importance for hearing aid acclimatization and benefit**

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**Background:** New hearing aid users report that background sounds are initially rather intrusive. Following regular hearing aid use, these sounds became less noticeable. Intrusive background sounds may adversely impact on speech recognition due to their distracting effects. Good selective auditory attention and regular hearing aid use, with consistent experience of background sounds, may allow users to selectively ‘tune out’ irrelevant background sounds and improve speech recognition in noise.

**Aims:** The present study examined the hypothesis that speech in noise perception improves over time in new hearing aid users. We further hypothesised that this improvement would be dependent on cognitive constraints (specifically,
selective auditory attention) and regular hearing aid use. We expected that improvements in speech perception over time in new hearing aid users would be higher for those with greater cognitive capacity and for those who use hearing aids consistently versus occasional users.

**Methods:** New adult hearing aid users (n=50) were tested longitudinally from the time of first hearing aid fitting on a battery of tests that included speech recognition in noise, auditory distraction and self-report measures of hearing aid benefit. Participants also completed measures of cognitive capacity, selective auditory attention and noise tolerance (acceptable noise level) at baseline. A control group of experienced adult hearing aid users (n=20) completed the same test protocol.

**Results/Progress to date:** The study is currently on-going and due for completion in late June 2014. We will present the first results from this study. [Work funded by Industry Research Consortium (IRC)]

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**C15**

**Speech recognition in diffuse noise: a comparison of bimodal hearing and bilateral cochlear implantation**

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A comparison of bimodal hearing and bilateral cochlear implantation was completed using a within-subjects, repeated-measures study design. We recruited 8 adult bimodal listeners each of whom ultimately received a second cochlear implant (CI). Speech understanding was assessed via the adult minimum speech test battery (MSTB) as well as for HINT and AzBio sentences in semi-diffuse noise with the R-SPACE™ sound simulation system. Listening conditions included unilateral hearing aid (HA) in the non-CI ear, unilateral cochlear implant (CI), bimodal (CI+HA), and bilateral CI. Three participants had bilateral hearing preservation and were thus also tested with bilateral CIs and bilateral hearing aids (BiBi). All participants had at least 6 months of CI experience (range: 0.5 – 6.4 years). Testing was initially completed in the bimodal hearing configuration and then repeated at least six months following activation of the second CI. All testing was conducted in the free field. For speech recognition in noise, the conditions included: 1) speech and multi-talker babble noise presented from a single speaker at 0 degrees azimuth, and 2) speech presented from 0 degrees and R-SPACE™ noise presented from 8 equally-spaced speakers surrounding the listener. For clinical measures of speech recognition (MSTB) with a single loudspeaker at 0 degrees, bilateral CIs provided no additional benefit over the bimodal condition for CNC words (p = 0.33) or AzBio sentences in quiet (p = 0.43); however, bilateral CIs yielded significant benefit over bimodal hearing for AzBio sentences at +5 dB SNR (t = 2.4, p = 0.02)—with 5 of the 8 participants demonstrating statistically significant benefit at the individual level using a binomial distribution statistic. For the R-SPACE™ adaptive speech reception threshold (SRT), bilateral performance was significantly better than bimodal hearing (t = 4.6, p = 0.002). In fact, all 8 participants achieved the same (n = 1) or better (n = 7) performance with a 4-dB improvement in the SNR, on average. The subgroup of 3 BiBi participants showed an additional 3.8-dB improvement over the bilateral CI condition—presumably due to the presence of interaural timing differences in binaural low-frequency acoustic hearing. These data suggest that for listening environments in which noise is present—particularly for diffuse noise environments—adding a second cochlear implant can significantly improve speech recognition. In addition, bilateral acoustic hearing can yield twice the benefit in noise as bilateral CIs when compared to the bimodal hearing configuration.

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**C16**

**Monaural versus binaural processing of spectral ripple stimuli**

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Broadband spectral resolution is a significant predictor of speech recognition in quiet and noise
for listeners with normal hearing and hearing loss (e.g., Henry et al., 2005). Recent studies have also highlighted the importance of broadband spectral resolution in predicting outcomes for non-linear hearing aid amplification (e.g., Davies-Venn and Souza, 2014). Broadband spectral resolution and temporal integration have mostly being assessed under monaural listening conditions. However, in naturalistic listening environments, binaural listening is often employed and thus involves integration across both ears. Several studies have suggested differences in narrowband frequency selectivity and modulation detection in monaural versus binaural listening (e.g., Verhey and Nitschman, 2014; van de Par et al., 2013). However, duration and level effects on binaural spectral modulation detection is not well understood. This study evaluated spectral modulation detection across different durations and levels using both a monaural and a binaural listening paradigm.

A within-subject repeated measures design was used to measure broadband frequency at different durations, levels and ripple densities. The spectral modulation detection task (Litvak et al, 2007) was used to measure listeners’ detection thresholds for a flat versus spectrally modulated signal. The spectral modulation detection threshold (SMT) was determined by varying the spectral contrast (i.e. peak-to-trough) of the test signal. The ripple stimuli was presented monaurally to both the left and right ear and also binaurally at different presentation levels.

Preliminary results suggests that listeners’ spectral modulation detection improves with increase in duration and worsens with increase in level. The effect of level is differential across durations and the effect of duration is differential across ears. In short, the relationship between binaural processing and monaural processing was influenced by the duration of the ripple stimuli. These findings are consistent with recent studies suggesting that the auditory system encodes spectral and temporal information differently. These results will be presented within the theoretical framework of monaural versus binaural temporal integration, asymmetrical sampling, and hemispheric asymmetry.

**C17**

**Binaural interaction of auditory evoked potentials in patients with Diabetes Mellitus**

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**Introduction**: Diabetes Mellitus (DM) is a metabolic disease that can lead to microvascular and neuronal complications like heart disease, stroke and neuropathy. Hearing loss is twice as common in adults with DM versus adults without the disease (Bainbridge et al., 2008), although the effects of DM are not limited to the periphery. Individuals with DM also show prolonged neural conduction times within the brainstem even after controlling for peripheral hearing loss (Diaz de Leon-Morales, 2005; Konrad-Martin et al., 2010). Further, Frisina and colleagues (2006) demonstrated that diabetics performed more poorly understanding speech in noise even in the easier condition of noise spatially separated from speech. This suggests that DM impacts binaural hearing. Since binaural comparisons occur at the level of the brainstem, we investigated the difference between monaural and binaural Auditory Brainstem Responses (ABR) to evaluate the effect of DM. By calculating the difference between the sum of the two monaural ABRs and the binaural ABR, we extracted a distinctly binaural potential known as the Binaural Interaction Component (BIC), representative of neural activity underlying central auditory processing of binaural stimuli.

**Purpose**: The goal of this research is to determine if Veterans with DM have a reduced BIC compared with non-diabetic controls.

**Methods**: ABR waveforms were obtained from 130 Veterans with unmanaged DM, managed DM and no DM based on medical diagnosis, as well as severity and control of the disease. The binaural response was subtracted from the aggregate of the two monaural responses (L + R) from wave V of the ABR to elicit the BIC. We measured the peak amplitude and area under the curve of each monaural, binaural and BIC waveform, and calculated amplitude ratios as BIC/(L+R). A regression model was used to identify group dif-
ferences in BIC after adjusting for potential confounders of age, hearing, noise exposure history and other factors.

**Results:** We expect to show that those Veterans with more severe or poorly controlled DM have a decreased or absent BIC of the ABR when compared to Veterans without DM.

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**C18**

Extended high-frequency bandwidth in simulated cochlear implant hearing

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Frequencies between approximately 0.5 and 5 kHz are often considered the most important for speech intelligibility. However, extended high-frequency cues have been shown to improve speech perception under certain conditions. Unlike acoustic hearing instruments whose effective frequency response is limited by the state of the cochlea, the frequencies delivered by a cochlear implant are more flexible. For example, cochlear implants are capable of delivering high-frequency information via direct stimulation of auditory nerve fibers. However, the optimal range and allocation of frequencies along the electrode array have not been well established. Cochlear implants typically code frequencies up to 7-8 kHz, but little is known about whether extended high-frequency cues may improve speech recognition or perhaps may even cause a detriment by reducing the number of electrodes allocated to lower frequencies. Moreover, optimizing the frequency allocation may depend upon the specific listening situation. The current study considers the role of high-frequency information up to 12 kHz in the recognition of vocoded speech. The use of cochlear-implant simulations was selected over experiments in cochlear implant recipients to control for confounds such as limited listening experience with high frequencies and acute degradations due to changing the frequency-place assignments. Normal-hearing listeners performed identification of plurals for female speech using vocoded stimuli processed with a range of high-frequency cutoffs. Noise conditions included speech-shaped noise and multi-talker babble in both co-located and spatially separated configurations. Perceived spatial separation was achieved over headphones with the use of head-related transfer functions. Results show that extending the high-frequency cutoff improves plural recognition, especially in spatially separated conditions. Additionally, the effects of high-frequency bandwidth and spatial separation demonstrate different patterns in the presence of either speech-shaped noise or multi-talker babble. These findings suggest that extending the high-frequency cutoff in cochlear implants may improve speech recognition in specific auditory environments.

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**C19**

Four ears may be better than two ears for A/B listening comparisons

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Often, in A/B listening comparisons, it is desirable to switch between the amplified sound at the hearing instrument wearers ear and the unamplified sound in unaided / open ears. Ideally, this is done in real-time (live) rather than listening to audio recordings. A test system set-up with the capability of real-time switching between unaided and aided conditions, using a pseudo-Kemar 4-ear simulation head, is described. The system uses standard G.R.A.S. ear simulators mount on G.R.A.S. KEMAR mounting plates attached to a specially designed frame such that there are two pairs of ears; one pair being situated directly above the other pair. The ears are horizontally separated to approximate the space of an average adult head. The ears are vertically separated so that there is a small amount of space between the bottom pair of ears and top pair of ears. In the test set-up described, the ears are placed in the centre of a circle of eight loudspeakers, inside an audiometric booth.

The poster will provide further details on how to create such a set-up and cover calibration challenges, the measured head related transfer function (HRTF) and interaural level differences for each loudspeaker azimuth and each pair of ears will be compared to the same measures obtained with a standard KEMAR.
Generalization of error tokens of hearing-impaired consonant perception

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What is the strategy of the hearing-impaired (HI) ear in detecting consonants? It is uniformly assumed that audibility is the main factor in speech perception, for both normal-hearing (NH) and HI listeners (Zurek & Delhorne, 1987). Based on an entropy measure, Trevino & Allen (2014) have shown that audibility is not the main issue for the HI ear. This observation is counter-intuitive. In this research, we hope to find answers to the following questions: What is the generalized strategy of each HI ear in detecting consonants? How can we determine the subject’s strategy? A classical consonant identification experiment is proposed, derived from Miller & Nicely (1955). To determine the strategy of the HI ear in detecting consonants, we study consonant group error patterns. The psycho-acoustic procedure is designed and implemented in such a way as to investigate errors of a specific well-defined set of tokens made by the HI subject. Only tokens having an SNR90 > 0 [dB] are investigated, meaning that NH listeners have <10% error for SNRs greater than 0 [dB], in speech-shaped thermal noise. There are three experimental phases: 1) We observe errorful tokens and randomly choose up to six. 2) We determine if the token’s errors are consistent, which requires audibility. 3) We generalize the errors with up to four strategies. S1: The frequency of the consonant’s primary cue is varied by changing the vowels, which slightly moves the cue frequency. S2: The conflicting cues are varied. Different tokens of the same consonant have different confusions, due to conflicting cues. S3: The masking of the primary cue is varied. The primary cue for many tokens of the same consonant-vowel is highly correlated with the NH SNR90. S4: The number of conflicting cues is varied, as measured by the error entropy. The entropy of a token tells us something about the number of conflicting cues and/or about the ambiguity of the primary cue. If we can establish error generalizability in the HI ears, we will gain insight into that ear’s decoding strategy.

Effective communication in real-world environments with hearing aids

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Although recent advances in technology have produced hearing aids that can provide substantial SNR benefit under laboratory conditions, experienced hearing aid users continue to complain that hearing aids provide limited benefit in noisy real-world environments. In part, this divergence between laboratory results and user experience may be related to the complexities involved in real-world communication environments, where listeners must be able to monitor multiple talkers and shift their attention in response to changes in the conversation. Laboratory environments also may not capture the dynamic behavior of talkers in the real world, who constantly adjust their voices in response to changes in the ambient sound level. The traditional approach to this problem has been to recreate the real-world noise environment as accurately as possible in the laboratory, usually through the use of advanced signal processing and distributed loudspeaker arrays. The purpose of this study was to explore a different technique for bridging the gap between laboratory measurements and real-world listening environments, where hearing aid users were asked to perform an interactive communication task while embedded in a noisy environment. Experienced binaural hearing aid users were fit with receiver-in-the-canal hearing aids that were programmed with four directional settings: omnidirectionality, asymmetric directionality, binaural directionality, and adaptive directionality. After a successful 4-week acclimatization period, these listeners where organized into 4-person teams who performed the task in a busy cafeteria with each participant set to a different directional setting in each block of trials. Each team member had a hand-held tablet which, on any given trial, instructed them to serve either as the “talker” (by saying a phrase from the Modified Rhyme Test, e.g. “You will mark den please”) or as a “listener” (by selecting a key word from closed set of six rhymes on the tablet). Accuracy and reaction time were measured for each listener in each trial, along with time series data of the overall sound level in the room and the partici-
The purpose of this study was to investigate: 1) the effects of distance on the acoustic properties of speech signals and 2) the effects of amplification, specifically WDRC-based amplification, on the acoustic properties of speech signals at a distance. Results of this investigation may form a future measurement protocol for evaluating distance perception.

We present two pilot experiments, involving field recordings of controlled scenarios. For both experiments, we recorded speech at varying distance in both unaided and aided conditions using a KEMAR. We conducted the aided condition recordings for a default fitting of a 40 dB flat hearing loss, with all adaptive hearing instrument features, except WDRC, disabled.

Study 1: Single sound source: In this study, we examined the perceptual effect of distance. We presented speech from a loudspeaker positioned at 5, 10, 15, 20 and 25 feet and 0° azimuth from KEMAR. We calibrated the speech signal for two presentation levels: 62 dB and 72 dB SPL measured at the 5 foot distance. We presented the same signal levels at all loudspeaker distances i.e., the signal presentation level remained constant regardless of distance to the point at which the level was measured.

Study 2: Multiple simultaneous sound sources: In this study, we examined the relative perceived loudness of simultaneous distant and near sound sources. We presented speech from one loudspeaker positioned 5 feet away from KEMAR to represent a primary conversation. A pair of loudspeakers was positioned 5 feet apart at a distance of 25 feet away from KEMAR to represent a secondary conversation.

Analysis and Implications: We developed custom software to facilitate comparative listening. We analyzed data for perceptual differences between unaided and aided recordings. Based on perceived differences, we compared spectral properties of unaided and aided data. These data are presented in this poster to initiate consideration of signal processing strategies that may preserve distance perception.
len, 2012 & 2013) along with improved analysis methods of confusion matrices are required. Two independent CV identification experiments in masking noise were conducted at various signal-to-noise ratios (SNRs), on 16 HI ears. In experiment 1 the CVs were presented with a flat gain, while in experiment 2 an ear-specific spectral compensation (i.e. NAL-R) was provided. All signals were presented at the most comfortable level (MCL). Based on our results: 1) A more rigorous definition of MCL audibility, based on entropy of the confusion groups is proposed, and 2) The effectiveness of NAL-R for CV perception was investigated by comparing the confusion matrices of the two experiments. A normed inner product vector space is defined, in which each row of a confusion matrix is represented as a probability vector. The Hellinger vector inner product is a distance metric that allows an in-depth comparison of response vectors that quantify individual confusions.

**Results:** The error and entropy decreased with NAL-R. The average error decreased from 20.1% (σ = 3.7%) to 16.3% (σ = 2.8%). By using the distance metric it was also found that for the NAL-R condition, the HI ears became more consistent (i.e. the distance between the responses for a token decreased). For 15.1% of the token-ear pairs (TEP), the entropy and error increased with NAL-R. These TEPs are effectively categorized by the use of the introduced distance metric.

**C24**

**Evaluation of new asymmetric directional microphone algorithm with automatic mode-switching ability**

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For hearing-support devices, it is important to reduce noises for speech recognition as well as to provide natural ambient sounds because ambient sounds provide information cues about the ambient environment. General spatial noise reduction, such as beamformer, reduces a dominant noise (DN) of null-direction. But it cannot provide natural ambient sounds. Asymmetric directional microphone (DM) fitting is possible to reduce noise and provide natural ambient sound. It is set to fixed DM in the left ear and omni-DM fitting in the right ear. It has partial benefit when the noise direction is left. When the noise direction is right, noise isn’t reduced. So in this study, a new binaural asymmetric DM algorithm that can overcome the defects of the conventional method was proposed. The proposed algorithm can estimate the position of a specific DN in the 90° – 270° range, and switch between directional- and omni-directional-mode devices automatically if the DM-mode device and the DN are placed in the opposite lateral hemisphere. Results of KEMAR recording tests (SNR, PESQ, HASQI) demonstrated that the performance of the conventional method was seriously deteriorated when the DM-mode device and the DN are placed in the opposite hemisphere; in contrast, the performance of the proposed algorithm was consistently maintained regardless of the directional variations in the DN. Results of quality test (Comparison Mean Opinion Score) for normal person showed that the performance of proposed algorithm was slightly improved. Because of normal hearing that is possible to have a superior hearing ability without between directional- and omni-directional-mode. Based on these experimental results, we expect that the proposed algorithm may be able to improve the speech quality and intelligibility of hearing-impaired persons who have similar degrees of hearing impairment in both ears. The algorithm will be evaluated with hearing-impaired persons in future work. [Work was supported by grants from Seoul R&B Program (SS100022).]

**C25**

**Modeling the neural representation of interaural level differences for linked and unlinked bilateral hearing aids**

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Interaural level difference (ILD) cues facilitate horizontal localization of sound frequencies above ~2 kHz and are processed by the lateral superior olive (LSO) of the brainstem. The LSO
is excited by sounds that are louder in the ipsilateral ear than in the contralateral ear. Bilateral hearing aids with independent wide dynamic range compression (WDRC) will tend to reduce the ILD cues. Hearing aids with wirelessly-linked WDRC algorithms are reported to have varying levels of success in restoring or maintaining an individual's ability to understand and localize sounds. In a target localization study with competing noise, hearing-impaired participants demonstrated improved localization ability with linked compression (Sockalingham et al., 2009). However, Wiggins and Seeber (2013) found that improved speech intelligibility in normal-hearing participants with linked WDRC was owed to an increase in SNR at the better ear, rather than preservation of ILD cues. In our study, we model the neural processing of the LSO in response to linked and unlinked WDRC schemes.

Two computer models are used to predict LSO responses. The first is a phenomenological model that takes inputs directly from ipsilateral and contralateral outputs of an auditory nerve (AN) model (Zilany et al., 2009, 2014). The LSO response is estimated from the difference in AN spike rates and a sigmoidal nonlinearity to account for the threshold and saturation in LSO spike rates. The second is a Hodgkin–Huxley-type biophysical model, which details the ion channel behavior of LSO cells. Inputs to this model are excitatory and inhibitory synaptic currents derived from the outputs of the ipsilateral and contralateral AN models, respectively. Both normal-hearing and hearing-impaired responses are generated. AUDIS (Blauert et al., 1998) and CIPIC (Algazi et al., 2001) head-related transfer functions are used to simulate sound sources ranging from -80° to +80° on the horizontal plane.

Preliminary results indicate that unlinked WDRC degrades the coding of sound source azimuth in the LSO by reducing ILDs. Wirelessly linked hearing aids maintain ILD cues and have the potential to enable the LSO response to indicate azimuth. However, with the standard gain prescriptions for WDRC, the output sound levels in both ears may not be sufficiently high for a robust representation of azimuth for moderately-loud sounds. These results will contribute to our understanding of the azimuthal perception of sounds at the neuronal level. This will have implications for the design of hearing aids to provide optimal spatial awareness to the listener.

[Supported by NSERC Discovery Grant 261736.]

C26

Monaural speech enhancement based on periodicity analysis and auditory excitation pattern subtraction

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An auditory-inspired algorithm for monaural speech enhancement in additive noisy conditions is described. The input noisy speech is decomposed into frames by a rectangular window with half-length overlap and each frame is decomposed into subbands by a phase-corrected complex-valued gammatone filterbank. The real part and the absolute value of the complex-valued filtered output are the bandpass signal and its Hilbert envelope, respectively. Periodicity of the bandpass signal in subbands with center frequency (CF) lower than 1 kHz and of the envelope signal in the remaining subbands is analyzed by means of two measures of periodicity, the normalized auto-correlation (NAC) and the comb filter ratio (CFR). The combination of NAC and CFR gives a reliable estimate of the fundamental frequency (F₀) and its salience (“periodicity degree”) in each voiced-speech frame-subband unit. Signal-to-noise ratio (SNR) can be estimated from the F₀ and the periodicity degree, and a revised Wiener gain is calculated based on the estimated SNR to be applied to the voiced-speech units. For unvoiced-speech units, the auditory excitation pattern (AEP, which is the energy of the subband filtered output) of the noise is estimated and subtracted from the AEP of the noisy speech. After the processing, the subband signal of each frame can be summed up directly to form the enhanced frame signal. The frames are multiplied with normalized Hamming window and added up with half-length overlap to form the enhanced signal. The enhanced signal is evaluated by two objective tests, segmental-SNR and perceptual evaluation of speech quality (PESQ), on the NOIZEUS corpus with 30 IEEE sentences in white noise, car noise (relatively steady), and
train noise (highly unsteady) at overall-SNRs of 0 dB, 5 dB, and 10 dB. The results show that the signal enhanced by the proposed algorithm has both large segmental-SNR and PESQ improvements in all test conditions compared to the original noisy signal. A state-of-art statistical model-based minimum mean square error (MMSE) algorithm is also evaluated. The result shows that the proposed algorithm has comparable performance as the MMSE algorithm in all test conditions. Informal listening test results are consistent with the above objective test results. [Work funded by DFG EXC1077.]

C27

Leveraging mobile technologies for large scale and distributed human studies of hearing
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Background: With the recent emphasis on outcomes-based research and functional performance, there is an increased need for technology that allows researchers and clinicians to reach a larger and more diverse subject population for recruitment and testing. While mobile computing offers a tremendous opportunity for the deployment of large scale human studies, systems with controlled acoustic output are still needed.

Overview: In collaboration with the Hearing Center of Excellence (HCE) and the Walter Reed National Military Medical Center, we have developed a flexible framework that enables researchers to easily design speech-in-noise tests and deploy them to tablets at multiple sites, in near-real-time. The system is designed for maximum flexibility for the researchers, while maintaining a simple interface for end users.

The overall system includes: a mobile application that resides on a tablet; a webserver application that is housed in the Cloud; and a test protocol that is generated by the researcher. The mobile application is implemented using standard Web technologies so that it can be easily deployed on a variety of platforms. The webserver application provides data management functionality, upload/download of new protocols and media, and soundcard-specific calibration of all acoustic media files so that the mobile application can control the output level of any presentation either in absolute form (e.g. dB SPL) or relative to a reference (e.g., dB or SNR). Researchers create and modify each test by editing a text template that enables a diverse set of functions including nested protocols and adaptive logic, randomized presentations, and timeouts.

Currently, this system is set to be used by the HCE in a study of functional hearing performance with 1500 military active subjects across multiple sites and as many as 50 Nexus tablets, each connected to a standard headset, using a single calibration for all tablets and headsets. The tablets have a remarkably stable output voltage across devices, and headsets produce tones with less than 5 dB of each other between 500Hz and 12kHz. Planned studies to norm new tests for functional hearing assessment will use the system with a new a wireless Bluetooth headset that includes circumaural noise attenuation and integrated calibrated sound production to enable audiometry-level measurements in a wide variety of settings. [This work is supported by the US Army Medical Research and Materiel Command, and the wireless headset development is supported by the National Institutes on Deafness and other Communication Disorders.]

C28

Optimizing benefit from combined electric and acoustic hearing
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The effectiveness of cochlear implants has increased markedly over the past years. As a consequence, candidacy for implantation has been relaxed and more CI users have functionally useful contralateral or ipsilateral acoustic hearing. Such users often report improved speech understanding, particularly in noisy situations, better tonal quality and – in case of contralateral acoustic hearing – improved lateralization of sound sources. The individual benefit from adding acoustic hearing, however, varies considerably. A potential reason for this is that usually acoustic
and electric stimulation are fitted and processed independently from each other.

Optimum aligned design and parametrization of the signal processing in the electric and acoustic path may facilitate the adaptation process and thus, help users to get the maximum benefit from combined electric and acoustic hearing. Of particular importance are: acoustic low-frequency audibility, spectral overlap of electric and acoustic stimulation, loudness balance, interaural tonotopy and (dynamic) synchronization of adaptive signal processing. In order to improve both effectiveness and efficiency of combined electro-acoustic stimulation, new approaches, which specifically address these challenges, are currently under investigation.

First implementations and evaluations of new tools for individual diagnosis, prescriptive fitting and individual fine-tuning indicate that ensuring optimum audibility and consistent dynamic signal processing can improve the benefit for many users. For instance, dynamic and static AGC-alignment can significantly improve speech understanding in noise, whereas differences in group delay barely have any effect. Besides fitting and signal processing, also spectral and temporal deficiencies of the patient affect the benefit from combined electric and acoustic hearing.

C29
Towards the development of a Listening in Spatialized Noise Tonal (LISN-T) test

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When communicating in complex acoustic environments the auditory system takes advantage of the temporal-spectral dynamics of the acoustic input at the two ears to analyse the spatial acoustic scene and thus, to understand speech. This ability is significantly reduced in hearing-impaired listeners, in particular in children with Spatial Processing Disorder (SPD). The Listening in Spatialized Noise-Sentences (LISN-T) test provides a clinical tool for assessing SPD, but requires proficiency in English.

Our goal is to develop a clinical test for assessing SPD that does not rely on speech as stimulus but assesses the same auditory mechanisms as the LISN-S. To approach this goal, auditory detection was measured for trains of 30ms-long tone-complexes with a long-term spectrum of speech and descending fundamental frequency. Thresholds were measured in three different distractor backgrounds: (1) two trains of tone-complexes that were similar to the target signal but with random fundamental frequency, (2) two trains of noise-bursts with temporal, spectral, and spatial properties similar to the tone-complexes distractor, and (3) two continues, incoherent noises with the same long-term spectrum as the two other distractors. To vary the spectral and temporal overlap between the individual target and distractor components, they were either synchronized or randomly jittered in time. The stimuli were spatialized with non-individualized HRTFs and presented via headphones. The target was always presented from the front and the distractors were either presented from the front or from ±90 degrees. The difference in thresholds between the two conditions defined the spatial release from masking (SRM). Fourteen normal-hearing subjects participated in the study.

When a temporal jitter was applied, the average SRM was 14.5 dB for the tone-complexes distractor and 5.6 dB for the noise-burst distractor. This large difference was mainly due to an increase in the collocated thresholds, which can be attributed to an increase in informational masking (IM). When signals were synchronized, the SRM was reduced to 6.3 dB and 0.8 dB, respectively, highlighting the effect of reducing spectro-temporal glimpses. For the continues-noise condition the SRM was 3.3 dB. The SRM as well as the individual collocated and spatially separated thresholds for the jittered condition with tone-complex distractors are very similar to the ones measured with the LISN-S. This may suggest that similar auditory mechanisms are assessed, making this condition a good candidate for a non-speech alternative to the LISN-S test for assessing SPD. However, further verification is required, in particular with children diagnosed with SPD.
Methods for evaluating spatial hearing in a clinical research environment

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Complex auditory scenes are frequently encountered in our everyday experience, and hearing aid users commonly report difficulties that suggest impaired spatial hearing abilities in these settings. However, reliable measures of spatial hearing in complex environments are difficult to achieve even in the controlled space of a research laboratory. In a clinical setting, audiologists have very few tools available to assess spatial hearing abilities or to try to optimize a hearing aid to improve auditory situational awareness. In order to explore possible solutions to this problem, our laboratory has constructed a Spatial Hearing Research Laboratory (SHRL) that consists of an array of 26 loudspeakers, each equipped with a multicolor LED display, mounted in a 300° arc that surrounds a listener seated inside a traditional double-walled clinical sound booth. This facility has been used to implement a series of tests of increasing complexity designed to rapidly assess spatial hearing performance in clinical patients. The baseline localization task in the facility is a single source localization task where the listener hears a brief broadband noise stimulus from a randomly selected speaker, and he or she uses an orientation-tracked “wand” to move a color-coded LED cursor to the perceived location of the sound. These LEDs are also used to provide feedback, where a green light indicates a correct response, while a red light indicates an incorrect response. The LEDs and wand also make it possible to add an interactive training task where a listener who is having difficulty discriminating between two speaker locations can explore the space by pointing and clicking at a given speaker to experience the resulting sound. This training has been especially beneficial in unilaterally deaf listeners learning to use a cochlear implant. The speaker array has also been used to develop a rapid localization-in-noise metric where listeners complete the single-source localization task in diffuse noise that adaptively increases or decreases in level until a listener reaches a specified criterion level of performance. It is also possible to implement more complicated spatial tasks in the SHRL. For example, one current task requires listeners to monitor multiple simultaneous voices in order to identify locations where a voice is added or removed from the scene. Experiments are also underway to compare free-field performance to headphone-based virtual auditory simulations in hearing impaired listeners. The results of these experiments will be used to develop more efficient measures of spatial hearing for use in clinical settings.

A cross-language evaluation of the Phoneme Perception Test

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A speech audiometry tool called the “Phoneme Perception Test” had been designed to measure the audibility, distinguishability and recognition performance of high frequency speech information. The test can be used in various listening situations, including those with hearing loss, with amplification, or with frequency lowering processing. It is comprised of recorded speech materials. The test was recorded using a German female speaker, but the sounds are not language specific and therefore should be able to differentiate abilities in various listening conditions regardless of the listener’s native language. To test this hypothesis and to identify possible differences, performance was measured in a group of German and a group of English speaking listeners. 10 English speaking subjects and 11 German speaking subjects were identified as matched pairs with similar pure tone audiograms. The results showed almost identical correlations between detection thresholds and the respective free field audiometric frequencies of both subject groups. With respect to recognition performance, the confusion data revealed no remarkable difference between the two groups in confusion patterns. The results are very similar between countries and underlines the applicability of the Pho-
Recent studies suggested that sensorineural hearing loss leads to an enhancement of temporal envelope coding arising from, e.g., broader auditory filters and loss of compression (Kale and Heinz, 2010; Henry et al., 2014). The present study tested the hypothesis of a relationship between enhanced envelope coding and the performance of hearing-impaired (HI) listeners in a pitch discrimination task, by investigating difference limens for fundamental frequency (F0DLs) for complex tones with components added in sine vs random phase. Previous studies found a larger benefit to F0 discrimination in some HI listeners than in a normal-hearing (NH) group, when components were presented in sine phase compared to random phase (Bernstein and Oxenham, 2006). While the difference in F0DLs for the two phase conditions has so far been mainly used as an additional estimate of harmonic resolvability, here the same difference was further investigated as a potential indicator of temporal envelope processing, on which pitch coding of unresolved complex tones is assumed to rely. F0DLs were measured as a function of F0 for complex tones filtered in a low (LF: 0.3-1.5 kHz) and high (HF: 1.5-3.5 kHz) frequency region, with components added either in sine or random phase. Consistent with earlier studies, the results for 14 NH listeners showed that F0DLs for HF complex tones decreased with increasing F0, and the transition point from large to small F0DLs served as an estimation of the transition from unresolved to resolved harmonics. While performance for the resolved complex tones was similar for the random- and sine-phase conditions, performance for the unresolved complex tones was, on average, 2% worse for components added in random vs sine phase. Results from the first three HI subjects showed as good as or even better performance than NH listeners for unresolved complex tones in the sine-phase condition. In the random-phase condition, one HI subject showed similar performance to NH listeners, while the other two subjects showed worse performance. Additionally, all HI subjects benefited more than NH listeners (by up to 10%) from sine-phase presentation, compared to random-phase presentation. These findings are qualitatively consistent with an effect of temporal envelope enhancement based on a broadening of auditory filters, which is expected to affect the envelope modulation depth at the output of the filters.

### Feedback path characterization for a direct acoustic cochlear implant

Giuliano Bernardi, Toon van Waterschoot, Marc Moonen, Jan Wouters, Jean-Marc Gérard, Joris Walraevens, Martin Hillbratt and Nicolas Verhaert

Acoustic feedback is a very common problem in hearing aids applications. Not only does it occur in traditional solutions, like behind-the-ear (BTE) or in-the-ear (ITE) aids, but it also affects bone-anchored hearing aids (Baha®) and more recent technologies, e.g. direct acoustic cochlear implants (DACI). For this reason, the design of a feedback canceller is a very important step when dealing with hearing aids. A characterization of the feedback path can make the feedback canceller design procedure easier, providing a rough idea of the maximum gain the hearing aid can supply without triggering instabilities or helping in the decision of the most suited feedback cancellation approach to adopt.
In this study, we present the data and analysis deriving from the feedback path characterization of a DACI, performed on fresh frozen cadaver heads. The Cochlear® Codacs system has been proven to be effective in the treatment of severe-to-profound mixed hearing loss; it consists of an implantable electro-mechanical actuator, placed within the mastoid cavity and firmly fixed through a bone plate, directly stimulating the cochlea via an artificial incus coupled to a piston prosthesis. The implanted actuator is controlled and driven from the outside through an RF link, and can be interfaced with the standard Cochlear BTE sound processor (originally developed for cochlear implants users).

To investigate the nature of the feedback we used exponential sine sweeps (ESS), allowing to assess the presence of nonlinear behaviors in the feedback paths. We used different stimulation levels to test for any level-dependent behavior. The recordings were made by means of two microphones, one incorporated in the BTE sound processor and one mechanically decoupled from the cadaver, called airborne (AB) microphone. The recording of these two different characterization signals could help to gain insights about the relations between the mechanical coupling and the induced acoustic feedback, since the BTE microphone is supposed to pick a much stronger mechanical component.

The analyzed data showed that the implant is characterized by a mechanical feedback component which is stronger and with a rather different frequency content compared to the AB one; additionally, the feedback seems to be dependent on the specific head morphology of the implanted specimen. Finally, the device did show some nonlinear behaviors; therefore, more measurements will be required to investigate the nature of such nonlinearities and be able to design a proper feedback canceller.

References:

C34
Hearing aid setting personalization using smartphones
Gabriel Aldaz and Larry Leifer
Stanford University

The smartphone has the potential to revolutionize the way users interact with their hearing aids, providing unprecedented personalization and increased satisfaction levels. We discuss the findings of a 6-week-long real-world study where 16 hearing impaired persons used a context-aware hearing system comprising two hearing aids, an Android smartphone, and a body-worn gateway to connect them wirelessly. Our objective was to determine whether participants could train their system, using a smartphone in the real world, to provide settings preferred over hearing aid automatic settings. We examined four modes: all combinations of microphone directionality (DIR/OMNI) and noise reduction (ON/OFF).

In the 4-week-long training phase, Awear randomly presented users with two types of listening evaluations. In the Self-Adjustment screen, participants toggled “Microphone Directionality” and “Noise Reduction” buttons until determining the best mode in that situation. Each Self-Adjustment result informed the SA Model, storing participant’s preferences according to sound environment, sound level, location, time of day, and other variables. In the A/B Test, the participant listened to both modes and gave a relative evaluation (“A is better,” “B is better,” or “No difference”). Each A/B Test informed the AB Model.

During the 2-week-long validation phase, Awear presented only A/B Tests, randomly assigning “A” to an adaptive model (AB or SA) and “B” to the hearing aid’s own fixed model (or vice versa). The chosen model received 1 point and the other model received -1 point, or both models received 0 if the user selected “No difference.” 11 of 14 participants* preferred settings predicted by the adaptive models they trained themselves (p = 0.02285). Encouragingly, degree of prefer-
ence for model-predicted settings positively correlated with (i) number of training phase evaluations taken and (ii) accuracy of evaluations.

We can understand why individualization is superior to a fixed algorithm by examining the proportion of times that each mode was chosen in a given sound environment. Pooling data across participants, the proportions average to be very similar. However, when we break down the data by participant, we see that individual preferences can be strong and idiosyncratic. This great variation in individual preferred mode per sound environment again supports our claim that personalized settings could provide a benefit not obtainable with an algorithm for everyone. In the future, our methods could be extended to vary frequency-gain curves, compression parameters, and other settings.

* One participant did not have sufficient training evaluations and another did not provide valid listening evaluation preferences.

### C35

**A public science experiment on hearing loss, tinnitus, and a lifetime of loud music exposure**

Michael A Akeroyd¹, William M Whitmer¹, Oliver Zobay², Deborah A Hall¹, Robert Mackinnon¹, Heather Fortnum³, David R. Moore⁴

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Over the 100 years since the founding of the Medical Research Council, there has been revolution after revolution in music reproduction technology, on stage and at home, with each revolution giving music that’s louder than the last. While it is well known that a lifetime of industrial noise exposure can lead to substantial hearing damage, the effect of a lifetime of loud music listening has been much less explored. To both celebrate the centenary of the MRC and assess the potential links between leisure music listening and hearing damage, we launched a public science experiment entitled “A century of amplified music.” The experiment was available for 18 weeks, from December 2013 to Easter 2014. It was programmed on a dedicated website, combining (1) a questionnaire that included questions on the amount of loud-music listening – concerts, clubs and personal-music players – over the lifespan as well as standard questions on hearing, and (2) a two-digit speech-in-noise test, measuring speech reception thresholds (SRTs), which was specifically designed for high-frequency hearing losses. The goal of this web-deployed experiment was to see how (1) related to (2).

In all 5300 respondents completed the test. We found a clear effect of age on SRTs (they increased by about 10 dB as age increased from about 40 to about 75), as well as effects of self-report hearing difficulty (up to about 5 dB) and exposure to noise at work (up to about 3 dB), but there were no effects of loud-music listening history on SRTs. Other analyses have, however, revealed small effects of loud-music listening history relating to self-reported hearing difficulty and tinnitus.

Our experiences in the design and implementation of this experiment, as well as suggestions for future public experiments, will also be discussed. [Work supported by the Medical Research Foundation, the Medical Research Council, and the Chief Scientist Office, Scotland.]

### C36

**Benefit of linear and nonlinear amplification with time-varying maskers**

Jayne B. Ahlstrom, William J. Bologna, Emily Franko-Tobin and Judy R. Dubno

Medical University of South Carolina, Department of Otolaryngology-Head and Neck Surgery

The purpose of this study was to determine the extent to which aided speech recognition in noise by older adults varies with the temporal characteristics of the masker, hearing-aid compression processing, and listeners’ cognitive abilities. Hearing aids with fast-acting, wide-dynamic range compression (WDRC) act rapidly over periods of time comparable to rapid fluctuations in speech, which may maximize effective moment-
to-moment speech audibility but may also distort the speech temporal envelope. This distortion may be particularly detrimental for older adults with reduced cognitive function. Given that speech recognition relies on precise timing for accurate coding of temporal information in speech, the benefits of compression amplification with rapid time constants may be limited for some older adults, especially for speech and maskers with abrupt temporal fluctuations. In this study, speech recognition in maskers that varied in their temporal characteristics was measured for older adults unaided, and then fit with hearing aids set with linear processing and output limiting compression, or with fast-acting WDRC. In this crossover design, a field trial was conducted to provide an acclimatization period, which was followed by laboratory measures; subsequently, subjects received the alternate hearing-aid fitting with a second field trial and second set of laboratory measures. 

IEEE sentences were presented at 0° azimuth at 65 dB SPL. The three maskers were each presented at ±60° at levels corresponding to -3, 0, and +3 dB signal-to-noise ratios. Maskers were a speech-shaped, steady-state noise, the same noise modulated by a 10-Hz square wave, and the same noise modulated by the temporal envelope of a four-talker babble. In the highly fluctuating square-wave modulated masker only, significant improvements in speech recognition were observed for aided vs. unaided listening, as compared to performance in steady-state maskers, and this benefit was equivalent for linear and WDRC processing. Moreover, hearing aids reduced listening effort during speech recognition tasks as assessed with the NASA Task Load Index, and significantly reduced perceived difficulty with speech communication during field trials as assessed with the Abbreviated Profile of Hearing Aid Benefit. In contrast to previous findings, working memory and processing speed, as assessed with Reading Span, Connections, and Purdue Peg Board, were not significant predictors of aided speech recognition, effects of compression processing, or masker temporal characteristics. [Work supported by NIH/NIDCD]

**C37**

**A linear relationship exists between HTLs and UCLs in individuals, unrelated to ear side, or frequencies**

Shinobu Adachi1, Myungmi Oh2, Takechiyo Yamada2, Tsunaki Saito3, Jun Ozawa1

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Hearing threshold level (HTL) and uncomfortable loudness level (UCL) are basic properties of hearing that are used in adjusting the dynamic range of sound pressure generated by hearing aids. Previous studies have reported that UCLs vary widely among individuals (by 50 dB or more), even in people with identical HTLs for each ear and frequency. However, little is known about the relationship between HTLs and UCLs in individual participants. We analyzed the distribution of HTLs and UCLs for each participant to clarify the relationship between them.

Adults with moderate hearing loss participated in this study (40 men and 32 women, mean age 73.4 years). Continuous tones of 250, 500, 1k, 2k, and 4k Hz were presented in an ascending order of loudness to each ear by using an SPL audiometer (D2-36H, DANA JAPAN). To measure HTL, participants were asked to raise their hands when a sound was heard. The sound level was recorded as the HTL for that ear and frequency. To measure UCL, participants were asked to raise their hands when the sound was too loud to listen for an extended period and the sound level was recorded similarly to HTL. HTLs were also measured using an HL audiometer (AA-98, RION) to confirm the average pure-tone measurements. The least-squares method was used to calculate a regression line for individual participants from their HTLs and UCLs for each ear and for each frequency. HTL or UCL measurements exceeding 120 dB SPL were excluded from the analysis.

The mean ± standard deviation of the pure tone average was 54.9 dB HL ±13.9 dB. The mean ± standard deviations of UCLs for each frequency (250, 500, 1k, 2k and 4k Hz) were 102.5 dB SPL ±12.3 dB, 98.5 dB SPL ±12.5 dB, 95.5 dB SPL
±12.1 dB, 99.0 dB SPL ±11.3 dB and 100.1 dB SPL ±11.4 dB, respectively. The maximum individual difference of UCLs in the same HTLs was 55 dB.

The mean residual error between the regression lines that were calculated and the measured UCLs for all participants was 3.2 dB. In 94% of participants, the mean residual errors were less than 5 dB. Interestingly, the linear relationship was true for participants with HTLs in one ear that were widely different from those of the other ear.

These results suggest that there is a linear relationship between HTLs and UCLs in individuals with moderate hearing loss.

### C38

**Predicting speech information transmission from the audiogram and vice versa**

*Peter J Blamey¹,²,³,⁴, Jeremy K Blamey⁵, Daniel Taft² and Elaine Saunders²*

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**Aims:** Information transmission analysis of vowel and consonant confusions from a monosyllabic word test (Speech Perception Test, SPT) can be used as an alternative to the audiogram for measurement of hearing loss, for hearing aid fitting, and for counselling people who have hearing difficulties about listening and communication strategies and tactics. The goal of this study was to investigate methods of converting between audiograms and SPT results and demonstrate their equivalence or otherwise.

**Methods:** The data used in this study included 6079 SPT results, 2534 audiograms, and 408 cases where both SPT and audiogram were available. Information transmission analysis was applied to the SPT results to produce an Infogram™. Principal Components Analysis and multiple regression were used to develop formulae to convert from the Infogram™ to the audiogram and vice versa.

**Results:** There was a high correlation between the conventional audiograms and the “speech audiograms” derived from the Infogram™ at every frequency (p < 0.001). Similarly, there was a high correlation between the measured information transmitted and the information transmitted estimated from the audiogram for every speech feature (p < 0.001). Although the “speech audiograms” followed the same general shape as the conventional audiograms, they tended to be smoother and have shallower slope because of the broadband nature of speech sounds.

**Conclusions:** The relationships and equivalence between word test results and audiograms will enable measurement of hearing and hearing aid fitting without specialised equipment or expertise. This is of particular relevance in regions with limited availability of audiologists, audiometers, sound booths, etc. It also improves self-fitting of hearing aids.

### C39

**Using acoustic phonetics to reduce the financial and social burdens of hearing loss for individuals**

*Peter J Blamey¹,²,³,⁴, Jeremy K Blamey², Daniel Taft² and Elaine Saunders²*

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**Introduction:** This research program is designed to improve the standard of hearing health care, to reduce costs in order to encourage more people to become hearing aid users, and increase the effectiveness of hearing aid fitting for people who already use hearing aids.

**Aims:** The specific goal of these studies was to investigate the efficacy of a novel tool for the measurement of hearing, hearing aid fitting, fine tuning, and validation. The tool is an online speech perception test (SPT) together with an
acoustic phonetic analysis that can be used to generate initial and improved hearing aid fittings and evaluate the benefit of hearing aids.

Methods: The SPT is a monosyllabic word test with 50 items that generates a display of information transmission for ten vowel and consonant features that can be used to characterise the shape and degree of hearing loss, analogous to an audiogram. A pilot study was performed with 88 people to assess the sensitivity and specificity of the SPT. Correlation analysis was performed for conventional audiograms and “speech audiograms” derived from the information transmitted for 408 people. Hearing aid fittings were derived from unaided SPT results and aided SPTs were used to measure the perceptual benefit of the hearing aids, and fine tune the fittings.

Results: The SPT had high sensitivity (94%) and high selectivity (98%) relative to other online and telephone hearing test methods. At every frequency, there was a highly significant correlation (p<0.001) between conventional audiograms and “speech audiograms”. Proof of concept for hearing aid fitting and fine tuning based on unaided SPT results and aided SPTs was achieved and illustrated with specific cases. These are in-situ measures and do not require corrections for factors such as venting, RECD, microphone position etc. SPT results and hearing aid test box speech maps indicated that actual and predicted performance were at least as good for the SPT-based fittings as for audiogram-based fittings.

Conclusions: The high correlation between SPT results and audiograms demonstrated the feasibility of this approach to hearing aid fitting. Pilot test results and preliminary fitting data indicate that the SPT and the acoustic phonetic analysis are effective tools for the measurement of hearing, hearing aid fitting, fine tuning, and validation.

Significance: When combined with an effective tele-audiology business model, the use of speech perception results and acoustic phonetic measures that are easily understood by the general public have the potential to reduce the costs of hearing aid provision and increase the effectiveness of hearing aid fitting, thereby reducing the financial and social burdens of hearing loss for individuals.
In summary, the proposed Asynchronous Warped Filterbank allows custom design for computational load, spectral coverage and resolution issues while maintaining all of the attractive properties of synchronously sampled Warped filterbanks.

Reference:
### IHCON 2014 Attendees

![Attendees](image)

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