IHCON 2012

International Hearing Aid Research Conference

August 8 – 12, 2012

Granlibakken Conference Center
Lake Tahoe, California
IHCON 2012

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Sigfrid Soli

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Michael Akeroyd, Technical Co-Chair (2010)
MRC Institute of Hearing Research, UK

Ben Hornsby, Technical Co-Chair (2010)
Vanderbilt University, USA
# Student Scholarship Recipients

<table>
<thead>
<tr>
<th>Name</th>
<th>Institution</th>
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<tbody>
<tr>
<td>Elin Barrock(^1)</td>
<td>Lund University</td>
</tr>
<tr>
<td>Alan Boyd(^1,2)</td>
<td>MRC Institute of Hearing Research &amp; University of Strathclyde</td>
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<tr>
<td>Naomi Croghan(^1)</td>
<td>University of Colorado at Boulder</td>
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<tr>
<td>Helen Glyde(^1)</td>
<td>University of Queensland</td>
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<tr>
<td>Sofie Jansen(^1)</td>
<td>ExpORL</td>
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<tr>
<td>Andrew Sabin(^1)</td>
<td>Northwestern University</td>
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<tr>
<td>Marina Salorio-Corbeto(^1,2)</td>
<td>University of Cambridge</td>
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<tr>
<td>Andrew Schwartz(^1)</td>
<td>Harvard-MIT</td>
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<tr>
<td>Jayaganesh Swaminathan(^1)</td>
<td>Massachusetts Institute of Technology</td>
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<tr>
<td>Eric Tarr(^1)</td>
<td>Ohio State University</td>
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<tr>
<td>Adam Westermann(^1)</td>
<td>National Acoustic Laboratories &amp; Macquarie University</td>
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<tr>
<td>Qun Wei(^1)</td>
<td>Kyungpook National University</td>
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<tr>
<td>Ian Wiggins(^1,2)</td>
<td>MRC Institute of Hearing Research</td>
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Student Scholarship funding is from the following sources:

\(^1\) NIDCD
\(^2\) Deafness Research UK
# Daily Schedule

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<tr>
<td>5:00 PM</td>
<td>Welcome Social</td>
<td>9:55 AM Poster Session</td>
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<tr>
<td>6:00 PM</td>
<td>Dinner</td>
<td>11:10 AM Morning Session B</td>
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<tr>
<td>7:30 PM</td>
<td>Welcome Remarks</td>
<td>12:40 PM Lunch</td>
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<tr>
<td>7:45 PM</td>
<td>Keynote Address</td>
<td>4:50 PM Evening Session</td>
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<tr>
<td>8:45 PM</td>
<td>Discussion</td>
<td>7:00 PM Dinner</td>
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<tr>
<td>9:00 PM</td>
<td>Social</td>
<td>8:30 PM Social</td>
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<table>
<thead>
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<th>Time</th>
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<tbody>
<tr>
<td>7:00 AM</td>
<td>Breakfast</td>
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<tr>
<td>8:00 AM</td>
<td>Morning Session A</td>
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<tr>
<td>9:40 AM</td>
<td>Poster Session</td>
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<td>11:10 AM</td>
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<td>12:20 PM</td>
<td>Lunch</td>
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<tr>
<td>7:00 AM</td>
<td>Breakfast</td>
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<tr>
<td>8:00 AM</td>
<td>Conference Concludes (check-out following breakfast)</td>
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Program Summary

Welcome and Keynote Address
7:30 PM – 8:45 PM

Welcome Remarks:  Sig Soli, Todd Rocketts

Keynote Address

Judy R. Dubno  Auditory training for aided listening by older adults
Thursday, August 9

Session One:
Pediatric Outcomes
8:00 AM – 9:40 AM
Moderator: Dawna Lewis

Mary Pat Moeller
(Keynote) Outcomes of children with mild to severe hearing loss: Insights and challenges

Andrea Pittman
Word learning in children with hearing loss using digital noise reduction

Susan Nittrouer
Combining hearing aids with cochlear implants in children: Results and implications for treatment

Poster Session I 9:40 AM – 11:00 AM

Session Two:
Large Scale Studies
11:10 AM – 12:20 PM
Moderator: Kathryn H. Arehart

Frank R. Lin
(Keynote) Hearing loss and healthy aging

James Miller
Multi-site study of the efficacy of speech perception training for hearing aid users: Phase I
Session Three:
Clinically Applicable Research—The Long and the Short View
5:00 PM – 7:00 PM
Moderator: Todd Ricketts

Sigfrid Soli (Keynote) Early hearing aid intervention: An international perspective
Francis Kuk A simple way to estimate aided in-situ audibility
Erin Picou Potential benefits of a bilateral beamformer for hearing aid users in realistic listening situations
Brian Moore Comparison of the CAM2 and NAL-NL2 hearing aid fitting methods
Benjamin Hornsby Tired of listening: Subjective and objective measures of hearing related fatigue
Odile Clavier Hearing fitness for duty for military personnel
Severin Fuerhapter First European results with a new transcutaneous bone conduction hearing implant
Friday, August 10

Session Four: 
Individual Differences in Patients 
8:00 AM – 9:55 AM
Moderator: Benjamin Hornsby

Catherine Palmer (Keynote) Individual differences: Impact in the lab and clinic
Larry Humes Predicting individual differences in aided speech understanding in older adults
Gabriel Saunders Hearing health benefits
Sofie Jansen Hearing screening for noise-induced hearing loss based on speech intelligibility in background sound

Poster Session II  9:55 AM – 11:10 AM

Session Five: 
Frequency Transforming Technologies 
11:10 AM – 12:40 PM
Moderator: Mary Cord

Susan Scollie (Keynote) Frequency lowering technologies—benefits and limitations
Kelly Fitz Additive synthesis of frequency-lowered consonants
Joshua Alexander Inverse frequency compression for precipitous hearing loss
# Session Six:
## Teleaudiology

4:50 PM – 6:55 PM  

**Moderator: Earl Johnson**

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<tr>
<td>Stefan Launer</td>
<td>Telemedicine and teleaudiology: A new approach in hearing health care?</td>
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<tr>
<td>Harvey Abrams</td>
<td>A survey of attitudes toward teleaudiology</td>
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<td>Gurjit Singh</td>
<td>Attitudes toward teleaudiology: A comparison of pediatric and non-pediatric hearing healthcare practitioners</td>
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<td>Thomas Lunner</td>
<td>Professional online rehabilitation of adult hearing-aid users—a randomized controlled trial</td>
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<tr>
<td>Gitte Keidser</td>
<td>A truly self-fitting hearing aid: Feasibility and challenges</td>
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Saturday, August 11

Session Seven:
EAS/Implantable Options
8:00 AM – 10:10 AM
Moderator: Ian Bruce

René Gifford (Keynote) Combined electric and acoustic stimulation (EAS): Hearing preservation cochlear implantation

Iris Arweiler Benefit of advanced directional microphone for speech understanding in cochlear implant users

Stefan Mauger Noise reduction optimized for cochlear implants

Peter Blamey Cochlear implant results show efficacy of hearing aids in preserving auditory function

Poster Session III 10:10 AM – 11:25 AM

Session Eight:
Stuart Gatehouse Lecture
11:30 AM – 12:30 PM
Moderator: Graham Naylor

Michael Akeroyd (Keynote) The localization of sound, without and with hearing aids
Session Nine:
Binaural Issues
4:45 PM – 7:00 PM
Moderator: Stefan Launer

Ian Bruce
Physiological prediction of masking release for normal-hearing and hearing-impaired listeners

Jing Xia
Effects of bilateral compression on spatial unmasking in hearing-impaired listeners

William Whitmer
The perception of free-field source width in hearing-impaired individuals

Suzanne Carr Levy
Effects of bandwidth and microphone location on hearing and hearing-impaired listeners

Andrew Schwartz
Binaural compression weakens spatial unmasking: The compression-blur hypothesis

Sunday, August 12

CLOSE OF CONFERENCE

Check-out following breakfast
Keynote Address

Auditory training for aided listening by older adults

Judy R. Dubno
Department of Otolaryngology-Head and Neck Surgery
Medical University of South Carolina, Charleston, South Carolina 29425

Speech recognition difficulty is a common complaint of older adults with hearing loss. Whereas most hearing aids increase speech audibility in relatively quiet environments, they may not improve the signal-to-noise ratio (SNR) in noisy environments. When technology provides a less-than-optimal SNR, a complementary approach is to improve the SNR by training the listener. That is, as a supplement to aided listening in noise, an effective auditory-training program may improve the use of audible speech information, enhance communication abilities, and increase the efficacy of hearing aids. Although promising results were reported in the 1970s and 1980s, interest in and evidence supporting auditory training for adults has been limited, perhaps due to technical restrictions of those programs and the need for laboratory-based training extending over many days and weeks. More recent training programs developed to utilize newer technology and automated, home-based training generated renewed awareness and additional evidence of benefits of auditory training for older adults who use hearing aids. This presentation will review key features and recent results of auditory-training paradigms, including those that (1) focus on individual sounds, commonly used, meaningful words in isolation, or words in phrases or sentences; (2) use auditory cues alone or auditory plus visual/orthographic cues with feedback; (3) emphasize the role of contextual information, cognitive function, and comprehension; (4) include multiple talkers and speech-like background noise to simulate realistic listening environments; and (5) assure audibility using spectrally shaped speech or participants’ own hearing aids. [Work supported by NIH/NIDCD and the Hearing Health Foundation.]
Thursday, August 9

SESSION ONE
Pediatric Outcomes
Moderator: Dawna Lewis

8:00 AM Outcomes of children with mild to severe hearing loss: Insights and challenges
Mary Pat Moeller, Ph.D.
Boys Town National Research Hospital

There is a relatively limited body of research related to the communicative outcomes of children with mild to severe hearing loss who wear hearing aids. Previous research involving this population suggests that mild to severe hearing loss places children at risk for delays in speech, language, academic and psycho-social development. However, most of these studies were conducted prior to the implementation of early identification programs and intervention with advanced hearing technologies. It is expected that recent practice innovations will result in improved outcomes for this group of children, but the requisite evidence to support this contention has not been collected on a large scale. Such evidence would provide important insights related to the influence of amplified hearing on key language and academic outcomes. These data may also provide insights about device effectiveness and individual differences in outcomes.

Our research team has long been interested in exploring how children’s behavioral outcomes might inform our understanding of device effectiveness. Through multidisciplinary research collaborations, we have examined aspects of language learning that are predicted to be challenging in the context of hearing loss (e.g., morphological development, learning novel words through overhearing, social cognition). We also have been exploring the influence of environmental factors (e.g., quantity of language exposure in the home, consistency of device use) as a way of understanding individual differences in outcomes. Results of these studies will be discussed, along with implications for clinical practice and future research.

The presentation also will include a discussion of recent results from a large, NIDCD-funded project called the Outcomes of Children with Hearing Loss (OCHL). This longitudinal research project a collaborative effort of the University of Iowa, Boys Town National Research Hospital and the University of North Carolina-Chapel Hill. To date, this interdisciplinary team has recruited children and families from seventeen states. A total of 316 children with bilateral mild to severe hearing loss, and 115 SES-matched children with normal hearing are currently being followed in the study. Key findings from the first three years of data collection will be shared. As an example, nearly 200 children in this project were identified through newborn hearing screening programs, yet only 66% received timely hearing aid fittings. Maternal education was the only significant predictor of timeliness of follow-up. Children of mothers with high school education or less were much later to be fit with amplification than those with more educated mothers. Results related to the appropriateness of pediatric hearing aid fittings will be discussed, along with findings related to the influence of audibility on child language outcomes. New insights about consistency of hearing aid use in young children will be described. Relative areas of strength and weakness in speech, language and academic development will be reviewed with implications for future research. Finally, children’s performance on theory
of mind tasks will be shared. Possible mechanisms for social cognitive delays will be discussed, along with considerations for providing enhanced auditory access.

8:40 AM  
**Word learning in children with hearing loss using digital noise reduction**

*Andrea Pittman*  
*Arizona State University*

This study examined the impact of noise on word learning in children with normal hearing (NH) and children with hearing loss (HL). Additionally, the effects of digital noise reduction (DNR) were examined. It was hypothesized that, compared to children with NH, word learning would proceed more slowly for children with HL in quiet and in noise. Consistent with previous research with DNR showing no benefit or detriment to children, DNR was not expected to deter word learning. Participants were 41 children with NH and 26 children with mild to moderately severe HL. Half of the children in each group were 8-9 years of age and the other half were 11-12 years of age. The children with HL were fitted with commercially available BTE hearing aids appropriate for this population. The aids were programmed with two identical memories except that DNR was disabled in one memory and enabled in the other.

Stimuli were 3 sets of 5 nonsense words constructed with the same phonemes but different vowels. The stimuli and steady-state noise were presented in the sound field at 50 dB SPL for a nominal SNR of 0 dB. Effective SNR was measured to be approximately 2.3 dB higher with DNR enabled.

The children’s task was to learn the words by matching each to a unique picture through a process of trial-and-error. One-hundred trials were presented. The trial-by-trial data were reduced to ten data points by calculating performance in bins of 10 trials each. The data were fit with an exponential growth function such that the number of trials corresponding to 70% correct performance could be determined. The children with NH learned words in quiet and in noise whereas the children with HL learned words in quiet, in noise, and in noise with DNR enabled.

Results showed that the older children with NH learned the words faster (required fewer trials) than the younger children with no differences between the quiet and noise conditions. No age effect was observed for the children with HL, however the children learned significantly slower (required more trials) in noise with DNR disabled. When DNR was enabled, word learning in the younger children was unchanged whereas word learning in the older children improved significantly to a level consistent with learning in quiet.

The results of this study suggest that, in addition to improving listening comfort, DNR may offer unique benefits to children with HL. [Note: This study was funded by a generous grant from the ASHA Foundation and presented as a poster at the ASHA convention in November 2011. It was published in the October 2011 issue of JSLHR.]

9:10 AM  
**Combining hearing aids with cochlear implants in children: Results and implications for treatment**

*Susan Nittrouer, Amanda Caldwell, and D. Bradley Welling*  
*Otolaryngology, The Ohio State University*

In a 1988 book chapter, Boothroyd reminds us that most deaf children can benefit from hearing aids (HAs). He also cautions that it is difficult to gather adequate information from infants regarding the amount of residual hearing they have, in order to know who
stands to benefit. With the proliferating popularity of cochlear implants (CIs), this perspective concerning the value of HAs for deaf children has largely been forgotten. Here the role of HAs in the language development of deaf children is re-examined.

In this two-part report, outcomes are first presented for 90 second grade children who have participated in a longitudinal study since they were 12 months old. Thirty children have normal hearing (NH), 20 children have moderate losses (mean better-ear PTA = 70 dB HL) and wear HAs, and 40 have severe to profound losses (pre-implant better ear PTAs > 80 dB HL) and wear CIs. Half the children with CIs spent at least one year wearing a HA in the ear contralateral to the ear with the CI (i.e., bimodal experience); half did not. Results from 12 standard or experimental measures of language ability are reported: three of phonological awareness, two of productive syntax and two of reading, as well as one each of expressive vocabulary, auditory comprehension, narrative skills, speech intelligibility, and working memory.

In general, results show that children with NH performed best and children with CIs performed poorest. Children with HAs performed intermediately. However, children with CIs who had histories of some bimodal experience performed better than children without that experience on many measures. Group differences were greatest for measures that require refined sensitivity to linguistic structure (phonological or syntactic), and smallest for tasks that can be accomplished with the help of real-world knowledge (such as auditory comprehension and reading comprehension).

In the second part of the report, potential contributions of low-frequency acoustic hearing to perceptual and language development of children with severe to profound hearing loss are discussed. Understanding these potential roles can help shape how we prescribe and fit HAs to best supplement CIs.

The conclusion is that Boothroyd's exhortations of a quarter century ago should be embraced today. Even though CIs are available, deaf children can benefit from the low-frequency signal provided through HAs. Understanding the roles played by this signal as an adjunct to CI input suggests ways that prescription and fitting of HAs can be optimized. [This work was supported by Grant No. R01 DC006237 from the National Institute on Deafness and Other Communication Disorders, the National Institutes of Health.]

Reference:
Thursday, August 9

SESSION TWO
Large Scale Studies
Moderator: Kathryn H. Arehart

11:10 AM  Hearing loss and healthy aging

Frank R. Lin
Assistant Professor of Otolaryngology and Epidemiology, Johns Hopkins University
School of Medicine and Bloomberg School of Public Health
Core Faculty, Johns Hopkins Center on Aging and Health

Age-related hearing loss (ARHL) in older adults is often perceived as being an inconsequential part of aging. This observation is borne out by the epidemiologic data demonstrating that <15% of adults with a clinically-significant hearing loss use hearing aids and by current funding levels for biomedical research on hearing compared to other sensory modalities. However, the broader consequences of hearing loss on the health and functioning of older adults are now beginning to surface in epidemiologic studies. I will discuss recent epidemiologic research demonstrating that hearing loss is independently associated with poorer cognitive functioning, accelerated cognitive decline, an increased risk of incident dementia, and accelerated rates of brain atrophy as measured by MRI. Results from analyses of several large epidemiologic datasets including the National Health and Nutritional Examination Surveys, the Health Aging and Body Composition Study, and the Baltimore Longitudinal Study of Aging will be presented. Finally, I will discuss ongoing and planned studies to investigate the impact of current hearing rehabilitative interventions on delaying cognitive decline and dementia in older adults.

11:50 AM  Multi-site study of the efficacy of speech perception training for hearing-aid users: Phase I

James D. Miller¹, Charles S. Watson¹, Gary R. Kidd¹, Judy R. Dubno² and Marjorie R. Leek³

¹ Communication Disorders Technology (CDT), Inc., Bloomington, Indiana, US
² Medical University of South Carolina, Charleston, South Carolina, US
³ National Center for Auditory Rehabilitation Research, Portland, Oregon, US

The Speech Perception Assessment and Training System for the Hearing Impaired (SPATS-HI) developed at CDT, Inc., and previous results of its use by hearing-aid and cochlear-implant users will be reviewed. Then Phase I of an ongoing “multi-site study of speech perception training for hearing aid users” will be described. The goal of this research program is to determine whether speech-perception training can improve aided speech perception for typical presbycusis hearing-aid users. Phase I of the program was to establish whether thirty hours of intensive, computerized speech-perception training: (1) is accepted by the target population and (2) leads to improvements of their aided speech perception. A third goal of Phase I was to explore cognitive and auditory measures that might be correlated with success with aided speech perception and benefit from training. Two groups of 12 adult hearing-aid users were given 30 hours of speech-perception training, one at the Portland Veterans Administration Medical Center and the other at the Medical University of South Carolina. Training was done using SPATS-HI wherein listeners are given intensive drills on the recognition of the syllable constituents of English:
onsets, nuclei, and codas. Syllable constituents are spoken by eight talkers, in varying phonetic contexts. The system is adaptive, concentrating training on those constituents of intermediate difficulty for each listener. Listeners are also trained to identify words in naturally produced, meaningful sentences spoken by 12 talkers and presented in multitalker babble at signal-to-babble ratios of 15, 10, 5, 0, and -5 dB. In addition to cognitive and auditory measures, criterion measures of speech recognition were obtained before and after training and the same measures were obtained from a group of non-trained controls. All trained participants completed the 30 hours of training and the results of a questionnaire indicated that they viewed the experience favorably. The average of the post-training criterion measures improved for individual trained listeners as compared to the control listeners, and the average trained listener out-performed the average control listener on each of the criterion measures. Some listeners had only a little measured benefit from the training, while others demonstrated much more benefit. Possible variables related to success will be discussed. More extensive research at six sites is continuing which may result in improved training regimes, more sensitive measures of benefit, and the prediction of individual differences in benefits. (Authors Watson and Miller are stockholders in CDT, Inc.)

Thursday, August 9

SESSION THREE

Clinically Applicable Research—the Long and the Short View

Moderator: Todd Ricketts

5:00 PM  Early hearing aid intervention: An international perspective

Sigfrid D. Soli\textsuperscript{1} and Yun Zheng\textsuperscript{2}

\textsuperscript{1} House Research Institute, Los Angeles, USA
\textsuperscript{2} West China Hospital of Sichuan University, Chengdu, PRC

Early Hearing Detection and Intervention (EHDI) are important for effective treatment of adults and children with hearing impairment. Pediatric EHDI programs seek to provide appropriate auditory input during the critical period in early childhood when speech and language normally develop. This input can enable the hearing impaired child to partake in this developmental process. These considerations underscore the importance of objective evidence of speech and language development following intervention to validate the success of the intervention (AAA, 2003). Early pediatric intervention strategies begin with hearing aid amplification. However, behavioral validation can be difficult in very young children who have been fit with hearing aids.

This research addresses these issues through the application of a Mandarin hierarchical outcome assessment battery to evaluate outcomes following early hearing aid fittings. The Mandarin battery is similar to an English battery widely used to evaluate early outcomes following pediatric cochlear implantation (Eisenberg \textit{et al.}, 2006). The Mandarin battery consists of measures of early prelingual auditory development obtained from a structured interview with the parents (Zheng \textit{et al.}, 2009), early closed-set word recognition (Zheng \textit{et al.}, 2009), early closed-set sentence recognition with competing back-
ground interference (Zheng et al., 2009), and a Simplified Short Form version of the Mandarin Communicative Development Inventory (Soli et al., in press).

A sample of 120 hearing impaired children with moderate-profound sensorineural hearing losses fitted with hearing aids between 1 and 5 years of age are currently participating in a 4-year longitudinal outcome study. This presentation will report longitudinal outcome data collected during the first 24 months after fitting. Since a primary goal of pediatric EHDI programs is to enable speech and language development like that of normally hearing children, outcome data for the hearing impaired subjects are compared with normal developmental trajectories previously established for each of the outcome measures. These comparisons are made by expressing the scores for hearing impaired subjects as normal equivalent ages, i.e., the chronological age at which a normally hearing child would achieve the same score.

Results for the hearing aid sample during the first 24 months after intervention will be compared with results for normally hearing children as well as with results for pediatric cochlear implant recipients in China and in the US. These data and comparisons can provide objective validation for pediatric hearing aid interventions, and help to establish best practices using principles of evidence-based medicine. [Sponsored by Widex A/S, Denmark, Cochlear Medical Device Co., Ltd., PRC, House Research Institute, USA, and West China Hospital of Sichuan University, PRC.]

5:30 PM

A simple way to estimate aided in-situ audibility

Francis Kuk
Widex ORCA-USA

It has often been reported that less than 30% of clinicians use real-ear measures to verify the aided real-ear output of a hearing aid. This author believes the use of in-situ audiometry, coupled with a display of simulated real-ear hearing aid output could be a simple way to verify the aided audibility across frequencies provided by a hearing aid. This presentation provides a rationale for and the validation data to support such a hypothesis.

In-situ audiometry is the measurement of hearing thresholds with the patients’ hearing aids in their ears to account for all the individual ear characteristics. Because in-situ thresholds and hearing aid output can both be calibrated on the same coupler, the hearing aid output that is above the in-situ threshold (which is expressed as the dB sensation level, or dB SL) should represent the output that is audible to the wearer. This should be true regardless of coupler (2 cc, simulated real-ear, or real individual ear) as long as the same coupler is used for both in-situ threshold calibration and hearing aid output calibration. An implication of this logic is that one can predict the real-ear audibility of the hearing aid output simply by comparing the coupler hearing aid output to the coupler in-situ thresholds without actual real-ear measurements.

This possibility was evaluated by comparing the sensation level (dB SL) of the amplified sounds (i.e., hearing aid output above the in-situ threshold of the wearers) measured between using the simulated real-ear display on the Widex fitting software (SoundTracker) and two real-ear measurement systems (Audioscan Verifit and Frye 6500) at two sites with over 20 subjects. For both REM systems, the agreement in SL between the SoundTracker and the REM was within 1-2 dB on average. This suggests that if the clinician measures in-situ thresholds, the SL of the amplified sounds can be reliably and accurately displayed in the manufacturer’s fitting software without additional equipment or effort. This could make verification simpler and encourage more frequent verification effort. However, it must be stated that the use of the SoundTracker provides only one piece of
Potential benefits of a bilateral beamformer for hearing aid users in realistic listening situations

*E.M. Picou, T.A. Ricketts and E. Aspell*
*Department of Hearing and Speech Sciences, Vanderbilt University Medical Center*

For hearing aid users, understanding speech in noise remains problematic, even with current strategies aimed at reducing noise levels (e.g., directional microphones, digital noise reduction algorithms). With more recent advances in hearing aid technology, it is now possible to share information and/or signals between hearing aids. One possible use for this technology is a bilateral beamformer, which combines the signal output from all four microphones across the two hearing aids in a single second or third order microphone array. While bilateral beamformers have the potential to enhance SNR compared to traditional directional microphone hearing aids, the binaural cues that listeners use for spatialization may be eliminated because the microphone array outputs are diotic. The potential consequences of reduced spatialization are unclear, but could include poor localization, feelings of disconnect with the environment, or even reduced speech understanding. To possibly take advantage of the bilateral beamformer technologies while maintaining some spatial cues, at least one hearing aid manufacturer has implemented a hybrid bilateral beamformer. That is, the bilateral beamformer is active for higher frequencies, while standard directional technology is active for lower frequencies. The effects of this beamformer technology compared to traditional directional and omnidirectional hearing aids for speech recognition in noise, listening effort, realistic localization, and spatial release from masking were investigated in 20 hearing aid users. Participants were tested in background noise in reverberant and non-reverberant environments using commercially available hearing aids. Participants were also asked to report their subjective impressions of the directional and bilateral beamformer microphones after experiencing live listening trials, including conversing inside and outside a medical center. Results indicated that, compared to the omnidirectional mode, the hybrid bilateral beamformer improved speech recognition in noise (up to 25 percentage points in some conditions), decreased listening effort, and improved recall. A smaller, but significant speech recognition advantage of up to 9 percentage points was also found in comparison to a traditional, full-band directional mode in the reverberant environment. Conversely, the technology has some specific limitations, primarily reduced localization performance in a realistic localization task. Despite this limitation, however, most listeners expressed a preference for the bilateral beamformer over a traditional adaptive directional microphone.

Comparison of the CAM2 and NAL-NL2 hearing-aid fitting methods

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We compared preference judgments for sounds processed via a simulated five-channel compression hearing aid with gains and compression ratios selected according to two recently introduced fitting methods, CAM2 and NAL-NL2. For high frequencies, CAM recommends higher gains than NAL-NL2. The fifteen participants had mild sloping sensorineural hearing loss, typical of people who might be candidates for wide-bandwidth
hearing aids. Within a given trial, the same segment of sound was presented twice in succession to one ear, once with CAM2 settings and once with NAL-NL2 settings, in random order. The participant had to indicate which one was preferred and by how much. Judgments of overall sound quality were obtained for female and male speech in quiet and for four types of music (classical, jazz, a man singing, and percussion). Judgments of speech clarity were obtained for female and male speech in speech-shaped noise, female speech in a male-talker background, and male speech in a female-talker background. Factors investigated included compression speed (slow or fast) and input sound level (50, 65, or 80 dB SPL). The pattern of the results was reasonably consistent across participants, but the magnitude of the effects was small. For judgments of overall sound quality, nine participants preferred CAM2 relative to NAL-NL2, and the remainder showed no clear preference. There was a significant overall preference for CAM2. The preference for CAM2 over NAL-NL2 was present for all types of stimuli, both compression speeds, and all three levels. For judgments of the clarity of speech in noise, five participants preferred CAM2 over NAL-NL2, one showed the opposite preference, and the remainder showed no clear preference. There was a significant overall preference for CAM2. The preference for CAM2 over NAL-NL2 was present for all types of stimuli, both compression speeds, and all three levels. For judgments of the clarity of speech in a background talker, CAM2 was significantly preferred overall relative to NAL-NL2, but the effect was very small. Further work is needed to establish whether similar preferences would be found in everyday life.

6:15 PM

Tired of Listening: Subjective and Objective Measures of Hearing Related Fatigue

Benjamin W.Y. Hornsby
Vanderbilt University

Subjective reports from the literature have suggested for many years that fatigue was an important, but overlooked, consequence of hearing loss. Consider this anecdotal report from a person with hearing loss: "I crashed. This letdown wasn't the usual worn-out feeling after a long day. It was pure exhaustion, the deepest kind of fatigue. I took a nap hoping it would refresh me, but when I woke up three hours later I was still so tired I gave up on the day.... The only cause of my fatigue I could identify was the stress of struggling to understand what those around [me] were saying..." (Copithorne, 2006). Despite the serious consequences of fatigue, its relationship to hearing loss and speech processing remains largely unexplored. Existing work is limited primarily to subjective or qualitative measures. Research using objective measures to explore speech processing and hearing related fatigue are lacking.

This presentation describes a series of experiments using subjective and objective measures to explore the relationship between speech processing, hearing loss, hearing aids and mental fatigue. We define fatigue subjectively as a mood state and objectively as a decrement in performance resulting from sustained (45-90 minutes) cognitive demands during a complex speech processing task. In one experiment we examined the effects of task difficulty on the development of mental fatigue in two groups of individuals with hearing loss. Each group was tested with and without hearing aids. In one group (n=14), SNRs were adjusted for each individual to result in aided word recognition of approximately 50% correct during a complex speech processing task. In a second group (n=15), SNRs were individually adjusted to achieve word recognition of approximately 75% correct. Subjective and objective measures of mental fatigue, obtained with and without hearing aids, were compared. In a second experiment we used the same paradigm and a secondary objective measure of fatigue, the Psychomotor Vigilance Task, to examine the
effects of directional and digital noise reduction processing on mental fatigue associated with sustained speech processing. Subjective and objective data suggest that, compared to unaided listening, hearing aid use has the potential to reduce listening effort and susceptibility to mental fatigue resulting from sustained speech processing demands. Advanced signal processing strategies, specifically directional processing and digital noise reduction, may provide some small additional benefit, in terms of reduced listening effort and susceptibility to mental fatigue, at least in some listening conditions. [This research was supported, in part, by grants from Starkey, Inc, Phonak, Inc, and the Dan Maddox Foundation.]

6:30 PM

Hearing Fitness for Duty for Military Personnel

Odile H. Clavier¹, Jed C. Wilbur¹ and Sigfrid D. Soli²

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Introduction: Presently, in the military, hearing abilities are only assessed with existing audiology tools, which rely primarily on standard auditory thresholds and monaural speech tests without regard to hearing-critical needs or noise environment. Assessing hearing fitness-for-duty requires a rigorous analysis of hearing critical tasks (HCTs) that Soldiers of various military specialties must perform in order to complete their missions. Development of new hearing standards for the Army requires hearing tests which correlate well with functional hearing tasks.

Methodology: A survey was developed to identify HCTs of dismounted soldiers. Subject Matter Experts (SMEs) were selected from instructors and combat trainers with a broad knowledge of the functional requirements for their military specialty. Interviews, ambient sound and noise recordings were conducted in several relevant locations. A human subject study was conducted to compare performance on the Hearing In Noise Test (HINT) with performance on functional tests. The functional tests consisted of HINT sentences in relevant military background noise and HINT sentences filtered through a radio transfer function combined with functional noise. The tests were administered to a cohort of normal hearing and hearing impaired subjects. Intelligibility versus the extended speech intelligibility index (ESII) was used as the comparison metric between the standard test and the functional tests.

Results: Using the results of the interviews with the SMEs, as well as ambient noise analysis, a list of hearing critical tasks was assembled and includes: hearing shouted speech in a noisy background (naked ear), hearing radio communications in background noise, hearing and listening to multiple radio channels at once, and noise discipline among others. Functional noise was processed to use as background noise in speech tests. Speech intelligibility obtained with the standard HINT was compared to speech intelligibility on functional tests. The military background noise is highly variable, both in the temporal and spectral sense and it includes competing speech, informational masking, and impulsive noise. As a result, initial testing showed the use of the (ESII) provided a much better metric than the signal-to-noise ratio in relating performance between the standard and functional tests. Study results with both normal hearing and hearing impaired subjects show that using the HINT scores, and an analysis of the background noise in which a person must perform their duty is a good predictor of performance on a functional test of HINT sentences in functional noise. [This work was supported through Army SBIR Contract # W81XWH-09-C-0048. The abstract has been approved for public release.]
First European results with a new transcutaneous bone conduction hearing implant

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\(^4\) Department of Oto-Rhino-Laryngology, University of Wuerzburg, Germany
\(^5\) VIBRANT MED-EL, Innsbruck, Austria

Introduction: Many patients who suffer from conductive or mixed hearing loss receive little or no benefit from acoustic amplification due to the conductive component in the middle ear. Many patients do not derive adequate benefit from hearing aids, may not be able to use hearing aids, or may be dissatisfied with their hearing aids. The Bonebridge\(^\circ\) is a partially implantable transcutaneous bone conduction hearing system for the treatment of conductive and mixed hearing losses up to moderate inner ear hearing impairment. The Bonebridge is intended for persons who cannot wear conventional acoustic hearing aids for medical reasons, or who are unsuccessful acoustic hearing aid users and who need hearing treatment via bone conduction means. Many patients who could be suitable for percutaneous bone conduction hearing systems but, who for medical reasons (e.g., scalp condition that precludes use of a percutaneous implant), need a transcutaneous system would be indicated for the Bonebridge.

Materials and Methods: 12 German speaking adults with either conductive or mixed hearing loss have been implanted with the Bonebridge (Vibrant MED-EL, Innsbruck, Austria) during this clinical investigation at 4 European clinics. Study endpoint was at 3 months after implantation. Safety evaluation included pre- and postoperative bone conduction testing and questionnaires to detect surgical and/or medical complications. Effectiveness evaluation included pre- and postoperative sound field testing, speech recognition threshold testing in quiet, and word recognition score testing. Furthermore, a questionnaire on subjective device benefit was evaluated.

Results: Evaluation of residual hearing revealed no significant decrease in bone conduction thresholds. No surgical complications were reported. Medical complications were few and included transient postoperative vertigo and transient tinnitus, and subcutaneous seroma. Audiometric results showed a significant benefit in terms of aided thresholds, word recognition score, and speech reception thresholds. Subjective benefit was high and reached a mean value of 79\% with the device.

Discussion: Patients implanted with the Bonebridge show a significant audiometric and subjective benefit when compared to the unaided preoperative condition. Complication rate was low.
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SESSION FOUR

Individual Differences in Patients

Moderator: Benjamin Hornsby

8:00 AM  Individual differences: Impact in the lab and clinic

Catherine Palmer
University of Pittsburgh School of Health and Rehabilitation Sciences

Individuals with hearing loss are not a homogeneous group yet most of our research designs depend on group data. In turn, the evidence base we extend to the clinic is based on average data. With the recent emphasis to establish evidence based practice on randomized controlled studies, the ability to account for individual variability is further compromised without a clear understanding of the variability that may be expected in sound perception by new hearing aid users and how to account for this variability in clinical practice. Our lab has focused a great deal of energy in quantifying the time course and extent of adaptation to newly amplified signals over the past decade. Data from our work in adaptation will be used to highlight individual variability and its impact on interpreting average data. Current and potential strategies to account for individual differences in hearing aid fitting protocols will be described in the context of ideal practice and sustainable practice given the demands of scientific rigor and clinical productivity. Needed research in this area also will be discussed.

8:40 AM  Predicting individual differences in aided speech understanding in older adults

Larry E. Humes, Gary Kidd and Jennifer Lentz
Dept. of Speech & Hearing Sciences, Indiana University

This study was designed to address individual differences in aided speech understanding among a relatively large group of older adults. The group of older adults consisted of 98 adults (50 female and 48 male) ranging in age from 60 to 86 years (mean = 69.2 years). Hearing loss was typical for this age group and about 90% had not worn hearing aids. All subjects completed a battery of tests, including cognitive (6), psychophysical (17), and speech-understanding (9) measures, as well as the Speech, Spatial and Qualities of Hearing (SSQ) self-report scale. Most of the speech-understanding measures made use of competing speech and the non-speech psychophysical measures were designed to tap phenomena thought to be relevant for the perception of speech in competing speech (e.g., stream segregation, modulation-detection interference). All measures of speech understanding were administered with spectral shaping applied to the speech stimuli to fully restore audibility through at least 4000 Hz. The measures used were demonstrated to be reliable in older adults and, when compared to a reference group of 28 young normal-hearing adults, age-group differences were observed on many of the measures. Principal-components factor analysis was applied successfully to reduce the number of independent and dependent (speech understanding) measures for a multiple-regression analysis. Doing so yielded one global cognitive-processing factor and five non-speech psychoacoustic factors (hearing loss, tone-in-noise masking, multi-burst masking, stream segregation, and modulation detection) as potential predictors. To this set of six potential predictor variables were added subject age and performance on the text-recognition-threshold (TRT) task, a visual analog of interrupted speech recognition. These eight variables were used to predict one
global aided speech-understanding factor and one global SSQ factor. A stepwise multiple regression analysis accounted for 53% of the total variance in aided speech understanding with 29.5% of the total variance accounted for by the global cognitive-processing factor. Additional significant predictor variables and the variance accounted for by each (in parentheses) were as follows: age (11.3%), multi-burst masking (5.8%), tone-in-noise masking (3.9%), and hearing loss (2.5%). For the SSQ, the multiple-regression solution, though significant, was not as successful, accounting for only 22.1% of the total variance with two predictors: global cognitive processing (15.2%) and hearing loss (6.9%). The implications of these results for our understanding of the relative contributions of various predictors of aided speech understanding in older adults will be discussed. [Work supported, in part, by NIA R01 AG008293.]

9:10 AM  Hearing health beliefs

Gabrielle Saunders, Melissa Papesh, Melissa Teahen and ShienPei Silverman
National Center for Rehabilitative Auditory Research

Help seeking for hearing impairment (HI) and uptake of hearing aids is notoriously poor, but does not differ greatly from help seeking for other chronic conditions. Further, factors associated with help-seeking behaviors are common across chronic medical conditions. We thus argue that help seeking should be examined within the framework of a generic multifactorial model, such as the Health Belief Model (HBM; Rosenstock, 1966). The HBM was developed in order to understand individual differences in decisions to practice health behaviors. Studies have shown it to predict health-related behaviors, such as prenatal care visits and enrollment in diabetes-related pharmaceutical services.

To examine the applicability of the HBM to hearing aids, a 24-item Hearing Health Beliefs Questionnaire (HHBQ) was developed to assess the constructs of the HBM. The HHBQ has seven scales, interpreted as follows. (1) Perceived Susceptibility - extent to which individuals believe themselves susceptible to HI. (2) Perceived Severity - extent to which individuals believe HI has negative consequences. (3) Perceived Impacts - extent to which individuals believe HI could impact them. (4) Perceived Benefits - extent to which individuals believe hearing is important. (5) Perceived Barriers - barriers that restrict hearing aid uptake/use. (6) Perceived Efficacy - confidence in ability to obtain help for HI. (7) Cue to Action - external factors influencing hearing aid uptake/use decisions.

HHBQ data were collected from 156 individuals attending an appointment at a VA Primary Care clinic. Analyses show scores on the HHBQ to differ among subsets of participants. For example, individuals who reported having had a recent hearing test had significantly higher scores on the Perceived Benefits and Perceived Efficacy scales than those who had not. Individuals who report HI but do not have hearing aids had significantly higher scores on the Perceived Susceptibility, Perceived Efficacy, and Cue to Action scales, than those who do not report HI. Individuals who report regular use of their hearing aids have significantly higher scores on the Perceived Benefits and Cue to Action scales than individuals who do not regularly use their hearing aids.

Data will be presented in the context of issues associated with help seeking for HI. Application of the HHBQ to our understanding of hearing help-seeking behavior, prediction of hearing aid outcome, and hearing-aid counseling will be discussed.
Hearing screening for noise-induced hearing loss based on speech intelligibility in background sound

S. Jansen¹, H. Luts¹, P. Dejonckere¹,², A. van Wieringen¹ and J. Wouters¹
¹ ExpORL, Dept. Neurosciences, KU Leuven, Belgium
² Federal Institute of Occupational Diseases, Brussels, Belgium

Introduction: Screening for noise-induced hearing loss (NIHL) is of major importance for noise-exposed workers, as well as for the rising group of teenagers and young adults frequently exposed to high levels of music in their free time.

Current hearing screening programs (e.g., in occupational medicine) are based on measuring pure-tone thresholds. However, reliable and valid screening results can only be obtained in a sound-proof booth, by a well-trained test administrator, using a high-quality, well-calibrated audiometer, and by determining thresholds for a large range of different frequencies. This makes the test time-consuming and expensive, which is not ideal for screening.

Since a speech-in-noise test uses supra-threshold stimuli presented in a controlled background sound, the results are influenced less by the testing environment and the absolute presentation level. It can also be implemented as a quick automatic self-test, potentially through the Internet. However, its sensitivity for NIHL was unclear up to present.

Aims: In this study, we investigated the sensitivity of the “Digit Triplet” speech-in-noise self-test for detecting and monitoring NIHL, and its similarity across two different languages. Compared to a regular sentence test, the Digit Triplet test is expected to have a better within-subject reliability and to be less influenced by non-auditory cognitive abilities.

Methods: 122 noise-exposed workers participated (84 Dutch speaking and 38 French speaking), representing the whole range from no to severe NIHL. Both pure-tone thresholds (in a professional setting) and the Digit Triplet speech reception threshold (in a home-like setting) were collected. One SRT measurement only took 3 to 4 minutes.

Results: For the 84 Dutch speaking participants, we found a strong linear relation between the two tests (R=0.858). Even at the lower range (better listeners), the linear relation did not level off as seen for sentence tests, which might be explained by the reduced redundancy of digit triplets. The sensitivity and specificity to detect subtle NIHL (PTA₂,₃,₄,₆ > 15 dB HL) was 0.98 and 0.83. When screening for hearing aid candidacy (moderate hearing loss), the sensitivity and specificity were 1.00 and 0.91. With a within-subject standard deviation of only 0.8 dB, the Digit Triplet test proved to be highly reliable. Preliminary results for the French speaking subjects showed similar findings.

Conclusions: The Digit Triplet test proved to be sensitive for detecting different degrees of (recreational or occupational) NIHL. Due to its simplicity and robustness, it is highly suitable for screening purposes.
SESSION FIVE  
Frequency Transforming Technologies  
Moderator: Mary Cord  

11:10 AM  Frequency lowering technologies—benefits and limitations  
Susan Scollie  
University of Western Ontario, Canada  
Over the past decade, advances in hearing aid digital signal processing have allowed the development of novel frequency lowering strategies that function in real time in widely available commercial devices. The clinical purpose of frequency lowering signal processing is to overcome the limitations of hearing aid bandwidth, or the limitations of residual hearing ability, by lowering high frequency sound and to a lower frequency region. This has been implemented in a variety of schemes that vary: adaptive or fixed processing, transposition or compression processing, and other factors. These processors are implemented in commercial products with software that allow the clinician to adjust the strength of the processor from very mild settings to very strong settings. This raises the possibility that some settings may be appropriate for some losses, while other settings may be inappropriate. Past studies have indicated that candidacy may be affected by developmental status, the magnitude and configuration of hearing loss, acclimatization and task type. In developing our understanding of this technology, we may consider the roles of fitting and fine tuning for the individual and objective indices of frequency lowering effects. This presentation will provide an overview of these issues, as well as recent data exploring the relationships between electroacoustic and behavioral metrics of the effects of frequency lowering strategies.

11:45 AM  Additive synthesis of frequency-lowered consonants  
Kelly Fitz, John Ellison and Tao Zhang  
Starkey Hearing Technologies  
High frequency sounds are critical to speech intelligibility, with a substantial portion of audible speech cues occurring at frequencies higher than 3 kHz. Restoration of audibility for these high frequency speech cues is often constrained by the power available in the hearing aid, by the amount of gain that can be applied without introducing feedback, and by the extent of the cochlear damage. Recent strategies for restoring audibility of critical high-frequency cues have translated the high-frequency information to lower frequency regions in which hearing loss is less severe. Most algorithms apply non-linear processing to the input signal to translate high-frequency features to lower frequencies. Methods for synthesizing translated spectral features, rather than generating them from the input signal, have been proposed, though no commercially available hearing aid uses such a process. We present and evaluate a spectral feature synthesis method intended for use in frequency translation processing. High frequency spectral features are identified and characterized in the signal to be processed, and translated features are then synthesized from that characterization. Adapting a technique from musical sound synthesis, spectral features are rendered by additive synthesis of spectral peaks, where the additive components are modulated copies of a single prototype narrowband noise. The proposed method gives a high degree of control and predictability of the translated feature spectra, and is pract-
cal for implementation in a digital hearing aid. We report on the results of an evaluation of the intelligibility of translated spectral features synthesized in this way, and a comparison with previously-proposed frequency lowering strategies. To isolate the effects of spectral feature synthesis from the effects of the spectral feature identification, we employed a single, "perfect", ground truth features analysis to identify high-frequency consonant features for all processing strategies. We compare performance on a word-final /s/ detection test, because this high-frequency cue provides significant linguistic information in spoken English. We also assess confusability among fricative consonants, since increased confusions are a commonly-reported consequence of frequency translation processing.

12:15 PM  

Inverse frequency compression for precipitous hearing loss  
JOSHUA M. ALEXANDER  
Purdue University, West Lafayette, Indiana

Common to almost all frequency-lowering techniques proposed over the last 50 years is a positive rank scaling of frequency, such that the ordering of frequency components in the input (before lowering) from lowest to highest is maintained in the output (after lowering). The fundamental problem with this approach for severely band-limited speech is an increase in confusions between sibilant fricatives. In particular, frequency-lowered “s” is more often perceived as “sh” which naturally has a lower spectral peak. Data collected on normal-hearing listeners unexpectedly revealed a non-monotonic relationship between the frequency of the lowered peak and perception in which classification of “s” returns when spectral peaks are very low in frequency. In the band of interest, this relationship appears as a reciprocal function of frequency. The Inverse Frequency Compression (IFC) algorithm uses this relationship as a primary component to remap frequency, adds a time-activation component that triggers the algorithm only when sufficient high-frequency energy is present, and adds a relative level component that complements the spectral differences provided by the frequency remapping.

The IFC algorithm was tested against a variety of positive rank scaling methods using normal-hearing listeners who identified consonants and vowels in nonsense syllables low-pass filtered at 1500 Hz. IFC outperformed the low-pass control and alternative methods on not only “s” and “sh” but also on several other consonants. Relative to the low-pass control, there was no degradation in overall vowel identification.

The IFC algorithm was also tested using hearing-impaired listeners with normal to moderate low-frequency thresholds. Several control conditions were used to understand how IFC compares to conventional amplification, to a popular nonlinear frequency compression technique (Simpson et al., 2005, Int J Audiol, 44, 281-92), and to an algorithm similar to IFC in which frequency remapping was not inverted. For each processing condition, listeners were provided with up to 90 minutes of training/exposure before being tested. Two groups were tested. Group 1 had a bandwidth of 1500 Hz and Group 2 had a bandwidth of 2500 Hz. Relative to conventional amplification, IFC did not degrade speech understanding. For Group 1, the greatest advantage of IFC was seen for fricative and affricate identification, which improved by over 15% in the most challenging listening context. For Group 2, the greatest advantage of IFC was seen when compared to nonlinear frequency compression, which significantly degraded speech understanding, especially vowel identification. [Supported by NIDCD 1RC1DC010601 and the Ewing Marion Kauffman Foundation.]
Telemedicine and teleaudiology: A new approach in hearing health care?

Stefan Launer¹, Gurjit Singh² and Jean Anne Jordan¹

¹ Phonak AG
² University of Toronto

The recent developments in connectivity of electronic devices has opened the door to completely new approaches also in healthcare and medical service provision. Today applications range from mental telehealth, dermatology, OB/GYN, monitoring chronic conditions while the patient is at home (diabetes, stroke,) to “mobile health apps” (watch your diet, fitness trackers.)

Telemedicine is the delivery of medical services at one location from another using a telecommunications medium. Currently telemedicine is poised to have an increasingly larger role in healthcare systems. Major drivers for telemedicine are increasing access to specialists (even in developed countries) on one side but also decreasing cost of service delivery (transportation, logistics, specialist role.) In the field of hearing health care also a number of applications are emerging. Telemedicine is being used as a means to provide audiological and ENT diagnostic expertise in daily clinical routine, to provide support in planning specific surgeries such as cochlear implantations but also to do cochlear implant mapping and fitting remotely.

Potential benefits associated with the use of teleaudiology include increasing client access to audiological services, reducing the costs and inconvenience associated with travel to appointments, minimizing stress on family caregivers who often provide transportation to appointments, increasing the ability to rapidly respond to audiological issues, supporting functional independence by seniors.

Important requirements for successful application of telemedicine are technical on one side but many other process aspects of health care service delivery need to be considered too. When the patient and the health care professional do not reside in the same room anymore a lot of additional, indirect information is lacking, eg facial expression, body language etc. Furthermore, specific diagnostic or verification tests, eg diagnostic measurement of ABR or RECD measurement for verifying the gain of a hearing instrument, require specific technical know-how in administering them. One of the major challenges is to assure the quality of audiological service in such a new approach. This talk will give an overview over current trend in teleaudiology and discuss specific challenges and potential solutions.

A survey of attitudes toward tele-audiology

Harvey B. Abrams
Starkey Hearing Technologies

Purpose: Research, using an online discussion forum, was conducted to investigate the attitudes of audiologists towards tele-audiology.
Methods: Potential participants were carefully screened to ensure a wide representation of age, work environment, relevant clinical experience, as well as to ensure receptiveness to alternative forms of service delivery. Participants responded to 49 questions over a 3-day period to questions posted on a virtual bulletin board that addressed: current knowledge of telehealth technologies; reactions to audiology-specific telehealth applications (screening, diagnostics, intervention); and specific ways in which telehealth might be executed in their practice. Eighteen hearing professionals were recruited. Thirteen started the discussion and 10 participated through to the last day.

Results: Remote hearing aid programming capabilities and mobile phone/smart phone applications were the most well-received and viewed as most personally valuable. Remote audiometry and home appliance/home-based hearing aid upgrades were not rated as positively and many did not feel they would be personally valuable. Diagnostic applications were the least well-received and perceived as the most problematic in terms of implementation and value.

The study design and findings will be discussed in terms of their implications for the adoption of telehealth technologies among the audiology community.

5:40 PM Attitudes toward teleaudiology: A comparison of pediatric and non-pediatric hearing healthcare practitioners

Gurjit Singh¹, Kathy Pichora-Fuller¹, Michael Boretzki², Jean Anne Jordan² and Stefan Launer²
¹ University of Toronto
² Phonak AG

Teleaudiology is the delivery of audiological services at one location from another using a telecommunications medium. Currently teleaudiology is poised to have an increasingly larger role in healthcare systems. Potential benefits associated with the use of teleaudiology include increasing client access to audiological services, reducing the costs and inconvenience associated with travel to appointments, minimizing stress on family caregivers who often provide transportation to appointments, increasing the ability to rapidly respond to audiological issues, supporting functional independence by seniors, and limiting CO₂ emissions associated with increased travel. Despite these potential benefits, more than 75% of telemedicine initiatives ultimately fail (Berg, 1999, Int J Med Inform, 55: 87-101). Meta-analytic reviews (e.g., Broens et al., 2007, J TelemedTelecare, 13: 303-309) suggest that successful telemedicine interventions are critically dependent on the attitudes, perceived usefulness, and acceptance of the technology by healthcare professionals and other key stakeholders such as patients and health care administrators.

It seems that the relationship between the attitudes of practitioners and their acceptance of telemedicine is a key factor, yet almost no research has considered the attitudes of practitioners toward teleaudiology. Here we report on a study designed to explore this factor. Specifically, a survey assessing attitudes toward teleaudiology was completed by practitioners who indicated that their primary caseload consisted of adults (N = 126) or infants/children (N = 55). The sample consisted of Canadian and American hearing healthcare professionals working in either publically-funded or for-profit environments. Overall, the majority of respondents indicated that teleaudiology is likely to have a minimal effect (either positive or negative) on the quality of hearing healthcare, although many respondents indicated that they thought that teleaudiology would have a positive effect on accessibility to service. Nevertheless, a minority of respondents indicated that they thought teleaudiology would likely have a negative effect on quality of care. Inter-
Interestingly, more positive attitudes toward teleaudiology were observed in the group of pediatric practitioners than in the group of clinicians who primarily see adults. Furthermore, pediatric practitioners expressed more willingness to conduct teleaudiology appointments with patients less than 18 years of age than the group of non-pediatric practitioners. These findings suggest that the previously reported reluctance of hearing healthcare practitioners to conduct teleaudiology appointments with pediatric populations may be due, in part, to a practitioner’s familiarity and expertise in conducting audiology appointments with children. Implications for the implementation of teleaudiology will be discussed.

6:05 PM  Professional online rehabilitation of adult hearing-aid users, a randomized controlled trial

Elisabet Sundewall Thorén¹², Marie Öberg¹³, Gunilla Wänström¹³, Gerhard Andersson⁴⁵⁶ and Thomas Lunner²⁴⁵

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⁵ Department of Behavioural Sciences and Learning, Linköping University, Sweden
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By using the Internet in the audiologic rehabilitation process it is possible to cost-effectively include components in the rehabilitation, such as guidance about communication strategies, hearing tactics and how to deal with hearing aids.

In related fields, the Internet has proven to be a good medium to use in different rehabilitation processes. In different studies patient groups who suffer from tinnitus, anxiety and depression have been helped by participating in Internet rehabilitation.

The purpose of this study was to examine and evaluate the effect of an online rehabilitation program for experienced hearing-aid users. The rehabilitation program consisted of self-study, guidance by professional audiologists and the possibility to discuss with other participants in a monitored, online discussion forum. The effects of the rehabilitation program were evaluated with online questionnaires and the results were compared with a control group. The control group had access to literature about historical hearing aids while waiting to undergo the online rehabilitation program.

78 hearing-aid users, recruited via advertisements in newspapers, participated in the study. The participants were on average 69 years and had used their hearing aids for at least a year.

Results from the study show significant improvements for the intervention group, for example, they estimated their problems related to the hearing loss as significantly lower after the study than they did before they took part in the rehabilitation program, while participants in the control group did not change the perception of their hearing problems.
A truly self-fitting hearing aid: feasibility and challenges

Gitte Keidser\(^1,2\), Elizabeth Convery\(^1,2\), Harvey Dillon\(^1,2\), Andrea Caposecco\(^3,2\), De Wet Swanepoel\(^4\) and Lena Wong\(^5\)

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For millions of people with hearing impairment, primarily living in developing countries, the ability to obtain hearing healthcare services is significantly compromised by a paucity of trained audiologists to fit hearing aids or provide diagnostic/rehabilitative services. Even in developed countries such as the USA, Europe, Canada and Australia, access to hearing services may be limited in non-metropolitan regions. One potential solution to meet this service gap is a hearing aid that can be “self-fitting”; that users can program themselves without direct diagnostic or fitting services provided by an audiologist or the need for access to a computer. A possible concept for such a device is a behind-the-ear style instrument equipped with an on-board tone generator that enables automated measurements of in situ hearing thresholds, from which the initial settings of the hearing aid parameters are derived, and a training algorithm that enables subsequent fine tuning of the device in the user’s own everyday listening situations. The potentials and limitations of this concept are currently under investigation.

To date the following questions, which will be discussed in this talk, have been addressed: 1) Can older adults across cultures and languages manage the task of assembling selected hearing aid components (body, tube and instant-fit tip), insert a battery, and insert the aid into the ear from a set of instructions specifically designed to take low health literacy into account? 2) Is it feasible to obtain valid and reliable in situ threshold measurements that are entirely managed through a hearing instrument? 3) Can contraindications for hearing aid fittings, such as asymmetry and conductive loss be measured with a self-fitting device? and 4) Is training of the instruments in own environments effective? Data collected to date lend support to the feasibility of a truly self-fitting hearing aid on all these accounts. In particular, it has been found that a conductive loss can be predicted with high accuracy from automatic tone tests presented in quiet and in noise.
SESSION SEVEN
EAS/Implantable Options

Moderator: Ian Bruce

8:00 AM  Combined electric and acoustic stimulation (EAS): hearing preservation cochlear implantation

René H. Gifford
Vanderbilt University, Nashville, TN

Cochlear implant recipients with preserved acoustic hearing in the implanted ear have two acoustically hearing ears that could potentially code interaural time and intensity differences as well as redundant acoustic information. The potential advantages of having binaural acoustic hearing will not be large when speech and noise are presented from a single loudspeaker—as is typically employed in most busy clinical settings. To assess the effectiveness of preserved hearing in the implanted ear, we collected speech recognition data from 20 adult hearing preservation patients who were implanted with either a Nucleus Hybrid S8, L24, conventional N24 series implant, or Med El Sonata implant. Speech recognition was assessed using HINT, AzBio, and TIMIT sentences in conditions including 1) a diffuse restaurant noise originating from eight loudspeakers placed circumferentially about the patient’s head and, 2) reverberation with a reverberation time (RT) of 0.6 seconds. Speech originated from a single loudspeaker fixed at either 0°, 90°, 270° or rotating source azimuth. Interaural time difference thresholds were also measured for a 250-Hz signal for a subset of the population. For all patients, speech recognition was assessed in the bimodal condition—with the ipsilateral ear occluded—as well as in the best aided EAS condition. Results showed a significant advantage for the EAS condition with the two acoustic hearing ears in noise, with all source azimuths, and in reverberation. Interaural time difference thresholds were found to be highly correlated with the degree of postoperative EAS benefit and were not significantly correlated with auditory threshold at the signal frequency. These data demonstrate 1) the efficacy of hearing preservation in the implanted ear with either a short or long electrode array for improving speech recognition in complex listening environments, 2) that the preservation of binaural cues is possible following surgical insertion of an electrode array, and 3) using standard clinical measures of speech perception will not highlight the benefit derived from hearing preservation cochlear implantation.

8:40 AM  Benefit of advanced directional microphones for speech understanding in cochlear implant users

Iris Arweiler¹, Stefan Fredelake¹, Phillippp Hehrmann¹, Karl-Heinz Dyballa², Volkmar Hamacher¹ and Andreas Büchner²
¹ Advanced Bionics GmbH, European Research Center, Hannover, Germany
² Department of Otolaryngology, Medical University of Hannover, Germany

In noisy listening conditions, multi-microphone technology represents the most effective method to increase the signal-to-noise ratio and thereby improve speech intelligibility. Hearing-impaired listeners have long been able to benefit from directional microphones in hearing aids while this technology was only recently introduced in cochlear implants. A conventional directional microphone is typically a combination of two omnidirectional
microphones that form specific beam patterns. With the possibility to wirelessly exchange audio signals between two hearing aids a binaural beamformer became available that can compare the signals from both ears and thus provide even more sophisticated beam patterns.

The benefit of such a binaural beamformer was tested with 12 cochlear implant users. Two state-of-the-art hearing aids were used that communicated with each other via a wireless link. One of the hearing aids was modified so that the analog output could be fed into the external input of the cochlear implant speech processor. Speech intelligibility was measured unilaterally with the Oldenburg Sentence Test (OlSa) presented from a loudspeaker at 0° azimuth. Stationary speech shaped noise was presented at 65 dB SPL from five loudspeakers positioned at ±70°, ±135° and 180° azimuth. The level of the speech was varied adaptively to determine the speech reception threshold (SRT). In addition to the binaural beamformer, the adaptive directional microphone and the omni directional microphone were evaluated. Each of these hearing aid microphone modes was furthermore combined with the noise reduction system ClearVoice.

With the adaptive beamformer the SRT was on average improved by 5 dB compared to the omni directional microphone. The binaural beamformer further increased speech intelligibility, resulting in a 7 dB advantage compared to the omni directional microphone. ClearVoice provided an additional benefit for most CI users independent of the type of microphone.

A binaural beamformer as used in modern hearing aids can significantly improve speech intelligibility for unilateral CI users. Currently, this technology is also being evaluated with bilateral CI users. For both, bilateral and bimodal CI users a binaural beamformer represents an attractive solution for noise reduction because no additional equipment is needed due to the wireless communication link. The results are encouraging to further consider hearing aid technology for cochlear implants.

9:10 AM  Noise reduction optimized for cochlear implants
S.J. Mauger¹, K. Arora¹², P.W. Dawson¹², A.A. Hersbach¹ and J.M. Heasman¹
¹ Cochlear Limited
² HEARing CRC

Objectives: Noise reduction (NR) strategies in hearing aids have had success in improving listening quality in noisy environments. In cochlear implants (CIs), these noise reduction algorithms have provided significant improvements in speech perception, as well as listening quality. Due to the differences between hearing aid and CI listeners, it is expected that hearing aid NR strategies will not be optimized for CI users. Recent psychoacoustic studies have demonstrated significant differences, and suggested changes optimizing NR techniques specifically for CIs.

Study Design: A NR strategy optimised for CI listeners was tested against the current state-of-the-art NR strategy and the ACE alone program. Everyday SmartSound option (ADRO and ASC) program was used as the baseline signal processing for all three test conditions. Word perception in quiet and sentence perception in noise were used to assess speech perception performance. Noise annoyance rating and overall quality rating were used to test listening quality. Speech perception and listening quality were tested in speech weighted noise, 20-talker babble and 4-talker babble.

Results: Acute clinical testing of an optimized NR strategy found significant sentence perception improvements in speech weighted noise compared to the ACE program by
19% (p<0.001) and the current NR strategy by 7% (p<0.05). The optimized NR strategy also provided significant improvements in babble noise when compared to the ACE program (p<0.001). No intelligibility degradation in quiet was found. Noise annoyance rating was significantly reduced compared to the ACE program by 54% (p<0.001) and the current NR strategy by 24% (p<0.01) in speech weighted noise. Overall quality rating was significantly improved over the ACE program by 49% (p<0.001) and the current NR strategy by 19% (p<0.01). Significant quality improvements were also found in babble conditions.

**Conclusion:** This study has shown significant improvements of this CI optimized NR strategy, and highlights major differences between acoustic and electrical hearing. This optimised strategy has shown benefit over the ACE program and current NR strategies in a range of noise types. Quality ratings of this CI specific NR strategy were also significantly improved over the ACE program as well as over the current NR strategy. This study has also been able to elucidate some of the reasons that CI listeners gain significant speech performance improvements where hearing aid listeners do not from such technologies. The use of CI optimised NR is expected to provide much benefit in real-world environments compared to the ACE program and current NR strategies.

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**9:40 AM Cochlear implant results show efficacy of hearing aids in preserving auditory function**

*Peter J Blamey*[^1] and *Diane S Lazard*[^3]

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Retrospective data from 2251 cochlear implant (CI) recipients implanted between 2003 and 2011 at 15 centres were analysed to identify factors affecting speech perception in quiet. The usual factors (cochlear implant experience, duration of severe-to-profound hearing loss (s/pHL), age at onset of s/pHL, age at implantation, and etiology) had a significant effect. The effects of these factors were compared with their effects in an earlier study that used similar methods (*Blamey et al.*, Audiology & Neuro-Otology 1:293-306, 1996). The negative effect of long duration of s/pHL was less important in the new data than in 1996; the effects of age at CI and age at onset of s/pHL were delayed until older ages; etiology had a smaller effect; and the effect of CI experience was greater with a steeper learning curve. Relaxed patient selection criteria, improved clinical management of HL, modifications of surgical practice, and improved devices may explain the differences.

The additional significant factors in the newer data were: the pre-operative pure tone average threshold of the better ear, the brand of device, the percentage of active electrodes, the use of hearing aids (HAs) during the period of s/pHL, and the duration of moderate hearing loss (mHL). Many of these factors are associated with plastic effects in the auditory system – leading to slow degeneration during periods of reduced auditory input, and partial post-operative recovery of auditory function when auditory input is restored with a CI.

A new model was designed showing a decrease of performance that started during the period of mHL, and accelerated during the period of s/pHL. The use of bilateral HAs had a protective effect. The use of HAs from the very beginning of hearing impairment slowed down the related central reorganization.
SATURDAY, AUGUST 11

SESSION EIGHT
The Stuart Gatehouse Lecture
Moderator: Graham Naylor

11:30 AM  The localization of sound, without and with hearing aids
Michael A. Akeroyd
MRC Institute of Hearing Research (Scottish Section)

Stuart had a long-standing interest in binaural and spatial hearing. It drove much of his experimental work at the Scottish Section and underpinned the development of the SSQ questionnaire. This talk will summarize what is known about the localization of sound by hearing-impaired listeners, what is known about the effects of hearing aids, and outline some potentially important issues on which little appears to be known. Both Stuart’s own work and more-recent experimental work at the Scottish Section will be outlined.

SATURDAY, AUGUST 11

SESSION NINE
Binaural Issues
Moderator: Stefan Launer

4:45 PM  Physiological prediction of masking release for normal-hearing and hearing-impaired listeners
Ian C. Bruce¹, Agnès C. Léger²,³, Brian C. J. Moore⁴ and Christian Lorenzi²,³

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³ Université Paris Descartes, Paris, France
⁴ University of Cambridge, Cambridge, UK

Léger et al., (JASA 2012) measured the intelligibility of speech in steady and spectrally or temporally modulated maskers. The stimuli were filtered into low- (< 1.5 kHz) and mid-frequency (1–3 kHz) regions. Listeners with high-frequency hearing loss but clinically normal audiograms in the low- and mid-frequency regions showed poorer performance than a control group with normal hearing, but showed preserved spectral and temporal masking release. Acoustic-based speech intelligibility metrics could not explain the patterns of masking release exhibited by these subjects.

Here, we investigated whether a physiologically accurate model of the auditory periphery (Zilany et al., JASA 2009) can explain these masking release data. Model auditory nerve (AN) fiber responses were simulated for a range of characteristic frequencies to produce a place-time pattern of AN activity referred to as a “neurogram”. For each stimulus from
the corpus of Léger et al. (2012), two neurograms were generated: a “mean-rate” neurogram with large time bins and an “all-information” neurogram with small time bins. Using the Neural SIMilarity (NSIM) metric (Hines & Harte, Speech Comm 2010 & 2012), these pairs of neurograms were compared with template neurograms obtained from the corresponding unmasked broadband speech stimulus to compute the predicted intelligibility. The mean audiograms of the different groups of listeners from the study of Léger et al. were simulated in the model by applying appropriate amounts of outer and inner hair cell impairment.

Both the mean-rate and the all-information NSIM metrics qualitatively explain the masking release exhibited by the normal-hearing subjects, but the all-information NSIM is more accurate in describing the relative effects of temporal and spectral dips in the masker on masking release. In addition, the mean-rate NSIM tends to predict a reduction in overall intelligibility and degraded masking release for the hearing-impaired groups of subjects. In contrast, the all-information NSIM continues to correctly predict the pattern of masking release for the hearing-impaired groups. Also under investigation are the effects of loss of AN fibers (as reported in animal studies of sensorineural hearing loss) on masking release. These results will provide insight into the different physiological contributions to suprathreshold deficits in hearing-impaired listeners and into the roles of mean-rate and spike-timing information in the masking release phenomenon. [Supported by the Erasmus Mundus ACN Training Network, NSERC Discovery Grant 261736, the Royal Society (International Joint Project, 2009R3), 7FP-SME222291 Dual-Pro program, MRC (UK), and a CIFRE grant from ANRT and Neurelec.]

5:15 PM

Effects of bilateral compression on spatial unmasking in hearing-impaired listeners

Jing Xia1, Olaf Strelcyk1, Drew Dundas1, Andrew Schwartz2, Sridhar Kalluri1 and Brent Edwards1

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2 Harvard-MIT Speech and Hearing Bioscience and Technology Program, Cambridge, MA

Listeners use spatial cues in acoustically crowded environments to focus attention on target speech that is spatially separated from interferers. Hearing-aid users have difficulty understanding speech in such environments. One of the reasons might be that hearing aids do not transmit spatial cues with adequate fidelity to preserve the benefit of knowing where to listen. Dynamic range compression operating independently at the two ears can optimally restore audibility. However, it reduces inter-aural level difference (ILD) cues by providing more gain to the ear that receives the lower sound level and less gain to the other ear. ILD cues can be preserved by coordinating compression such that the same gain is applied at both ears. We report results on two experiments measuring speech intelligibility for listeners with symmetrical hearing losses, comparing coordinated compression and independent compression. In Experiment 1, two speech interferers are placed symmetrically about the frontal target with various angular separations. Experiment 2 addresses a situation where two interferers are placed on the same side of the target. In addition, a monaural condition is measured with sound presented only to the better ear in terms of signal-to-noise ratio. In both experiments, target speech and interferers have the same sound levels and are high-pass filtered in order to encourage the use of ILD cues. The stimuli are spatialized by convolution with generic hear-related transfer functions and presented via headphones. The findings inform the development of bilateral assistive technology.
The perception of free-field source width in hearing-impaired individuals

William Whitmer¹, Bernhard Seeber² and Michael Akeroyd¹
¹ MRC Institute of Hearing Research (Scottish Section), Glasgow, UK
² MRC Institute of Hearing Research, Nottingham, UK

In a previous study (Whitmer et al., in press), we found that older hearing-impaired (HI) participants were worse than younger normal-hearing (NH) participants at discriminating changes in apparent width based on interaural coherence. When asked to produce visual sketches of headphone-presented noises, older HI responses were insensitive to changes in coherence. The current study further explores this insensitivity by presenting stimuli in the free field to HI and NH participants, and examines potential links in their perception of auditory source width to their sound-localization ability and binaural temporal resolution.

The stimuli for the sketching task are 1000-ms low-pass, high-pass and speech-spectrum noises. A noise is presented from a center (0°) loudspeaker in a sound dampened room. Two independent flanking noises are simultaneously presented from loudspeakers at ±45° to act as arbitrary reflections. The flanking noises are attenuated 0-24 dB in 4 dB increments from the center signal level, mimicking the asymmetric three-generator method for generating partially coherent noises. HI and NH participants sketch the width of each stimulus onto an image of the apparatus in front of them using a touch screen. To investigate how width responses relate to precision in localization, participants also locate 500-ms speech-spectrum filtered click trains in quiet. To consider how width responses relate to binaural resolution, participants also discriminate interaurally phase-shifted tones from diotic tones presented over headphones.

We predict that HI participants will show a greater sensitivity to source width based on interaural coherence than exhibited in our previous headphone study due to the free-field presentation and visual-response anchors used in the current study. The widths ascribed to more punctate sounds are expected to covary with binaural phase discrimination. If confirmed, these results will suggest there are impairment-related changes to the spatial perception of sound sources that are relevant to unaided and aided listening strategies in complex environments for HI individuals. [Supported by intramural funding from the Medical Research Council (grant number U135097131) and the Chief Scientist Office of the Scottish Government.]

Reference:

Effects of bandwidth and microphone location on understanding of masked speech by normal-hearing and hearing-impaired listeners

Sunil Puria¹, Suzanne Carr Levy¹, Daniel J. Freed³ and Michael Nilsson¹
¹ EarLens Corporation, Redwood City, CA
² Stanford University

Three experiments were conducted to investigate the effects of bandwidth and microphone location on target speech understanding in the presence of spatially separated masking speech. Reception thresholds of sentences (RTSs) were measured for two different target and masker spatial configurations: 1) asymmetric, with the target at -45° and two maskers at +45°; and 2) diffuse, with the target at 0° and four maskers, one each at ±45° and ±135°. Hearing in Speech Test (HIST) materials were presented using low-pass filter cut-off frequencies of 4 and 10 kHz, as well as full bandwidth (22.05 kHz).
In Experiment 1, the effect of bandwidth was tested in 24 normal-hearing (NH) listeners in the free field. Mean RTs for the 4 and 10 kHz cutoffs were -16.3 and -19.7 dB in the asymmetric configuration and -7.7 and -10.3 dB in the diffuse configuration. In both configurations, significant improvement was obtained from extending the bandwidth.

In Experiment 2, the effect of bandwidth was tested in 24 mild-moderate sensorineural hearing-impaired (HI) listeners. Spatial configurations were simulated over earphones using Head-Related Transfer Functions (HRTFs) measured at a behind-the-ear (BTE) microphone location. Amplification was provided with a multi-band wide dynamic range compressor, with gain prescribed by the CAM2 fitting algorithm (Moore et al., 2010). Mean RTs for the 4 and 10 kHz cutoffs were -10.6 and -11.9 dB in the asymmetric configuration and -2.1 and -2.4 dB in the diffuse configuration. The improvement from extending the bandwidth was significant in the asymmetric configuration but not in the diffuse configuration.

In Experiment 3, the effect of microphone location was tested in 24 NH listeners with full bandwidth materials. HRTFs measured at the BTE microphone and tympanic membrane (TM) locations were used to simulate spatial configurations over earphones, in order to evaluate the effect of listening through a BTE microphone as compared to natural acoustic listening. In the diffuse configuration, the mean RT was 2.4 dB better in the TM compared to the BTE microphone location, which was a significant improvement. In the asymmetric condition the effect of microphone location was not significant.

The results suggest that acoustic cues above 4 kHz provide an increased ability to understand target speech in masking speech by up to 34% for NH and 13% for HI subjects (using 1dB~10%; Nilsson et al., 1994). The effect of the BTE microphone location likely has an effect on aided benefit in the diffuse condition for HI listeners. [Supported in part by R44 DC008499 SBIR and ARRA supplements to SP from the NIDCD of NIH.]

6:45 PM

Binaural compression weakens spatial unmasking: the compression-blur hypothesis

Andrew Schwartz1 and Barbara Shinn-Cunningham2
1 Harvard-MIT
2 Boston University Center for Computation

Dynamic range compression is used in many hearing aids to manage the reduced dynamic range that often accompanies hearing loss. However, such compression can alter the interaural differences, such as interaural level differences (ILD), that normally allow us to determine the location of sound sources and to selectively attend to one source in a mixture of competing sounds. Many natural sounds have fluctuating levels; because of this, a mixture of sounds from different directions can lead to complicated temporal dynamics in hearing aid compression, as different sources will dominate overall intensity at different moments. As a result, independent compression of the signals reaching the ears can cause the effective ILD of the sources to fluctuate, which may affect hearing aid wearers’ spatial perception in everyday settings. Objectively, measures of localization accuracy and target speech intelligibility yield mixed results when comparing independent compression to linear amplification, while subjectively, compression increases the occurrence of “moving” and “split” images (Wiggins and Seeber, J. Acoust. Soc. Am. 130 (6), 2011). This study explores whether hearing aid compression that is independent at the two ears degrades the ability to selectively attend to a speech target in the presence of spatially separated speech maskers, and whether linking the left and right-ear compressors, which preserves ILD, ameliorates any such degradations.
We hypothesized that independent compression would increase the amount of spatial separation needed to reach a fixed, threshold level of performance in a spatial attention task (“spatial tuning”). We also hypothesized that linking the compressors would improve spatial tuning relative to independent compression. We investigated these hypotheses by asking normal-hearing subjects to attend to a target sequence of spoken digits in the presence of two masking sequences of digits. Linear amplification, independent compression, and linked compression were each tested in separate blocks, with the average overall stimulus level in each block set to be 70 dB SPL.

Consistent with our hypotheses, we found that when using fast (10 ms) attack and release time constants, independent compression broadened spatial tuning compared to the linear condition, and that spatial tuning was sharper when the compressors were linked compared to when they were independent. However, when slower (100 ms) attack and release times were used, no significant effects were observed. The results of this study suggest that speech understanding in multi-talker environments can be adversely affected by fast, independent compression, but that linking the compressors can restore performance.
Poster Program

Posters for Session A should be put up by 8:00 AM Thursday, August 9, and taken down after 10:00 PM Thursday, August 9, or before 7:00 AM Friday, August 10. Presenters should be at their posters from 9:40 AM – 11:00 A.M.; 8:30 PM – 10:00 P.M.

POSTER SESSION A
Thursday 9:40 AM – 11:00 AM

A1
Predicting speech intelligibility in real-world noise environments for screening of functional hearing ability

Akiko Amano-Kusumoto¹, Sigfrid D. Soli¹, Yang-soo Yoon¹ and Ivan Pal²
¹ House Research Institute
² Compreval, Inc

Functional hearing is the ability to detect, recognize, and localize sounds, and to communicate with speech. Certain jobs (e.g., law enforcement) require sufficient functional hearing ability to perform hearing-critical tasks. Proper screening of functional hearing ability is required for individuals who do these jobs. In standard screening procedures, pure-tone hearing thresholds are used, but they do not predict functional hearing abilities, such as understanding speech in non-stationary real-world noise environments. The long-term goal of this study is to develop a functional hearing screening protocol for predicting speech intelligibility in real-world noise environments, where hearing-critical tasks are routinely performed. Specifically, we sought to determine whether speech intelligibility can be predicted in real-world noise environments using the Hearing In Noise Test (HINT) and the Extended Speech Intelligibility Index (ESII). The ESII has been developed to predict Speech Reception Thresholds (SRT) in non-stationary noise environments (e.g., Rhebergen & Versfeld, 2005). Real-world noise environments were identified and characterized in terms of sound pressure level and ESII values. Perceptual experiments were carried out with two groups of subjects: (1) normal-hearing (PTA < 25 dB HL in both ears), and (2) hearing-impaired (PTA ≥ 25 dB HL in at least one ear). The ESII-intelligibility function for each subject was derived by measuring speech intelligibility in stationary noise with the HINT (Nilsson et al., 1994), with signal-to-noise ratios (SNRs) converted to ESII values. The resulting ESII-intelligibility function includes both the audibility and distortion components of sensorineural hearing loss. Speech intelligibility in the real-world noise environments was predicted from ESII analyses of the noise and the subject’s ESII-intelligibility function. Predictions were validated by comparing measured and predicted speech intelligibility at specified SNRs in real-world noise environments.

The perceptual results showed that 85 % of the variance in intelligibility in real-world noise environments can be predicted from HINT measures and ESII analyses, regardless of the subjects’ hearing status and noise condition, evidence of the feasibility of predicting speech intelligibility in real world noise environments. An individual’s ESII-intelligibility function can also be used to predict intelligibility in novel noise environments. The HINT-ESII method can provide a screening technology that incorporates both the audibility and distortion components of sensorineural hearing loss, providing an objective link between the screening measure and the performance of hearing-critical tasks.

A2
PC-based Rehabilitation of Auditory Function

Tanya Arbogast¹, E.W. Yund² and David Woods¹,²
¹ Human Cognitive Neurophysiology Laboratory, VA Medical Center, Martinez CA
² Department of Neurology, UC Davis

The principal complaint of individuals with hearing loss is problems in understanding speech in noise (CHABA, 1988.) Indeed, objective tests show that approximately 90% of Veterans over the age of 65 show some impairment in speech comprehension in the presence of competing
messages (Wilson et al., 2010). While HAs can significantly improve speech comprehension in noise, difficulty in hearing noisy situations is still a leading complaint in aided individuals. The Consonant Identification in Noise Training (CINT) protocol was developed to improve speech comprehension in noise in patients with mild-to-moderate SNHL who wear hearing aids. It was hypothesized that neuroplastic changes in consonant cue weightings caused by SNHL were not effectively reversed or normalized by amplification alone. CINT was designed to produce neuroplastic changes to normalize the utilization of cues that have been made audible by amplification.

Twelve binaurally-aided subjects completed a two-month, aided, in-home training protocol with consonant-vowel-consonant (CVC) syllables in speech-spectrum noise. CINT tokens are comprised of 9,600 multi-talker syllables that include the 40 most common initial and final consonants in spoken American English. Tokens were randomly-sampled with the constraint that all consonants, talkers, and vowels were sampled equally. SNRs during training were adaptively adjusted based on signal detection measures. Subjects were assessed pre- and post-training in the laboratory on consonant identification in noise with the California Syllable Test (Woods et al., 2010) and sentences in noise with HINT, QuickSIN and the California Sentence Test.

Among patients with mild-to-moderate SNHL who wear HAs, two months of CINT improved the SNR at which consonants were identified in comparison with pre-training aided-listening performance. Improvements were seen for syllables used in training and for syllables spoken by unfamiliar talkers. The patterns of improvement for different consonants over the course of training, changes in consonant confusions, and the generalization of training to sentence tests will be discussed.

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A3

Do voices matter? Interactions between hearing aid signal processing and talker characteristics

Kathryn Arehart¹, Huiwen Goy², James M. Kates¹,³, Kathleen Pichora-Fuller² and Pascal van Lieshout¹

1 University of Colorado Boulder
2 University of Toronto
3 GN ReSound

Hearing aid signal processing algorithms are often evaluated with professional recordings of voices. However, hearing aid users often listen to speakers who are older and who may have poor voice quality caused by disorders associated with aging (e.g., Parkinson’s disease). The purpose of this study was to quantify the extent to which the acoustic and perceptual consequences of hearing aid digital signal processing algorithms differ for talkers that vary in their vocal characteristics. Six different talkers were considered: 1) male talker with good voice quality; 2) male talker with moderate voice quality; 3) male talker with poor voice quality; 4) female talker with good voice quality; 5) female talker with moderate voice quality and 6) female talker with poor voice quality, as judged by normal-hearing, naïve listeners. The voices were presented in quiet and in the presence of babble noise (10 dB SNR). The voices were processed with varying amounts of frequency compression, wide dynamic range compression and noise suppression (spectral subtraction). Data are presented showing the interactions between signal processing and voice characteristics. The interactions are measured using the Hearing Aid Speech Quality Index (HASQI), which compares the envelope time-frequency modulation of the hearing-aid processed signal with that of the input. The interactions are also measured using listener ratings of perceived sound quality of the processed speech. Results will have implications for the clinical fitting of hearing aids in terms of providing insight into whether it is important to consider the vocal characteristics of the primary communication partners of the hearing aid wearer in the adjustment of signal processing parameters. [This research was undertaken, in part, thanks to funding from the NSERC, the Canada Research Chairs program, a grant from GN ReSound to University of Colorado, and NIH R01 DC012289-01.]
Relations among detection thresholds for amplitude modulation, frequency modulation, and temporal fine structure

Thomas Baer and Brian C.J. Moore
University of Cambridge, UK

Many recent studies have focused on listeners’ use of the auditory information in temporal fine structure (TFS). Tests have been developed to assess the limits of performance (detection thresholds) based on either monaural or binaural TFS information while reducing the contribution of temporal-envelope and/or excitation-pattern cues. In contrast, amplitude-modulation (AM) detection at moderate levels is thought to involve only temporal-envelope and/or excitation-pattern information. Frequency-modulation (FM) detection, on the other hand, may depend on TFS and/or temporal-envelope excitation-pattern information. Performance in any of these tasks depends on (1) the magnitude of the cues transmitted by the peripheral auditory system and (2) the listener’s ability to process these cues. The present study was designed to investigate the relative role of the latter factor, i.e., general processing efficiency.

Twelve young adult listeners with audiometrically normal hearing (≤20 dB HL) in both ears were tested. For each listener, detection thresholds for AM, FM, and TFS were assessed. The two modulation-detection tasks were performed at two carrier frequencies (0.5 and 2 kHz) using modulation frequencies of 2 and 16 Hz. Sensitivity to TFS near 2 kHz was measured with a monaural task (Moore and Sek, 2009) while a binaural task (Hopkins and Moore, 2010) was used for 0.5-kHz.

The across-subject pattern of threshold measures was analyzed. For the 2-kHz carrier, there was little correlation between the monaural TFS measure and the modulation detection thresholds. For the 0.5-kHz carrier, binaural TFS detection was correlated with FM detection for 2-Hz modulations but not for 16-Hz modulations. Although this suggests that TFS information might contribute to performance in the low-rate (but not the high-rate) FM task, a similar pattern of results was found for AM detection versus TFS sensitivity; there were correlations for the 2-Hz but not the 16-Hz modulators. Since the AM and TFS tasks rely on different information, it appears from these preliminary results that overall auditory-processing efficiency is likely to be the common factor.

References:

Audiological outcomes for children who wear hearing aids

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Through technology and evidence-based protocols, permanent hearing loss in infants can be detected and hearing aids can be provided safely and accurately within the first few months of life. This supports the infant’s natural potential to develop language and literacy skills. The provision of hearing aids to hearing impaired infants involves a process which includes selecting a hearing aid based on accurate assessment information and prescriptive targets and verifying the output of the hearing aid. Another key component of the process is outcome evaluation of device effectiveness in daily life. Clinical practice guidelines typically do not recommend specific measures for tracking the infant’s auditory development and performance over time. Research has focused on the communication outcomes of children involved in EHDI programs and what factors may
impact outcome\textsuperscript{e.g.,1}. In such studies, test batteries developed for research purposes may not be feasible for use in clinical practice.

The current work describes the audiological outcomes of infants, toddlers, and preschool children who wear hearing aids using a systematic outcome evaluation guideline known as the University of Western Ontario Pediatric Audiological Monitoring Protocol (UWO PedAMP\textsuperscript{2}). It consists of functional outcome tools in the form of caregiver report questionnaires. These are supported by each child’s hearing aid fitting information (\textit{i.e.,} Speech Intelligibility Index [SII]). The UWO PedAMP has been implemented in routine clinical practice with a clinical sample of over 115 children who wear hearing aids of varying ages, developmental levels and degrees of hearing loss. The impact of these variables on outcome will be presented through cross-sectional group data as well as individual longitudinal data.

\textbf{References:}


\textbf{A6}

\textbf{Listening effort: Comparison of self-reported effort and dual-task methods based on speech materials with both high and low context}

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A potential benefit of signal processing in hearing aids is a reduction of listening effort, \textit{i.e.,} a reduction of cognitive resources needed when listening to speech. To objectively measure listening effort, different kinds of dual-task tests have been used. Subjective measures of listening effort are also used, but these might not be related to objective measures (Desjardins, 2011; Gosselin & Gagné, 2011). The purpose of this study is threefold. First, a dual-task paradigm using a low context Danish speech material was developed and validated. Secondly, a Danish version of an existing dual-task test was created. Third, a comparison between three tests was performed: the newly developed dual-task test (Dantale CSC), the similar dual-task test using the Danish HINT sentences (HINT CSC), and a self-reported effort using a visual analog scale.

Twenty normal hearing and 12 hearing aid users participated in the study. The dual-task tests consisted of a primary task (listening to and repeating words) and a secondary task which was to remember words for later recall, either in a free or serial recall task. Our hypothesis was that noise would increase the listening effort, and therefore speech was presented in two conditions: Quiet and Noise (+5 dB SNR). The results revealed no difference between the conditions, in neither of the dual-tasks. However, the two groups performed differently in the dual-task tests; the hearing impaired group had a significantly lower score. The analysis of recall as function of word position showed that recall was similar to the findings of previous experiments; both primacy and recency effects were present. The hearing impaired group had lower recall in primacy word positions than the normal hearing group. An analysis of repeated testing indicated that a learning effect was present for the hearing impaired group in Dantale CSC. There was no evidence of any effects of repeated testing in the HINT CSC. In the self-reported effort, as expected, the noise condition was rated significantly more effortful than the quiet condition. However, the self-reported effort did not correlate to the dual-task measures.

The hypothesis that noise would affect the listening effort in the dual-tasks was not confirmed. This finding is contradictory to findings in previous studies. Further developments of the method are therefore discussed.
Auditory stream segregation of sequentially presented vowel sounds
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Hearing-impaired listeners have greater difficulty understanding speech in background noise than individuals with normal hearing sensitivity. This well-established finding may be partially attributed to potential differences in the auditory stream segregation abilities of the two populations. Auditory stream segregation is the ability to separate auditory inputs into differential streams of sounds, and has been studied extensively in normal-hearing listeners. One method of study exploits the rhythm changes perceived in sequentially-presented interleaved stimulus pairs. The purpose of this study was to determine which stimulus characteristics contribute to the segregation of vowel pairs presented alternately in sequence for listeners with hearing impairment and with normal hearing. The vowel pairs had systematic changes in fundamental frequency (F0), as well as frequency differences of the first and second formants (F1 and F2). Four synthetically-produced vowels, 80ms in length, were generated based on the relative mapping of F1 and F2. These vowels were then paired in combinations so that they differed on F0, F1 and/or F2 or had all the same characteristics (i.e., vowels paired with themselves). A rhythm-disruption method described by Roberts et al. (2002) was used to determine if alternating members of a vowel pair were heard either as two segregated streams, or were integrated into one stream. The tendency to integrate or segregate the members of a vowel pair was dependent on differences of both the F0 and the frequencies of F1 and F2, with relative formant structure providing the stronger influence. Interestingly, there were no significant differences in the probability of integrating or segregating the vowel pairs between individuals with hearing loss and those with normal hearing. These findings suggest that sequential vowel processing, at least as measured in this study, may not be affected by cochlear damage. However, because observed effects were related to the formant structure of the vowels, these results may have implications for signal processing in frequency-lowering hearing aids that will alter the frequency difference between formants for some vowel pairs, potentially interfering with the ability to separate targets from background speech. [Work supported by VA Rehabilitation Research and Development Service and NIDCD R01 DC00626]

Towards more realistic speech tests for hearing aid research
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Sentence tests are routinely conducted in the laboratory and the clinic to assess the ability of a particular listener to understand speech. However these tests are often poor predictors of the performance of that listener in real world listening situations, where a high level of individual variability is encountered. Moreover, current laboratory tests often cannot accurately predict the real world benefit of rehabilitation devices/features. As technology advances, better predictions would be extremely useful for demonstrating a competitive advantage of one device/feature over another.

There are several ecologically relevant variables that are missing from current speech tests that may prove to be important, including dynamic variations in spatial and level characteristics of the acoustic environment, the presence of reverberation, and co-ordinated visual information. In addition, verbatim sentence recall does not capture features of real speech communication such as the extraction of meaning and the ongoing engagement of cognitive processes.

Our overall aim is to create new speech tests that provide more accurate predictions of real-world performance. Our hope is that by carefully adding realism to speech tests, we will engage the auditory and cognitive processes involved in real communication tasks, and generate more meaningful performance measures.
An experiment is described that aims to understand the psychophysical effects of introducing realistic variations to speech and noise variables, and preliminary results are presented. Specifically, we examine (a) the effects of conducting a conventional sentence test in a simulated cafeteria that is dynamic and reverberant, and (b) the effects of moving from a speech recall task to a speech comprehension task. The reference task in both cases is a conventional sentence test conducted in anechoic space using spatially diffuse speech babble. Critically, the same set of listeners is used so that correlations across the different tasks can be examined. In addition, psychometric functions are collected to compare sensitivity of the different tests to changes in signal-to-noise ratio. These analyses will establish whether the more realistic tests provide new information over the conventional test, and thus have the potential to better capture individual real-world experiences.

An alternative to the audiogram for hearing aid fitting

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The standard methods for fitting hearing aids use prescriptions based on the audiogram, with the goals of normalising loudness, or optimising speech perception. This study used a new method that is capable of optimising speech perception directly without involving the audiogram. Information transmission measures were derived from perception of consonant-vowel-consonant (CVC) words in the unaided condition. Information transmission measures for voicing, vowel height, consonant place of articulation and other speech features were arranged in order from low to high frequency to provide a graphical representation of the hearing loss, the InfoGram.

A group of normally-hearing listeners were tested with the CVC word test via a hearing loss simulation. The InfoGrams from the normally-hearing subjects were compared with the simulated audiograms. The results indicated that a short test of ten CVC words could provide frequency-specific information equivalent to the simulated audiogram over a range of frequencies from 250 Hz to 4 kHz and over a simulated hearing loss range from 40 to 70 dB HL.

A second experiment was conducted with a small group of potential hearing aid users attending a hearing aid clinic. InfoGrams were measured in the unaided condition, and a frequency-specific gain function was derived from the InfoGram, designed to enhance the perception of speech. An ADRO hearing aid was fitted with the frequency-specific gain function instead of the usual gain function. Another CVC word test was conducted in the aided condition to evaluate the improvements in speech perception and information transmission. A second iteration of the analysis and gain derivation was conducted if the speech perception score was not satisfactory after the first iteration.

A comparison of the speech perception results obtained from fittings based on the InfoGram method were compared with those obtained with fittings derived from a conventional audiogram-based method, and with results for fittings based on a loudness-balancing procedure. The initial speech perception results were similar for all three methods, and greater numbers of subjects will be tested before the conference at IHCON in August.

Toward a Perceptually Relevant Measure of the Occlusion Effect

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GN ReSound

With the increased popularity of receiver-in-the-ear hearing aids, the occlusion effect has been reported as a complaint for cases in which the receiver is large relative to the size of the ear canal. Unfortunately, it is difficult to predict how a hearing aid user will perceive occlusion. Numerous studies have attempted to quantify the occlusion effect, in terms of both objective and perceptual metrics. However, no previous research has been able to show a strong correlation between the objective and perceptual metrics. Part of this
difficulty may be due to the absolute rating scales used in past studies.

In this study, we employ a very different strategy to quantify the perceptual occlusion effect. By directly comparing recordings made in open and occluded ear canals and asking listeners to rate similarity between the two, we are able to consistently measure the amount of subjective occlusion effect. Two pairs of stimuli are compared in each trial, allowing the listener to rate the relative amount of occlusion in two different configurations. Relative occlusion effect is then quantified as the difference between the two configurations, and the subjective value is compared to relative objective metrics for the same pair of stimuli. Preliminary results suggest that the subjective occlusion effect is highly correlated with objective measures of the occlusion effect. We will present the results of this experiment and discuss which objective measures can be used to predict the amount of perceived occlusion.

On the externalization of sounds by hearing-aid users

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Hearing-aid wearers have reported sound source locations as being perceptually internalized (i.e., inside their head). Previous work (Boyd et al., 2012) has shown that hearing-impaired (HI) listeners are not as sensitive as normal-hearing (NH) listeners to possible changes in externalization cues caused by hearing-aid microphone position and bandwidth. These static cues provided neither a full externalization nor full internalization of sounds. We have further explored this apparent contraction or “flattening” of auditory space for HI listeners with an experiment testing the internalizing effects of suddenly loud (impulsive) sounds that have been clinically reported to be more likely internalized by hearing-aid users.

Impulsive sounds have been clinically reported to be more likely internalized by hearing-aid users. To examine this experimentally, 80-dB noise bursts with varying bandwidths, envelopes and front-hemifield directions of arrival were presented to ten HI and six NH listeners in a large room during continuous, rear-hemifield, 60-dB speech. Listeners were asked to rate the degree of externalization experienced on a five-point discrete egocentric scale with and without hearing aids. Hearing aids were set to a flat gain for NH listeners. Direction of arrival had the strongest effect on internalization of impulsive sounds; HI listeners rated on-axis (0°) sounds to be more internalized than off-axis sounds. There was little difference between HI listeners’ aided and unaided ratings of externalization. Large differences, however, were observed between NH listeners’ aided and unaided ratings, but these were attributed to acclimatization issues.

The internalization of impulsive sounds for HI listeners appears to depend solely on the interaural cues available. These results show relatively weak effects of hearing aids in internalization. In addition, a short survey of HI individuals (n = 272) examined the prevalence of clinically reported externalization issues such as impulsive sounds. The survey results corroborated the extant findings that (a) the flattening of auditory space is the primary issue for those who experience a loss of externalization, and (b) only a minority of hearing-aid users report externalization problems. [This work was supported by the MRC, CSO and the University of Strathclyde].

References:


A test battery to assess the benefits of bilateral amplification with hearing aids

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Aim: The aim of this study was to investigate which (combination of) laboratory tests show the full benefit of bilateral fittings.

Methods: Forty experienced bilateral hearing aid users with mild and moderate to severe hearing
losses were tested with one and two hearing aids, together with twenty normal hearing subjects, where the unilateral situation was simulated. Different questionnaires were completed by the hearing impaired subjects and this was objectified by several free field speech reception tests, subjective performance tests and localisation tests.

**Results:** The results of the AVETA questionnaire indicated a consistent benefit from the second hearing aid for speech discrimination in noise and in quiet and a tendency towards a larger effect with increasing hearing loss. The results of the AVETA questionnaire indicated a consistent benefit from the second hearing aid for detection and localization (p<0.05) and also a tendency towards a larger benefit with increasing hearing loss. Regarding SSQ-C, there is a significant effect of hearing loss on the total SSQ score (p<0.05), but not on the individual subscales. The speech subscale shows a bilateral benefit for the total group (p<0.05). The speech reception test showed a positive effect for smaller hearing losses and for bilateral fittings due to spatially separated sound sources. Larger effects were found for higher hearing losses due to the second hearing aid (squelch and head shadow). Detection and localization showed a significant bilateral advantage which was larger for more severe hearing losses (p<0.001). There was no difference in the ANL test between unilateral and bilateral provision.

**Conclusion:** The analyses indicated a bilateral benefit on the speech perception scores, especially when the noise source was located at the unilaterally aided side. Also a bilateral benefit was found for localization abilities for different hearing losses, especially when the source was located at the unilaterally unaided side. A benefit in subjective listening effort is evident, even for subjects with mild hearing losses. This is partly supported by the self-reported measures in the questionnaires. More focus is needed in future research to obtain more information about the sensitivity of the tests.
relevant correlations between diverse screening tests as well as finding some meaningful screening measures in relation to the directionality benefit.

A14

Speech recognition with bilateral and bimodal hearing devices in noise
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In recent years, the current trend to resort to electrical stimulation in both ears (bilateral electrical hearing) has begun to shift, and an increasing number of patients who have residual acoustic hearing are now offered the option of complementing the cochlear implant (CI) on one side with a contralateral hearing aid (HA) on the non-implanted side. Insofar as it can be ascertained from the current literature, the principal benefits of bilateral electrical hearing in speech perception and sound localization are well proven. Still, the literature on whether significant differences exist between bilateral and bimodal stimulation paradigms for any of the listening effects associated with binaural hearing is sparse. In this contribution we assess to what extent listeners with bilateral and bimodal fittings should expect to benefit from binaural hearing advantages. We measure the three primary binaural advantages and compare the performance between bimodal hearing (electroacoustic) and bilateral hearing (electrical) stimulation paradigms. The effects of head-shadow, binaural redundancy (summation) and binaural unmasking (squelch) are assessed in ten normal-hearing (NH) listeners (audiometric thresholds less or equal than 20 dB HL from 0.25 to 4.0 kHz) using IEEE sentences corrupted by steady speech-weighted noise at +5 dB SNR. We report results in settings wherein the target male speech is placed directly in front of the listener (0°) and a single speech-weighted noise source is located either to the right (+90°) or the left (-90°) of the listener. For each of these three spatial configurations, we test presentation conditions through a CI-only, CI + CI and CI + HA.

Our preliminary results indicate that, in most cases, hearing through bilateral electrical stimulation yields significant binaural summation, binaural squelch, and head-shadow effects, while bimodal devices often only secure access to binaural summation and head-shadow benefits. [This work was supported by NIDCD/NIH.]

A15

Acoustical and perceptual comparison between noise-reduction algorithms of different hearing aids
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The use of noise reduction algorithms in hearing aids is almost as common as the use of compression. There are a number of similarities, e.g.: 1) The effects on speech intelligibility are only small 2) The main benefit is found in terms of comfortable listening / sound quality

However, there are also important differences: 1) Noise reduction is commonly presented as a “black box”; there is no information about the details of the signal processing. 2) Compared to compression, there is only little research on hearing-aid noise reduction. Consequently, the perceptual effects of noise reduction are basically unknown and there are no prescription rules.

So there are relatively large uncertainties about the actual implementation of noise reduction algorithms in commercial products, about their perceptual effects, and about fitting strategies. This is quite unexpected and undesired given the broad application of noise reduction in hearing aids.

We conducted a study to gain insight in the “black boxes” and in the perceptual effects of noise reduction. We recorded hearing-aid noise-reduction output and were able to isolate the noise reduction from the other hearing-aid characteristics. We evaluated these signals acoustically as well as perceptually. We found substantial differences between noise-reduction algorithms from different hearing aids in terms of the resulting attenuation strength and spectral and dynamical behaviour. Perceptual evaluation with nor-
mal-hearing subjects showed little or no differences in terms of intelligibility and listening effort, but preference differed between noise-reduction systems as well as between subjects. Even within the homogeneous group of normal-hearing listeners, individuals differed from each other in whether their preference was driven by the absence of noise or by the naturalness of speech.

We recently started measurements with subjects from a heterogeneous group of hearing-impaired listeners, and we will present the first perceptual results from this group as well.

**A16**

**Optimizing the benefit of frequency compression in band-limited speech signals**

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Real-time frequency compression has become a common feature in hearing aids, but thus far no guidelines have been established on how to use frequency remapping to maximize intelligibility in situations where listeners have access to only a limited usable speech bandwidth. Two key variables that can be manipulated when fitting a frequency compression algorithm are the kneepoint, which is the lower bound of the frequency range where the compression is applied, and the compression ratio, which measures the amount of compression applied to the processed frequency region. In this study, a frequency compression algorithm based on a sinewave encoder was used to examine the impact that variations in kneepoint, compression ratio, and signal bandwidth had on consonant recognition scores for normal-hearing listeners. In the first experiment, the kneepoint was held fixed, and the compression ratio was adaptively adjusted to the point where listeners were just able to achieve a 75% correct score on the Modified Rhyme Test while listening in quiet. In the second experiment, the kneepoint and compression ratios were held fixed and the SNR relative to a speech-shaped noise was adjusted to track the 75% correct Speech Reception Threshold. In the third experiment, the kneepoint was fixed at 2500 Hz, and a factorial design based on articulation index was used to determine the relative contributions of six equal-intelligibility frequency bands, ranging from 2500-6100 Hz, with each of six different compression ratios. The results show three major findings: 1) When the kneepoint falls in the range from 1000 Hz to 2500 Hz, frequency compression is much more likely to impair the intelligibility of female voices than male voices, presumably because their vowel formants occur at higher frequencies for female talkers; 2) Although frequency compression generally leads to a decrease in performance for normal-hearing listeners, there are a small number of specific cases where frequency-compressed speech was found to produce better performance than an uncompressed speech signal that was low-pass filtered to the same effective bandwidth; and 3) some bands are more tolerant to frequency compression than other bands, which means that the optimal benefit from frequency compression can may only be achievable with a frequency compression system that allows more than one kneepoint and more than one compression ratio. The results also provide insights into the types of consonant confusions that can occur when frequency compression is applied.

**A17**

**Auditory localization in reverberant multisource environments by normal-hearing and hearing-impaired listeners**

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The ability to correctly localize sounds is important for general awareness of the auditory scene and communication in adverse acoustic conditions. However, most localization studies are performed in rather simple and artificial conditions. In particular, very few studies have considered localization in reverberant environments or in the presence of complex interferers, and no
studies have systematically investigated the effect of distance. In the present study, localization performance was measured as a function of both source-receiver distance and signal-to-noise ratio (SNR) using a virtual auditory environment. With increasing source-receiver distance the direct-to-reverberation energy ratio decreases and the auditory system increasingly relies on mechanisms related to the precedence effect (PE). Both aspects may be particularly problematic for hearing-impaired listeners.

The acoustics of a cafeteria were simulated with the ODEON software for a large number of target sources located in the horizontal plane at 1, 2, and 4 m distance to the listener. Signals were generated for a 3D array of 41 loudspeakers using the loudspeaker-based room auralization (LoRA) toolbox. Localization performance for the target word “two” was measured in eight normal-hearing (NH) and eleven hearing-impaired (HI) listeners with a moderate symmetric hearing loss in the simulated cafeteria with and without a multi-talker speech background. In order to minimize the influence of audibility, individual masked thresholds (MTs) were first measured for the target word in the multi-talker background using an adaptive tracking procedure. The SNRs used in the localization experiment were then adjusted relative to the individual MTs.

For the HI subjects the localization performance showed a very large inter-subject variation. This variation could not be explained by the audiograms, except that the very poor performers tended to have slightly worse hearing at low frequencies. Localization performance in quiet was found to be independent of distance for the NH subjects but for most HI listeners deteriorated with increasing distance. This suggests that the PE helps localization in realistic environments but is less effective in HI listeners. In the presence of background noise all NH and HI subjects showed a significant decrease in localization performance. Additionally the NH listeners now also showed a significant distance effect, which is in line with the observation that the PE is weakened in background noise.

The present study highlights the importance of considering more realistic scenarios in hearing research and may serve as a basis for developing spatial hearing tests with improved ecological validity.

A18

A comparison of NAL and DSL prescriptive methods for pediatric hearing aid fittings: Estimates of speech intelligibility, loudness, and safety

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Objective: This paper examined the impact of prescription choice on predicted speech intelligibility (SII), loudness, and safety for young children.

Design and study sample: Four studies were conducted. The first used a between-group design to compare SII and loudness for hearing-aid settings of 200 children who were fitted with NAL-NL1, or DSL v4.1 or DSL m[i/o]. The second used a within-group design to compare the SII and loudness of targets prescribed by the NAL-NL2 and DSL m[i/o] for 57 children. The third used a within-group design to compare the SII and loudness of real-ear prescriptions and real-life fitting of the DSL m[i/o] for 57 children. The fourth used a series of Modified Power Law-based calculations for predicting asymptotic temporary threshold shift caused by hearing aid use (Macrae, 1994). These calculations were employed to examine whether the loudness, or more accurately the intensity, of both the NAL-NL2 and DSL m[i/o] targets might be considered safe for hearing impaired listeners.

Results: The first study showed that on average, SII was higher for children fitted with DSL prescriptions than with NAL-NL1 at low input level, equivalent across prescription groups at medium input level, and higher for NAL-NL1 than DSL.
prescriptions at high input level. The DSL prescriptions resulted in greater loudness than NAL-NL1, across a wide range of input levels. The second study revealed that SII was higher for targets prescribed by DSL m[i/o] than by NAL-NL2 at low input level, but SII was higher for NAL-NL2 than for DSL m[i/o] at medium and high input levels. Estimated loudness was significantly greater for DSL m[i/o] than for NAL-NL2. The third study indicated no significant difference in SII or loudness between prescribed targets and hearing-aid settings for children fitted with DSL m[i/o]. The fourth study suggested that despite the loudness differences between the NAL-NL2 and DSL m[i/o], the prescribed gains across the frequency response from 500-4000 Hz at octave intervals are safe for typical listeners with up to a moderately-severe sensorineural hearing loss when listening to conversational level speech.

**Conclusion:** The choice of prescription has minimal effects on speech intelligibility but marked effects on loudness. The loudness resulting from applying the NAL-NL2 and DSL m[i/o] to hearing aids for amplification, however, appear to pose minimal risk for noise-induced hearing loss.

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**Cross-correlation based low-rate sound localizer for hearing support devices**

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In noisy environments, hearing support devices, such as digital hearing aids, need to have an ability to localize the sound direction with low computational burden to amplify a speech and to reduce a noise at the same time.

In this study, we developed a cross-correlation based sound localizer algorithm and evaluated its accuracy under the low sampling frequency condition in vitro using a KEMAR manikin in a non-reverberant room in the Samsung Medical Center.

Experimental results demonstrated that the developed algorithm showed a good performance with low sampling frequency situations, but it also showed a low stability by the head shadow effect. [This work was supported by grants from the Strategic Technology Development Program of Ministry of Knowledge Economy (10031764) and from Seoul R&BD Program (SS100022).]

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**Effects of venting and microphone directional-ity in wind and wind-induced vent effects**

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**Objectives:** Wind noise can be a nuisance to hearing aid users, especially for those who enjoy outdoor activities. With the advent of open fit hearing aids, people with lesser degrees of hearing loss and younger ages are fit with hearing aids. Many of them lead active lives. The purpose of this study was to examine effects of venting on wind noise in the ear canal for hearing aids with omnidirectional (OMNI) and directional (DIR) microphones.

**Design:** Two digital behind-the-ear hearing aids were programmed when they were worn on a Knowles Electronic Manikin for Acoustic Research. The hearing aid worn on the right ear was programmed to the omnidirectional microphone mode and the one on the left to the directional microphone mode. The hearing aids were adjusted to linear amplification with flat frequency response in an anechoic chamber. Wind noise samples were recorded at hearing aid outputs in a wind tunnel at velocities ranging from a gentle to a strong breeze when the hearing aids were coupled to #13 tubings (i.e., open vent), or conventional skeleton earmolds with no vent, a pressure vent, and a 3 mm vent. Polar and spectral characteristics of wind noise were analyzed off-line using Matlab programs.

**Results:** Wind noise levels in the ear canals were mostly predicted by vent-induced frequency re-
spone changes in the conventional earmold conditions for both hearing aids. The open vent condition, however, yielded the lowest levels which could not be entirely predicted by the change in frequency responses measured in a non-windy sound field. This indicated that wind permitted an additional amount of sound reduction in the ear canal that could not be explained by known vent effects.

**Conclusion:** Open fit hearing aids yielded lower noise levels at the ear drum location than conventional behind-the-ear hearing aids in the conditions where testing was carried out. They appear to be a more desirable option than regular behind-the-ear hearing aids for use in wind.

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**A21**

**Sound-field evaluation of commercially-available frequency-lowering hearing aids**

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Frequency-lowering (FL) hearing aid algorithms are designed to move high frequency sounds into lower frequency regions for improved audibility. There are two major approaches to accomplishing this in commercially-available hearing aids. Frequency transposition shifts high frequency sound to a lower frequency region, where it overlaps the existing lower frequency information. With frequency compression, a high frequency band is compressed with a fixed lower-frequency edge and a lowered upper-frequency edge so that its overall bandwidth is reduced.

The importance of clinical verification of FL hearing aid fittings has been stressed by Bentler (2010) and others. Some real-ear measurement systems have tests available to specifically assess this technology. For example, isolated high-frequency 1/3 octave bands may be used to visualize the amount of frequency shift and the overall sensation level of the lowered components of a speech signal. However, these methods do not fully describe the action of the FL algorithm, and clinicians still have questions about these algorithms and how to evaluate them.

In this study, we evaluated the three commercially-available FL hearing aids on a KEMAR manikin in soundfield. The gain and frequency shifting behavior of each device was measured by presenting 1/12th octave bands of random noise at four different signal levels (50, 60, 70 or 80 dB SPL) and 13 different center frequencies (1/3rd octave intervals from 500 to 10,000 Hz). In each case, the overall frequency shift of the device was determined by analyzing the peak spectral envelope to find the center frequency of the 1/12th octave band with the highest output and comparing this frequency to the center frequency of the input signal, and the gain was determined by comparing the overall level of the output signal in this band to the signal level in the unprocessed input signal. These measurements were made with the FL algorithm activated and de-activated for two different audiometric configurations: a steeply-sloping high frequency loss, and a flat hearing loss configuration. Manufacturer’s first-fit software was used to program the hearing aids, and fine-tuning adjustments of gain and FL parameters were made based on real ear measurements on KEMAR’s ear.

Results are displayed as arrow plots that show the relationship between the center frequency and gain for each data point with the FL turned on, and the center frequency and gain for each data point with the FL turned off. This method of plotting the response of the hearing aid makes it much easier to visualize the complex interaction between the frequency lowering behavior of the hearing aid and its compressive gain in each frequency band than do current analysis techniques.

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**A22**

**Quantifying the acoustic features of recorded music processed through hearing aids**

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² GN ReSound

Research in hearing-aid design has primarily emphasized speech perception, with less attention given to music. A lack of consideration of the differences between speech and music (e.g., differences in dynamic range, crest factor, and important frequencies) may lead to dissatisfaction for hearing-aid users listening to music. Most
current hearing aids utilize wide dynamic-range compression (WDRC), with parameters based on speech characteristics. However, the effectiveness of WDRC applied to music may depend on the characteristics of the music input signal. For example, recorded music often undergoes dynamic-range compression as part of the recording process, and our recent work with normal-hearing listeners has shown that music quality judgments vary with the amount of music-industry compression applied. Therefore, hearing-aid users who listen to recorded music may experience the effects of music-industry compression, hearing-aid WDRC, and combined music-industry and hearing-aid compression. In order to determine the best signal processing strategies for music, an understanding of the music signal is necessary at both the input and output of the hearing aid, including the interactions that may take place among multiple forms of processing. To this end, the current study will characterize the effects of hearing-aid signal processing on the acoustic properties of recorded music through several objective analyses. The analyses will quantify changes to the amplitude and frequency characteristics of the music signals with parametric variations of music-industry compression, hearing-aid compression, and combined compression. The analyses will include: time-amplitude waveforms, wideband amplitude histograms, dynamic-range plots, and changes in the time-frequency envelope modulation. Amplitude features of the music signals will be analyzed using time-amplitude waveforms and wideband amplitude histograms, illustrating changes to the envelope and crest factor of the signals with different compression parameters. Dynamic-range plots will show level distributions within analysis bands equal to human auditory filters, including the effects of hearing loss, providing a perceptually relevant analysis. Finally, the Hearing Aid Speech Quality Index (HASQI) will measure the amount of linear and nonlinear envelope modification due to different combinations of compression. The results will provide quantitative descriptions of recorded music signals at the input and output of a hearing aid and will be related to perceptual quality ratings from listeners with hearing loss. This study will have implications for signal processing development by offering a better understanding of acoust-
ured using the spectral ripple density and depth discrimination tasks. Predicted auditory filter bandwidths were obtained using the spectral ripple thresholds and model equation from Supin (1994).

Results revealed that ripple predicted auditory filter bandwidths were narrower than notched-noise measured auditory filter bandwidths, especially at high levels. For all listeners, the effect of level was much exaggerated with the narrowband measured filter bandwidths compared to the ripple predicted filter bandwidths. Auditory filters were up to four times wider with notched-noise compared to ripple measures, and the difference between measured and predicted filters was greatest for the higher level test signal. Taken together, these results suggest the effect of level may be exaggerated with the notched-noise masking task because it is affected by the broadened filter tip and tail. However, spectral ripple discrimination may be dependent on filter tip resolution and thus show a much smaller level effect, which also correlates with perceptual measures of speech recognition at high levels. [Work supported by NIH R01-DC008306]

**A24**

**Hearing aid use and perceived benefit of the hearing aid in bimodal listeners (cochlear implant and hearing aid)**

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With changes in audiometric criteria for cochlear implantation, there is an expanding population of unilateral cochlear implant users with usable residual hearing in the contralateral ear. As part of our research on bimodal hearing, we administer a questionnaire to subjects to obtain information about the decision to use a hearing aid, the pattern of hearing aid use, and the perceived benefit of using the hearing aid in addition to the cochlear implant. The information from the questionnaire will be used in conjunction with information about the hearing loss, hearing history, and objective measures of bimodal benefit to determine characteristics of long-term bimodal listeners. Preliminary data will be presented. [Research supported by NIDCD (1R01DC011329).]

**A25**

**Unilateral and bilateral hearing aids, spatial release from masking and auditory acclimatization**

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Spatial release from masking (SRM) was tested at first fitting and after ~ 12 weeks hearing aid use for unilateral (n=24) and bilateral (n=17) adult hearing aid users. A control group of experienced hearing aid users (n=15) completed testing over a similar time frame. The main research aims were i) to examine auditory acclimatization effects on SRM performance for unilateral and bilateral hearing aid users, ii) to examine whether hearing aid use, level of hearing loss, age or cognitive ability mediate acclimatization, and iii) to compare and contrast the outcome of unilateral versus bilateral aiding on SRM. Hearing aid users were tested with and without hearing aids, with SRM calculated as the 50% speech recognition threshold advantage when maskers and target are spatially separated at +/- 90° azimuth to the listener compared to a co-located condition. The conclusions were i) on average there was no specific improvement over time in aided listening conditions, ii) there was large variability in outcome, with greater improvement associated with better cognitive ability and younger age, but not associated with hearing aid use, iii) overall, bilateral aids facilitated better SRM performance than unilateral aids, with audiometric factors the primary determinant of performance.

**A26**

**Hearing aid gain characteristics during aided cortical auditory evoked potentials testing**

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*University of Western Ontario*
Measurement of cortical auditory evoked potentials (CAEPs) has gained interest recently, for the evaluation of infants who use hearing aids (e.g., Golding et al., 2007). CAEPs are time-locked transient evoked potentials generated from the auditory cortices in response to external sounds. In an aided CAEP procedure, CAEPs are elicited by the stimuli, while amplified and processed by hearing aids. The interactions between hearing aid processing and the CAEP stimulus have not been fully evaluated. Therefore, it is possible that hearing aid signal processing could alter the CAEP stimulus in important and unexpected ways. This study will apply a wide range of hearing aid signal processing options to typical CAEP stimuli, and evaluate the effects acoustically.

Speech CAEPs are elicited by isolated speech segments, typically shorter than 100 - 200 ms, played every one to two seconds. These stimuli differ from naturally spoken speech because of this brief duration and frequent silent intervals. There is a possibility that the hearing aids may not react similarly to the same speech sound presented both in running speech and in isolation. The purpose of this study is to investigate the impacts of modern hearing aid signal processing on speech segments appropriate for elicitation of CAEPs. For this study, we have created a matched stimulus set of phonemes either embedded in running speech (gold standard) or in isolation (as appropriate for CAEP). We will record aided signals across a wide range of signal processors (compression, expansion, digital noise reduction, feedback cancellation) and audiograms. Measures for analysis will include output sound pressure level, and gain per stimulus phoneme, with the gold standard used as reference. Across these measures, comparison between the gold standard and CAEP stimulus types will reveal whether processing changes for CAEP stimuli are representative of hearing aid function for running speech. This finding will provide important new information about whether aided CAEP measurements can provide assessment of aided function during typical speech.

Reference:
A28

**Intervention for restricted dynamic range and reduced sound tolerance**

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Hyperacusis and related sound tolerance problems (diagnosed or undiagnosed) can be primary reasons why hearing-impaired persons reject hearing aids. Tinnitus Retraining Therapy (TRT), which includes a protocol for treating hyperacusis, has been shown to induce a significant secondary treatment effect, namely, elevated Loudness Discomfort Levels (LDLs) and, in turn, an expanded dynamic range for loudness over the course of treatment. TRT uses specialized counseling and sound therapy, typically achieved with continuous soft “seashell-like” sound from bilateral noise generators (NGs), to foster treatment success. To date, there have been no controlled studies to assess the efficacy of TRT as an intervention for reduced sound tolerance. We therefore designed and conducted a randomized, double-blind, placebo-controlled clinical trial to assess the efficacy of TRT as an intervention for reduced sound tolerance among hearing-aid eligible persons with hyperacusis and/or restricted dynamic ranges. The 2 x 2 study design called for eligible hearing-impaired subjects to be assigned randomly to one of four treatment groups: 1) counseling and sound therapy implemented with NGs, 2) counseling and sound therapy implemented with placebo NGs, 3) NGs without counseling, and 4) placebo NGs without counseling. This study design enabled us to evaluate the efficacies of the individual treatment components, the specialized counseling and sound therapy protocols, in comparison with full TRT and with a neutral treatment control. The subjects were evaluated on a variety of audiometric tests, including LDLs, the Contour Test for Loudness for tones and speech, and word recognition measured at each session’s “comfortable” and “loud” levels. The results for 36 subjects, who completed treatment and follow-up testing, revealed those subjects receiving the full TRT treatment achieved significantly greater treatment benefit (7-dB increase in loudness judgments after six months) than was achieved by any of the other treatment groups. These positive treatment effects were significant and consistent across frequency (500, 2000, and 4000 Hz) for Contour categories corresponding to judgments of loud but comfortable, loud, and uncomfortably loud. The average treatment effects were not significant for any of the other groups. These findings suggest that TRT principles can be applied successfully to achieve a clinical intervention to enhance sound tolerance and to expand the dynamic range for loudness among hyperacusic hearing-impaired individuals. Case examples of successful treatments will be presented to highlight enhanced word recognition at increased presentation levels and, in turn, increased benefit and satisfaction from hearing aids post treatment. [Supported by NIH Award R01DC04678]

A29

**SCORE bimal: a new sound processing strategy and fitting method for combined hearing aid and cochlear implant stimulation**

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Due to changing implantation criteria, many recently implanted cochlear implant users have residual hearing in the non-implanted ear, which is stimulated using a hearing aid. This combination is called bimal stimulation. Currently devices are used that were developed separately and are often fitted separately. This results in very different growth of loudness with level in the two ears, potentially leading to decreased wearing comfort and suboptimal perception of interaural loudness differences. To address this, we developed a loudness equalization strategy and fitting method named ‘SCORE bimal’. The strategy aims to equalize loudness growth for both modalities using existing models of loudness for both acoustic and electric stimulation, and is suitable for implementation in wearable devices.
We performed loudness balancing experiments with six bimodal listeners to validate the strategy. In a first set of experiments, monaural loudness balancing was done for four harmonic complexes of different bandwidths, ranging from 200 Hz to 1000 Hz. Both the electric and acoustic loudness model predicted the psychophysical data well. In a second set of experiments, binaural balancing was done for the same stimuli. It was found that on average SCORE significantly improved binaural balance by 59 percent.

Speech perception performance in quiet and in noise, and sound localization ability of the same six bimodal listeners were measured with and without application of SCORE. Speech perception in quiet was measured with only acoustic, only electric, or bimodal stimulation, at soft and normal conversational levels. There was a significant improvement in phoneme score with application of SCORE. Speech perception in noise was measured with a steady-state noise, a fluctuating noise, and a competing talker, at conversational levels for bimodal stimulation. There was no significant effect of application of SCORE.

As SCORE was found not to interfere with speech perception and to improve loudness perception, it seems suitable for implementation in clinical devices. We believe that this is the first sound processing strategy and fitting method developed specifically for bimodal stimulation. [Tom Francart was sponsored by a Post-Doctoral Fellowship of the Fund for Scientific Research of the Flemish Government and a Marie Curie International Outgoing Fellowship of the European Commission, grant agreement number PIOF-GA-2009-252730. The Bionics Institute acknowledges the support it receives from the Victorian Government through its Operational Infrastructure Support Program.]

A30
Understanding speech-in-noise perception across the adult lifespan: Contributing factors beyond audibility

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The aims of this still-ongoing cross-sectional study are (i) to establish whether or not speech perception in noise declines as a function of age, even when peripheral hearing sensitivity (i.e., audibility) remains in the normal range, and (ii) to identify, using correlational analyses, potential contributing factors such as changes in temporal fine-structure (TFS) processing and cognitive functioning.

A total of 136 volunteers aged 18-90 yrs were recruited, with a least 15 per age decade up to 80 yrs. To participate in the study, volunteers had to have unilaterally (UNH) or bilaterally normal hearing (BNH), defined as audiometric thresholds ≤20 dB HL between 125 and 4000 Hz. In a subset of these volunteers, outer-hair-cell function was assessed using otoacoustic emissions. Volunteers aged ≥60 yrs had to score ≥28/30 on the Mini Mental State Examination to exclude cases of gross neurological dysfunction. The average number of years of formal education was very similar across age decades.

Sensitivity to TFS was measured using two psychophysical tests. In the first, UNH and BNH participants discriminated a monaurally presented harmonic tone complex from an inharmonic tone complex, obtained by shifting all components of the first complex upwards in frequency. Fundamental frequencies (\(F_0\)) of 91 and 182 Hz were used. The spectral envelope was fixed by applying a filter with a bandwidth of 1\(F_0\) and centered on the 11\(^{th}\) harmonic (corresponding to 1001 and 2002 Hz, respectively). In the second test, BNH participants discriminated a diotic 500- or 850-Hz pure tone from the same pure tone with a phase difference between the two ears.

Eight cognitive tests were administered to assess different cognitive functions such as processing speed, executive function, short-term/working memory, verbal fluency, and reasoning.

The results show that: (i) there is large inter-individual variability in monaural and binaural TFS processing but average sensitivity declines with age; (ii) this age-dependent change already becomes significant in early middle age; (iii) hearing sensitivity at the test frequency or in the high frequency range (>4000 Hz) does not pre-
Speech identification for consonants and vowels in quiet and in “steady” and modulated noise is currently being measured and will be compared to TFS sensitivity and cognitive scores. [Work supported by the Oticon Foundation (CF) and the MRC UK (TB, BCJM).]

A31
The relative importance of ITDs & ILDs to spatial processing
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Previous research has demonstrated that hearing-impaired people experience deficits in spatial processing ability. Spatial processing ability is defined as the capacity to focus on target sounds coming from one direction while suppressing sounds from other directions. In order to understand the processes that are disordered in hearing-impaired people, and therefore how best to design technology to compensate for these deficits, it is necessary to understand how the majority of normal hearers segregate speech streams based on their spatial location. It has long been assumed that both interaural time differences (ITDs) and interaural level differences (ILDs) are used in spatial processing. However, little is known about the relative importance of these cues. This study investigated the proportional contribution of ITDs and ILDs to spatial processing in a group of normal-hearing adults. Twelve participants aged between 24 and 53 years were tested on three modified versions of the Listening in Spatialized Noise - Sentences test (LiSN-S). One version of the LiSN-S provided participants with access to both ITDs and ILDs, one version only contained ILDs and the other only contained ITDs. Significantly poorer spatial processing ability was found in the ITD-only condition when compared to the ILD-only condition or the condition with both ITDs and ILDs (both p < 0.01). This suggests that ILDs provide the dominant cues used by normal-hearing adults to spatially segregate speech even when there are no overall differences between the signal-to-noise ratios at the two ears.

A32
Predicting masking release in hearing-impaired listeners using measures of cochlear compression and temporal resolution
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Hearing-impaired (HI) listeners often show poorer performance on psychoacoustic and speech-perception tasks than do normal-hearing (NH) listeners. For example, listeners with cochlear hearing loss often show less benefit, or masking release (MR), than do listeners with NH when a steady-state masker is replaced by a temporally fluctuating one. This had been found even when overall audibility is equated by, for instance, adding noise that equalizes the detectability thresholds of NH and HI listeners. It is possible that the loss of MR in HI listeners is due in part to reduced cochlear compression. This study tests whether changes in cochlear compression can predict the degree of MR loss. Behavioral estimates of compression, using temporal masking curves (TMCs), were compared with MR for various stimuli in hearing-impaired (HI) individuals and age-matched, noise-masked normal-hearing (NMNH) listeners. Compression estimates were made at 500, 1500, and 4000 Hz. Masking release for speech in a background of noise to equalize thresholds was measured by comparing performance for band-limited (500-4000 Hz) IEEE sentences in a steady-state, speech-shaped noise to performance in a 10-Hz square-wave gated noise. Pure-tone masking period patterns, with the threshold equalizing noise, were measured in a 10-Hz square-wave-gated noise for the same 500-, 1500-, and 4000-Hz tones used for the TMC measurement. In addition, an estimate of temporal resolution was calculated using the slope of the off-frequency curve from the TMC measurements. No strong relationship was found between estimates of cochlear compression and MR for either speech or pure tones. There was a non-significant correlational trend between temporal resolution estimates and MR for speech overall, and NH had significantly steeper recov-
A new probe noise approach for acoustic feedback cancellation in hearing aids

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Acoustic feedback is a big challenge in hearing aids. If not appropriately treated, the feedback limits the maximum possible amplification and may lead to significant sound distortions. In a state-of-the-art hearing aid, an acoustic feedback cancellation (AFC) system is used to compensate the artifacts caused by acoustic feedback. The AFC is usually carried out by adaptively estimating the feedback paths (from receiver to microphones) and using these estimates to reduce the artifacts introduced by the acoustic feedback. One major limitation in the widely used adaptive filter technique for AFC systems is the biased adaptive filter estimation problem, especially when tonal signals such as music and alarm tones enter the hearing aid microphones. The consequences of this biased estimation might be significant sound distortion or even worse, howling.

In principle, unbiased adaptive filter estimation can be achieved by adding a probe noise signal to the receiver signal and basing the estimation on the probe noise signal. However, the traditional probe noise approach requires a high-level probe noise signal, which is clearly audible and annoying for the hearing aid user. Hence, this high probe noise level makes the traditional probe noise approach less useful in hearing aid applications.

We present a new probe noise approach which utilizes a low-level probe noise signal, which is inaudible in the presence of the receiver signal even for people with normal hearing. The probe noise signal is generated based on the short-time spectral envelope of receiver signal. Furthermore, this probe noise signal is generated with a specific characteristic so that it can facilitate unbiased adaptive filter estimation with fast tracking of feedback path variations/changes despite its low signal level, which is not possible with the traditional probe noise approach. We show simulation results of a challenging situation for AFC systems, where the acoustic feedback path changes momentarily while the hearing aid user is listening to music. The traditional AFC system fails completely with significant sound distortions and howling as consequences, whereas the new probe noise based AFC approach is able to remove feedback artifacts caused by the feedback path change in no more than a few hundred milliseconds.

Perceptual evaluation of a binaural beamforming algorithm

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Starkey Laboratories, Inc.

Improving speech understanding in noise for listeners with hearing impairment has long been a goal of various signal processing algorithms. Such an improvement can be achieved by using a microphone array within each hearing aid (i.e., bilateral directional hearing aids) or by forming a microphone array using two hearing aids (i.e., binaural directional hearing aids). The binaural advantage for speech recognition in reverberant environments and a background of noise is very well documented for both listeners with and without hearing loss (Bronkhorst & Plomp, 1992; Yost, 1997). The magnitude of this binaural advantage (~3 dB) remains unchanged with bilateral omnidirectional or bilateral directional hearing aids (Ricketts, 2000c). Binaural directional hearing aids may be able to provide additional benefits for speech recognition. However, such benefits may disappear if binaural cues such as interaural time difference (ITD) and interaural
level difference (ILD) are not properly preserved in the binaural beamforming design.

The goal of this study was to investigate the perceptual performance of a binaural directional hearing aid as compared to bilateral hearing aids. The following conditions were assessed for speech intelligibility, localization and spatial release from masking performance:

a. Omnidirectional
b. Bilateral Directional
c. Binaural Directional with partially preserved binaural cues
d. Binaural Directional with no preservation of binaural cues

For speech intelligibility, speech recognition threshold (SRT) was measured using the hearing in noise test (HINT). For localization, subjects’ localization performance was measured using Hartmann’s method (1983). For spatial release from masking, Summerfield’s method (1998) was used with speech and masker in different combination of locations.

The results will be discussed with the implications for binaural beamforming design trade-offs.

A beamformer post-filter for noise reduction in cochlear implants

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Many hearing aid and cochlear implant sound processors aim to reduce noise in order to improve speech perception. An adaptive beamformer is generally viewed as the most successful technique, superior to single microphone methods due to the ability to filter sounds based on their direction of arrival. However, effectiveness declines quickly as the noise field becomes spatially diffuse. An adaptive beamformer cannot adjust to cancel multiple noise sources simultaneously if they are spatially separated.

To address this shortcoming, a beamformer post-filter was evaluated that enhances the output of the main beamformer by reducing energy in frequency bands that are considered noisy. Signal-to-noise ratio (SNR) is estimated by analysing signals from front-facing and rear-facing fixed directional microphones, and the SNR is used to control a parametric Weiner filter with adjustable gain. The result is a spatial post-filter that removes sounds from behind the listener and only passes sounds from the front.

Speech reception thresholds (SRTs) were measured with a group of 12 cochlear implant recipients by presenting sentences in 4-talker babble. A difficult noise environment was used for testing where the four talkers were spatially separated and presented from random speaker locations in the rear hemi-field. The noise locations were changed after each sentence while target speech was presented from the front. The post-filter was compared against three beamformers commercially available in the Nucleus 5 CP810 sound processor from Cochlear Ltd: a moderately directional beamformer called Standard (Everyday program), a highly directional beamformer called Zoom (Noise program), and an adaptive beamformer called Beam (Focus program). Four gain settings for the post-filter were evaluated. An analysis of speech distortion and noise reduction formulated as binary mask errors was conducted to investigate how the gain setting affected the number of errors.

Results show significant speech perception improvements using the post-filter compared to the commercially available beamformers. The group mean SRT improvement was 9.6 dB compared to Standard, 5.6 dB compared to Zoom, and 4.6 dB compared to Beam (p < 0.001). The post-filter gain setting changed aggressiveness of noise reduction by trading-off speech distortion and noise reduction errors. This led to changes in intelligibility, and the optimal setting was found for this noisy environment.

The benefit due to the post-filter is largely attributed to the ability to cancel multiple noise sources simultaneously, not possible with a traditional beamformer. This research suggests that cochlear implant users (and possibly hearing aid users) are likely to obtain significant advantage from the post-filter algorithm in their everyday lives. It represents a substantial leap forward in noise reduction performance, similar to the leap forward observed when a second microphone...
was first introduced. [This research was financially supported in part by the HEARing CRC established and supported under the Australian Government’s Cooperative Research Centre’s program. The Bionics Institute acknowledges the support it receives from the Victorian Government through its Operational Infrastructure Support Program.]

### A36

**Input noise direction detection method using beamformer**

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A beamformer algorithm enhances a signal of target speech direction by reducing a signal of input noise directions entered from different directions. Fixed Directional Microphone (FDM) is a simple beamformer. It is composed of subtraction two input signal from Omni-Microphone. And it reduces a noise signal of single fixed direction. But when a input noise direction is changed, FDM doesn’t reduce a noise signal of a changed input noise direction any more. Therefore, a beamformer need to know an input noise direction to reduce exactly a signal of input noise directions. An Adaptive Direction Microphone (ADM) using adaptive filtering is a conventional method to detect an input noise direction. But it is much higher computational complexity and lower reduction performance than FDM.

In this study, we proposed a concept of novel input noise direction detection method using a multiple FDM (m-FDM) and evaluated the new method by a simulation on Personal Computer.

A basic assumption of the proposed method is that an output SNR of beamformer is highest when null direction of beamformer is aligned with an input noise direction.

The proposed method is composed of five FDMs (m-FDM). Each FDM has a different null direction each other. A null direction of fist FDM is azimuth 45°, second is 90°, third is 120°, fourth is 150° and fifth is 180°. After an input signal is applied by five FDMs simultaneously, the output SNR of them is compared. A FDM of highest SNR is estimated to input noise direction approximately. The proposed method used a Voice Activity Detection for noise update at noisy only period.

The proposed method is implemented and evaluated by MathWorks Simulink. Two objective evaluation environments also are implemented by Simulink. First evaluation is when an input noise direction is fixed. Second is when an input noise direction is changed.

At the result of first evaluation, a performance of the proposed method was similar with a FDM whose null direction was aligned with a noise direction.

At the result of second evaluation, the proposed method was adapted by an input noise direction. In this paper, the direction detection of input noise signal was presented. And it was evaluated to compare with a FDM. Although it is simple method, it is robust detection at which a direction of the input noise is change. Next, we will study to compare with an Adaptive Directional Microphone beamformer and to evaluate a subjective test. [This work was supported by grants from the Strategic Technology Development Program of Ministry of Knowledge Economy (10031764) and from Seoul R&BD Program (SS100022)]

### A37

**The potential of the ideal Wiener filter and ideal binary mask for speech intelligibility improvement in adverse listening conditions**

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Hearing impaired people suffer from the inability to understand speech in adverse listening conditions like noisy or reverberant conditions. State-of-the-art single channel noise reduction algorithm like the Wiener filter (WF) and binary-
mask (BM) enhance the signal by applying a signal-to-noise ratio (SNR) depending gain to the noisy mixture that is continuous or binary, respectively. In this study, we investigated the potential for speech intelligibility improvement of soft-mask approaches and binary mask approaches represented by the ideal Wiener filter (IWF) and the ideal binary mask (IBM), respectively. Besides the study on speech intelligibility improvement, a quality rating for both noise reduction strategies was conducted.

The masks that were applied to the noisy mixture were derived with a frequency resolution according to the Bark scale with its 25 critical bands. The frequency resolution was chosen to be relevant for usage in hearing aids and cochlear implants. First, a sentence recognition task with the Dutch VU sentences was conducted with four normal hearing (NH) listeners at SNRs from -35 dB to 10 dB in multitalker babble noise and in an interfering talker condition. In a second sentence recognition task with six NH listeners, the influence of estimation errors in the mask derivation on speech intelligibility was simulated by adding a white noise to the noise power estimate. A two-stage pairwise preference rating was conducted with eight NH listeners.

The results of the sentence recognition tasks showed that even at the lowest SNR the IWF approach restored perfect intelligibility in both noisy scenarios. A decrease in speech intelligibility was observed for the IBM at SNRs lower than -15 dB. In the second sentence recognition task with corrupted estimates, the IWF approach was more robust to estimation errors. The quality rating showed a clear preference for the soft-mask IWF approach in comparison to the IBM approach.

In this study, we showed that the soft-mask approach IWF outperforms the binary-mask approach IBM in terms of speech intelligibility, robustness to estimation errors and speech quality when using a Bark scale frequency resolution. The outcomes suggest that for the application in hearing instruments, soft-mask approaches should be used instead of binary mask approaches especially due to its robustness to estimation errors.

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**Fabrication of round window driving transducer for adjustment of frequency characteristics using a spiral spring**

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According to the medical advance and rapid industrialization, the number of people with hearing difficulty is also increasing. So, various hearing aids are developed to overcome their hearing loss. There are available hearing aids, such as air conduction hearing aid, cochlea implant, implantable middle ear hearing aid and so on. But air conduction hearing aid is inconvenience caused by howling or performance problem, and ossicle chain driving type implantable middle ear hearing aid has some weak point due to problem of possible necrobiosis of coupling spot along incus long process.

In recent years, in order to improve these shortcomings round window (RW) driving hearing aid has been paying attention. The RW driving method can be maintained the advantage of ossicle chain driving method. In addition, these methods can be applied patient who has middle ear disease.

In general, the development stage of the implantable hearing aid transducers must be made to fit the design criteria. Development criteria of ossicle chain driving transducer are specified in ASTM (American society for testing and materials) F2504. But the RW driving transducer is still at a developmental stage and its evaluation method of performance is not known. And, there are no criteria of frequency characteristics of RW driving transducer. So, adjust vibrant frequency characteristics of the transducer design is required.

In this paper, we designed round window driving electromagnetic transducer for adjustment of frequency characteristics using a spiral spring. The transducer was designed on the basis of RW vibrator. The transducer is consists of a central coil
in one direction and two end coils in the opposite direction. Two magnets attached end to end in opposing directions are positioned inside the coils. And then a spiral spring was placed inside the transducer to adjust frequency characteristics of RW vibrator.

The frequency characteristics of the proposed transducer compared to that of the RW driving vibrator. The RW driving vibrator is generated resonance frequency at 1.6 kHz. And the proposed transducer is generated resonance frequency at 2.8 kHz.

By comparison between two frequency characteristics, it was confirmed that the proposed transducer can be adjusted frequency characteristics.
POSTERS FOR SESSION B SHOULD BE PUT UP BY 8:00 AM FRIDAY, AUGUST 10, AND TAKEN DOWN AFTER 10:00 PM FRIDAY, AUGUST 10 OR BEFORE 7:00 AM SATURDAY, AUGUST 11. PRESENTERS SHOULD BE AT THEIR POSTERS FROM 9:55 AM – 11:10 AM; 8:30 PM - 10:00 PM.

POSTER SESSION B
Friday 9:55 AM – 11:10 AM

B1
A qualitative study of acclimatization to hearing aids by adults

Piers Dawes¹, Michael Maslin¹, Kevin J Munro¹ and Sridhar Kalluri²
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The aim of this study was to describe the process of becoming accustomed to using hearing aids and factors which facilitate this process from the perspective of adult hearing aid users. A focus group was carried out in the north of England with adult hearing aid users (n=6). Hearing aid users owned hearing aids for less than two years and had used them at least once in the past three months. A topic guide and discussion exercises were used to elicit participant’s views on becoming accustomed to hearing aid use. Focus group discussion was audio recorded, transcribed verbatim and analysed according to qualitative content. Participants described becoming accustomed to hearing aids as a multi-factorial process which included adjusting to altered sensory input, managing practical matters such as cleaning and maintenance, discovering benefits and limitations of hearing aid use and managing the psychological impact of hearing aid use, such as on self-image. Factors that support this process included acceptance of hearing loss, persistence and consistent hearing aid use, support from friends, family and clinicians, and provision of information about hearing aids. Conclusions were that becoming accustomed to hearing aids is a challenging multi-factorial process with both psychological and practical difficulties besides demands of adjusting to hearing aid input. Addressing these diverse challenges may offer novel ways of supporting new hearing aid users.

B2
A statistical modeling approach to characterizing sound localization on a horizontal plane

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Hearing loss and hearing instruments are both potential sources of error in human localization performance. The study of localization in the horizontal plane usually involves measuring either the detection of differences in location (the “minimum audible angle”; MAA) or the absolute localization ability of human listeners. While the statistical analysis of the MAA experiment is well established, a variety of methods exist for establishing performance in the absolute localization paradigm, some of which have substantial drawbacks from a statistical perspective. The approach described here stipulates that the localization function is a probability density satisfying several key assumptions. The first is that the distribution must be circular (0 degrees equals 360 degrees) and the second is that it is bimodal, due to the possibility of front-back confusions. The proposed model of the localization distribution is a mixture of two Wrapped Cauchy distributions, one concentrated in front of the listener and the other concentrated behind the listener. Using data collected from listeners who vary in age and hearing impairment, the analyses to be presented will demonstrate that this model of listener performance allows one to jointly estimate the front-back confusion rate concurrently with the precision of the localization performance. The advantage of this approach is that it can be used to independently characterize these two key aspects of listener performance for listeners who vary substantially on both. In addition to capturing the effects of listener characteristics, such as age and hearing impairment, the model also provides a formal basis for testing the impact of signal characteristics, such as signal position and signal type, on localization performance. The equations will be presented both mathematically and in a
format suitable for implementation in Matlab, along with several examples demonstrating the efficacy of the model.

**B3**

**Multi-channel sound field reproduction for hearing aid evaluation**

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Spatially complex acoustic scenes play an increasing role in hearing aid evaluation. In most spatial evaluation setups, a set of loudspeakers positioned around the listener is used. Depending on the number of loudspeakers, this setup allows for spatial sound reproduction techniques of different complexity. In this study, vector-based amplitude panning, higher-order ambisonics and wave-field synthesis were tested in a loudspeaker system with moderate complexity (8-24 speakers). The accuracy of sound-field reproduction was assessed by predicting spatial perception from a binaural, auditory-model based direction-of-arrival estimator and by measuring the performance of a standard adaptive directional microphone (ADM) algorithm. Data show that the three methods can reasonably well reproduce the spatial perceptual information and maintain the functioning of the ADM algorithm. We conclude that the reproduction methods, when applied properly, can be used to generate static spatial acoustic scenes for hearing aid evaluation. A proper simulation of moving sound sources, however, is more difficult to achieve and we discuss and demonstrate the constraints and limitations of the different reproduction methods on the simulation of sources moving at different speed. We also demonstrate the application of the setup to investigate the subjects’ gestures in response to changes in the source configuration that might be used to interactively control hearing aid function.

**B4**

**Measuring listening effort with digits in noise**

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**Objective:** The purpose of this study is to determine if it is possible to measure listening effort for intelligible speech with triplets of spoken digits.

**Design:** We added various amounts of stationary noise to digit triplets and measured its influence on the reaction time for two tasks. In the first task, participants had to quickly identify the last digit of a triplet. In the second task they had to quickly add the first and the last digit.

**Study Sample:** Twelve normal-hearing participants.

**Results:** Response time increases with lower (i.e., worse) signal to noise ratios for both tasks. The response time on the arithmetic is more influenced by the noise than the response time on the identification task, but the arithmetic task has a higher variance.

**Conclusions:** Listening effort can be measured with digit triplets at signal in noise ratios at which speech is highly intelligible. The optimal task may depend on the signal to noise ratio that is of interest. The potential audiological application (evaluating hearing aids and their signal processing) has yet to be studied. If positive, this listening effort test would fill a gap in the evaluation of assistive hearing devices when listening effort plays a role.

**B5**

**An automated response detection procedure for human frequency following response elicited by voice pitch**

Jiong Hu and Fuh-Cherng Jeng

Ohio University

Frequency following response (FFR) has received a fair amount of attention in recent years.
Researchers have probed different aspects of the response and yet, in most of those studies, the presence of such a response is based on subjective interpretation of the examiner. Aside from a sole recent report examining two algorithms for detecting such responses, there has been very limited number of attempts to further develop a sound automated procedure for FFR. The purpose of this study is to (1) develop an automated procedure derived from response detection algorithms that are based upon the statistic properties of the temporal and spectral energy distributions in the recorded waveforms and (2) explore the effectiveness, accuracy and efficiency of the automated procedure and compare it with those obtained from conventional algorithms and human judgments.

**B6**

**Investigation of the optimal dose and duration for auditory training in older adults**

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This presentation provides an interim report of an ongoing evaluation of the dosage and duration for an innovative auditory-training program designed to improve the speech-recognition performance of older adults listening in noise. The training program has been demonstrated previously to be efficacious in prior studies. Here, the focus is on the best dosage—90-minute sessions either two or three times per week (2x/wk and 3x/wk, respectively)—and the best duration (5, 10, or 15 weeks for 3x/wk and 8, 16 or 24 weeks for 2x/wk) to use with this training program. At study completion, 60 older adults with impaired hearing will have participated in this study with 20 older adults assigned randomly to each of three groups (control—no training, 2x/wk, 3x/wk). At this stage, 51 subjects (27 males, 24 females) ranging in age from 61-79 years have been enrolled and are at various stages of study completion. The study protocol has the following general features: (1) pre-training open-set recognition performance is measured in background noise for a variety of materials (words, phrases, sentences) spoken by a variety of talkers, including some materials and talkers not used in training; (2) cycle 1 of closed-set speech-identification training in noise is conducted for 5 weeks for the 3x/wk group or 8 weeks for the 2x/wk group with a mix of materials and talkers used during training; (3) post-training evaluation of open-set speech-recognition performance is completed using materials equivalent to those used at the initial pre-training assessment; and (4) steps 2 and 3 above are repeated for cycles 2 and 3. Currently, depending on the group, 5-9 subjects have completed the entire study protocol, 8-12 have completed the first two training cycles, and 15-19 have completed the first training cycle. Analysis of these interim data indicates that both trained groups perform better than the control subjects on the evaluation measures following the first and second training cycles. There are insufficient data at this point for examination of the group differences after three training cycles. Examination of the progression of closed-set identification performance during training for the 2x/wk and 3x/wk groups shows greater gains for the 3x/wk group with steady improvement through at least the first two training cycles. The preliminary findings from this ongoing study will be updated at the meeting and the implications for the development of an at-home training system will be discussed. [Work supported, in part, by NIDCD R01 DC010135].

**B7**

**The effect of dichotic processing on the perception of binaural auditory cues**

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Hearing impaired individuals often have difficulty hearing in noise because of reduced spectral resolution. Previous research suggests that dichotic processing, where information from neighboring frequency regions is sent to opposite ears, may benefit those individuals. However, dichotic processing can degrade binaural auditory cues, reducing spatial release from masking when
speech and noise are spatially separated. In this study, an eight-channel filter bank was used to create diotic (all channels presented to both ears), full dichotic, and partial dichotic stimuli that used a combination of diotic and dichotic configurations. In the partial dichotic conditions, the lowest two channels (High Dichotic), the highest two channels (Low Dichotic), or both the lowest and the highest two channels (Middle Dichotic) were presented to both ears. To test the effect of dichotic processing on binaural auditory cues, speech intelligibility in noise with and without spatial separation and sound localization were evaluated in normally hearing subjects (N=8). Head-related transfer functions (HRTFs) were used to simulate sound field conditions, and stimuli were presented through headphones. Results showed that dichotic processing degrades speech intelligibility in noise and sound localization, but degradation can be reduced by using Middle Dichotic filtering.

In order to further examine contributions of interaural time differences (ITDs) and interaural level differences (ILDs) as binaural cues in each test configuration, ITD-only and ILD-only HRTFs were reproduced in the simulation. Results showed that sound localization performance was improved both when ITDs are reproduced in the High Dichotic, and when ILDs were reproduced in the Low Dichotic as compared with the Middle Dichotic configuration. This pattern of results shows that ITDs and ILDs provide binaural cues for lower and higher frequency components, respectively.

B8
Effects of noise reduction on AM and FM reception
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Noise-reduction (NR) algorithms are employed in digital hearing-aid devices to improve the listening experience of the user in a noisy background. This improvement may be sought from an increase in speech intelligibility, an improvement in listening ease, or a reduction in listener fatigue. To improve speech intelligibility, NR algorithms attempt to reduce the background noise whilst preserving as much of the original signal as possible. Although these algorithms may increase the signal-to-noise ratio (SNR) in an ideal case, they generally fail to improve speech intelligibility. However, due to the complex nature of speech, it is difficult to disentangle the numerous effects of noise reduction which may underlie the lack of speech benefits. The goal of the present study was to assess the effects of NR algorithms on the ability to discriminate two basic acoustic features known to be crucial for speech identification, namely amplitude modulation (AM) and frequency modulation (FM). The discrimination of complex AM and FM patterns was measured for normal-hearing listeners using a same-different discrimination task. The stimuli were generated by modulating 1-kHz pure tones by either a two-component AM or FM modulator, with modulation rates centered around 3 Hz.

Three types of AM and FM patterns were used, (i) narrow-bandwidth AM patterns, (ii) narrow-bandwidth FM patterns and (iii) broad-bandwidth FM patterns. The discrimination of each type of pattern was measured in quiet and in the presence of a Gaussian white noise. Stimuli were left as such or processed via a NR algorithm based on the spectral subtraction method.

Noise reduction was found to (i) slightly improve discrimination at the higher SNRs of the AM conditions; (ii) have little effect, if any, on narrow-bandwidth FM discrimination and (iii) slightly reduce discrimination of broad-bandwidth FM patterns. The results suggest that the absence of benefit from noise reduction on speech perception may result either from a limited effect on the transmission of modulation or a counterbalancing effect between AM and FM transmission. Nevertheless, detailed analysis of individual data revealed that some listeners were more sensitive to the effects of NR (both positive and negative) than others. This suggests that a customized approach to the prescription of NR algorithms may be beneficial to the user.
**Methods on improving signal-to-noise ratio of a speech signal using visual features from a speaker’s face**

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The benefit of having access to visual cues while trying to comprehend speech in a noisy environment has been estimated as 5-8 dB of equivalent signal-to-noise ratio improvement in word recognition when compared with audio-alone speech-recognition tasks (Erber, 1969, Sumby & Polack, 1954). However, despite the fact that a 60% improvement in word recognition can be achieved in a noisy environment when adding visual cues, current hearing aid algorithms do not use visual cues to improve the audibility and intelligibility of speech for people with hearing impairment. The objective of this project is to develop and evaluate new signal processing algorithms which supplement auditory input with visual cues, including lips and facial movements, to improve speech intelligibility. An audio-visual data corpus of a single speaker reciting the 400 revised speech-in-noise (RSPIN) sentences has been recorded. The audio was acquired at 44.1 kHz and the video frame rate was 96 fps. The sentences were masked by different types of noise including white noise, pink noise, and speech babble. The dimensionality of the video frames were reduced to the location of 66 relevant facial features around the speaker’s lips, chin, nose, cheeks, and eyes at each frame of video acquired. The audio features used include the spectral peaks taken from a periodogram within a given frame of data. Two algorithms were developed to determine whether overlapping information between the auditory and visual information could be optimally fused to create an auditory filter through which the noisy acoustical signal would be passed. One algorithm used for the filter generation was based on maximal mutual information; the second was based on canonical correlation analysis between the auditory and visual data streams. The ideal output of the filter would be an improved signal (higher SNR) that is comprised of only the auditory content which corresponds with the visual cues. Contrasts and comparisons will be made with existing single-microphone noise reduction strategies including spectral subtractive, subspace, statistical-model based, and Wiener-type algorithms. Methods for augmenting current signal processing strategies with visual features will be discussed.

**Evaluation of an automatic procedure for detecting frequency-following responses to voice pitch in American and Chinese neonates**

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Ohio University

Voice pitch carries important lexical and prosodic cues for speech understanding. Objective measures of the brainstem’s responses to voice pitch, however, were not achieved until recent years. These results indicated that when neurons in the human brainstem were activated by speech signals, synchronized neural activities reflecting the changes in voice pitch were preserved in the scalp-recorded frequency-following response (FFR) to voice pitch. Recent studies expanded the scope of the FFR to voice pitch by showing characteristics of such a response to speech and non-speech stimuli in normal-hearing adults. Jeng and colleagues reported that the FFR to voice pitch recorded in young infants and neonates accurately reflected the pitch contours of the acoustic stimuli.

Although characteristics of the FFR to voice pitch have been reported for normal-hearing adults, infants and neonates, the presence of such responses has been dependent on subjective interpretation of experimenter. If the FFR to voice pitch is meant to be an objective method to examine pitch processing in the human brainstem, development and validation of an automated procedure suitable for detecting the presence of such a response is needed. One previous study reported adequacy of an automated procedure, including a control-experimental protocol and response-threshold criteria, to detect the presence of an FFR in normal-hearing adults. Applicability of this automated procedure, however, has not been examined for normal-hearing neonates. The goal of the present study was to evaluate the ade-
Accuracy of this automated procedure in detecting FFRs for neonates who are 1-3 days after birth.

Twenty American (9 boys; age: 1-3 days) and 20 Chinese (10 boys; age: 1-3 days) neonates were recruited. All the American neonates were born to parents who were native speakers of American English and were recruited from O’Bleness Memorial Hospital in Athens, Ohio. All the Chinese neonates were born to native speakers of Mandarin Chinese and were recruited from Nanjing Maternity and Child Health Care Hospital in Nanjing, China. All neonates passed an automated auditory brainstem response screening and were negative for neurologic or syndromic disorders. All recordings were obtained in quiet nursery stations in the neonates’ respective hospitals. Results demonstrated that the automated procedure provided adequate sensitivity (53-90%) and specificity (80-100%) for both the American and Chinese neonates.

B11
The effect of acoustic cue redundancy on the perception of stop consonants by older and younger adults
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Older adults process speech differently from younger adults and thus may need different hearing aid processing. In previous studies, we found that with all other acoustic cues intact, older adults were tolerant of a certain amount of amplitude alteration caused by wide-dynamic-range compression (Jenstad & Souza, 2007), but there was a threshold beyond which changes became detrimental to speech understanding. The detrimental effect of amplitude alteration was greater in conditions of reduced acoustic redundancy, when the listeners were given fairly impoverished speech signals with increasing amounts of distortion applied. To follow up on that finding and explore whether older listeners require more redundancy in the speech signal than do younger listeners, we examined how older adults with normal hearing combine two primary acoustic cues (the stop gap closure and the release burst amplitude) to detect the presence of a stop consonant in a word for two phoneme contrast pairs (/p/ in speed/seed and /t/ in steam/seam) constructed from natural recordings.

Six older and 6 younger participants with normal hearing (better than 25 dB HL from 250-4000 Hz) were tested. Using a 2-alternative forced choice (AFC) paradigm, participants indicated whether they heard the word containing the stop consonant or not. ANOVA of the results revealed a main effect of burst amplitude and inconsistent effects of age but no interaction between burst amplitude and age, p = .803 for /p/ and .232 for /t/. For those steam/seam contrast stimuli in which closure gap was the only cue to stop presence, older listeners reached threshold perception of /t/ as gap duration increased but younger listeners did not. Because they do not show an interaction between age and the presence of redundant acoustic cues, these combined patterns of results do not support the redundancy hypothesis. We found that, contrary to expectations and similar to younger adults, older adults with normal hearing can use a single acoustic cue to detect a stop consonant, if that single cue is strong enough, and integrate the two cues in the same manner as younger listeners with normal hearing.

B12
Prediction of optimal gain for speech intelligibility, considering differences in cochlear damage
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Several hearing-aid fitting rationales share the aim at providing gain which leads to maximal speech intelligibility (SI) and at the same time producing comfortable listening levels to the hearing-aid wearers. The various attempts have led to different prescription rationales. Many of the approaches are based on obtaining maximal SI using the articulation index (AI) or speech intelligibility index (SII, ANSI 1997) theories. However, the calculation schemes of standardized procedures do not account for effects of hearing impairment other that elevated hearing threshold levels (HTL). Other consequences of hearing loss, such as decreased spectral resolution is recognized and have been found to vary
greatly among individuals with similar HTLs (e.g., Moore et al., 1999). The variation is thought to reflect differences in the proportion of dysfunctional inner and outer hair cells (IHC and OHC). To control for comfortable loudness one can consider using a loudness model, such as in the NAL-NL2 (Keidser et al. 2011) rationale where the loudness model of Moore et al. (2004) was applied.

The present work demonstrates how excitation patterns (EP) produced by a modern loudness model (Chen et al., 2011) may replace power spectra of the speech and noise in the SII calculation, leading to an EP-based SII (EP-SII) similar to the procedure suggested in Rhebergen et al. (2010). The described procedure will lead to a more thorough account for the consequences of hearing impairment, and with optional input of estimates of the amount of individual-specific IHC and OHC loss to supplement the HTLs. The EP-SII model will be the basis for a gain rationale for optimal intelligibility. This model will generally suggest that a gain lower than that suggested by the SII is beneficial for hearing impaired listeners in good listening conditions. This hypothesis generated by the EP-SII was tested experimentally in a number of hearing-impaired listeners. Speech intelligibility and preference for gain settings were measured using the EP-SII based rationale and the NAL-NL2 rationale. Furthermore, it will be discussed what a reasonable default setting for the OHC and IHC loss proportion could be based on recent findings in the literature (Moore et al., 1999; Jepsen and Dau, 2011; Jürgens et al., 2011). It is discussed how individual differences in speech intelligibility for listeners with similar hearing thresholds can be partly explained by allowing the IHC and OHC proportion values in the EP-SII model to deviate from the default setting.

**B13**

**What factors influence variation in directional microphone benefit?**

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Based on the difference in signal-to-noise ratio (SNR) required to recognize 50% of speech in noise with directional and omnidirectional microphones, there is a large volume of studies that have demonstrated that the average benefit of directional microphones is about 3.5 dB. This is a value that corresponds well with the average reported directivity index for directional instruments of the time. Individual variations in benefit have also been noted, and are reportedly rather large, ranging from no benefit to about 12 dB. The large variation is curious as one would expect the physical performance of a directional microphone to remain the same for all individuals. A better understanding of the reasons for the large variations could lead to better clinical guidelines for fitting directional devices, and to more effective developments of directional microphones in the future. Based on speech-in-noise performances with omnidirectional and directional microphones by 60 hearing-impaired listeners with varied degrees of hearing loss, a current study investigates whether variation in directional benefit is predominantly due to variation in physical SNR improvement, variation in a person’s ability to use the SNR improvement, measurement error, or a combination of these. Directional benefit will be related to such factors as in situ directional patterns, in situ SNRs, vent and amplification paths, angle of microphones relative to target, cognitive performance, and repeated measures. Findings from this study will be presented and discussed.

**B14**

**Acoustic simulation using 3D modeling of the digital hearing aids**

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Small changes in various parameters (shape of housing, material of housing, HTRF, wearing...
position, type, etc...) is causing a large change of acoustic output sound that be listened the hearing aid wearer through a receiver. It gives a big difference in speech quality and intelligibility to the digital hearing wearer. And the satisfaction of hearing aid wearers will appear as a variety of response (“satisfaction,” “comfort,” “dissatisfaction,” and “discomfort” sounds good, but inconvenient.)

For this reason, acoustic simulation that considered small change by various parameters is very important to develop the digital hearing aid for the output predicting and evaluation. We was established the simulation environment using 3D model of digital hearing aid in this study. The purpose of this study was done in observation of the output signal according to changes in various parameters that include acoustic properties (structure, spatial position, and spatial environment) of digital hearing aid. And, the accurate simulation is also the purpose of this study. And this study was conducted with the following hypothesis: To predict the result and effectiveness of various prescriptions that are purposed to enhance the speech quality and intelligibility of hearing aid wearer, the evaluation and simulation must be carried out in the environment that is considered the perspective of digital hearing aid.

In the abstract, we was conducted the performance comparison of beam-forming algorithm (delay and sum, 1st order directional, and 2nd order directional beam-forming) according to the structure of hearing aid (“shape of microphone cover,” “microphone location,” and “distance between the microphone”) spatial position, and spatial environment (reverberation, non-reverberation) in the developed simulation environment. Model of 3D hearing aid based on “dot²” that is RIC type hearing aid of GN Resound used in this abstract. The 3D models have been implemented through the Solidwork. And, acoustic simulation according to various parameters was conducted through Comsol. [This work was supported by the strategic Technology Development Program of Ministry of Knowledge Economy (No. 10031764) and the Seoul R&BD Program (SS100022).]

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**B15**

**Binaural vs. monaural beam-forming in complex acoustic scenes, a lab comparison**

Gabriel König, Juliane Raether, Michael Kramer, Peter Derleth and Stefan Launer

Phonag AG

The exchange of audio signals between hearing aids which has become available in some commercial products in recent years allows for algorithmic solutions called ‘Binaural Beamformer’. Binaural beam-forming techniques promise a more selective directivity pattern compared to monaural beam-forming techniques. A performance benefit is expected especially in demanding acoustical scenes with a single talker as listening target in a noisy environment including distracting talkers.

Four different algorithmic variants of monaural and binaural beam-forming techniques implemented in wearable devices and used over several weeks by 20 moderately to severe hearing impaired listeners are compared. Two types of acoustic couplings (‘RIC’ and ‘Earmold’) were chosen depending on the preference of the listener. During laboratory visits performance comparisons were performed mainly for two scenes (‘distracting babble noise from side’ and ‘diffuse cafeteria’). Results are presented for the perceptual dimensions: Speech Reception Threshold (LSA); Subjective speech intelligibility, noise suppression and overall quality (MUSHRA); Localization of a single talker including a subjective rating of compactness; Questionnaire on spatial hearing and sound quality (SSQ).

**B16**

**The effect of enhanced onsets of the speech envelope on speech reception of hearing impaired listeners**

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Recent studies have shown that the transient parts of the speech signal contribute much to speech intelligibility in normal hearing (NH) listeners. Motivated by this, transient enhancement algorithms were developed for NH listeners which confirmed the potential for speech intelligibility.
improvement with enhanced transient cues of the speech signal. On the other hand, transient enhancement algorithms were developed for cochlear implants motivated by the fact that the cochlear implant bypasses the auditory nerve synapse and therefore also the rapid adaptation effect that emphasizes transient parts by nature. It was shown in CI listeners and CI simulations with NH listeners that enhancing the onsets of the speech envelope lead to a speech reception threshold (SRT) improvement in stationary speech shaped noise and in an interfering talker condition.

In this study, we investigate the potential of the onset enhancement of the speech envelope on speech intelligibility improvement with hearing impaired (HI) listeners. The signals are processed with a new version of the enhanced envelope continuous interleaved sampling (EE) algorithm that was developed for cochlear implants. An adaptive procedure is used to determine the SRT in stationary speech shaped noise. To ensure audibility of the signal the processed signals were rescaled to the same overall loudness of 65 dB SPL and fitted to the audiogram of the HI subject with the NAR-RP or the NAL rule depending on the respective hearing loss.

Results of a pilot study with HI subjects suggest that a speech intelligibility improvement with the enhanced onsets of the speech envelope can be achieved. The enhancement of the onsets of the speech envelope also increases speech intelligibility in hearing impaired listeners where the rapid adaptation effect is not bypassed. These results underline the outcome that transient parts are important for speech intelligibility in adverse listening conditions. Further tests are carried out.

**Evaluation of a transient-noise reduction algorithm: Speech performance and comfort**

*Petri Korhonen*

*Widex ORCA*

This study examined the functional utility of a commercial transient noise reduction (TNR) algorithm using speech intelligibility and user preference measures. The goal was to demonstrate that the algorithm does not degrade speech identification performance, while providing greater listening comfort in the presence of transient noise sounds. Thirteen experienced hearing aid users with symmetric bilateral sensorineural hearing loss participated. A single-blinded repeated-measures design was used to study the effect of TNR on speech identification performance (ORCA-NST) and wearer preferred gain. In addition, subjective preference was evaluated in a paired comparison task. The algorithm did not degrade speech performance in quiet. The participants showed a preference for TNR algorithm when listening to transient noise sounds in speech and quiet. Overall hearing aid gain was adjusted while participants listened to speech presented in the presence of transient noise sounds. Participants lowered the prescribed gain on average 3 dB less when TNR was activated. The phoneme identification performance was 12.1% better with the gain adjusted with TNR than without. Results demonstrated that the TNR algorithm provided more comfortable listening in the presence of transient noise sounds, while having no negative impact on speech identification performance. Consequently, the algorithm ensured more consistent audibility through less gain reduction in the presence of transient noise.

**A study of fitting formula for enhancement of Korean speech intelligibility**

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Generally, hearing aids compensate for hearing loss by amplifying input signal and compress the amplified sounds. Hearing aids fit the sounds in reduced dynamic range. Reasonable insertion gain must be offered because each person has different hearing loss. So for enhancement of hearing aids, many fitting formula have been developed and many studies are going on.

In this study, K-formula which based on NAL-NL1 (NAL-nonlinear, version 1) was suggested for maximize speech intelligibility. K-formula was derived from the same procedure which is used for deriving NAL-NL1. K-formula consider the long-term average speech spectrum of Korean
instead of English and BIF(Band Importance Factor) since the frequency characteristic of Korean is different from English. New insertion gains of K-formula were derived using the SII (speech intelligibility index) program provided by ANSI (American National Standards Institute). In addition, the insertion gains were modified to maximize the intelligibility of high frequency words. To verify effect of new fitting gain, we compared K-formula with NAL-NL1 from word recognition score (WRS) and preference test. In the WRS, a word set as test material consist of 50 I-syllable word and four of sentence files generally used in hearing clinic. The normal hearing subjects (n=10) who had no history of auditory pathology participated in this experiment.

In the experiment with the normal hearing subject, we distorted the word sets and sentences using the hearing loss simulator from NIOSH (National Institute for Occupational Safety and Health). Two of the sensorineural hearing loss models (case 1: the moderate hearing loss with severe loss at high frequency; case 2: The severe hearing loss with sever loss at high frequency) were used in hearing loss simulator. For the accurate result, WRS was conducted in the dictation. As a result of experiment the WRS of K-formula was better than NAL-NL1.

Finally, the result was obtained that increase gain of mid-high frequency bands and decrease gain of low frequency bands were effective way for maximize speech intelligibility of Korean. [This work was supported by grant No. 10031764 from the Strategic Technology Development Program of Ministry of Knowledge Economy and grant No. SS100022 by Seoul R&BD Program.]

Towards a spatial speech-in-speech test that takes SNR confounds and ecological validity into account

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Measuring speech intelligibility by means of adaptive Speech Recognition Threshold (SRT) procedures has become increasingly popular, for good reasons: they always yield a result with desirable statistical properties and they are easy to administer. However, it is becoming increasingly clear that adaptive SRT measures have certain drawbacks, related to the unbounded nature of the Signal-to-Noise-Ratio (SNR) at which criterion performance (the SRT) is achieved. Quite often the SRT will be a double-digit negative number, which corresponds to listening conditions very rarely encountered in real life. This compromises the ecological validity of the result. Furthermore, if testing involves hearing aids, it also means that these devices and the signal-processing algorithms in them may be operating in conditions for which they are not intended.

Another problem is that different groups of listeners (e.g., hearing-impaired listeners and normal-hearing listeners) may have radically different group mean SRT’s, which introduces a potential ‘between-groups SNR confound’, as pointed out by Bernstein and co-workers. Even within a group of hearing-impaired listeners, large variation in SRT is often observed, which introduces a possible ‘within-group SNR confound’. Both types of SNR confounds have the possibility to cause faulty conclusions, if test results are interpreted without regard to potential SNR confounds.

One way to address these issues is to modify the basic adaptive SRT protocol, providing the experimenter with control of the SNR at which testing takes place, as the aim is to ensure that all test subjects in a given experiment reach criterion performance at (or at least close to) a common target SNR. The present work is one step towards such a test. More specifically, the aim is to devise a spatial speech-in-speech test with a selection of experimental conditions for the experimenter to choose from. In the present work a range of different background talkers were tested together with the Danish Dantale2 target sentences, which are spoken by a female. In all cases the background was continuous speech. Three different female and three different male background talkers were tested, in order to examine the effect on the SRT of changing background-talker gender, as well as individual talker variation. In addition, the effect of word-scoring versus sentence-scoring was examined. 18 hearing-aid users served as test subjects.
The examined ‘SRT manipulators’ will be discussed in terms of their magnitude and consistency, and their potential side-effects.

B20
Establishing and qualifying a hearing impaired expert listener panel
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3 GN ReSound

In the food and fragrance industry sensory panels are widely used as a means of developing products to have the desired features and quality. Using trained assessors for evaluating how products are perceived is based on well-established methodologies primarily from the food science which can be transferred to other sensory domains like sound and hearing. The advantage of the sensory methods is that they can provide a reliable collection of quantitative measures for the product characterization through experimental designs and double blind test paradigms. A trained listening panel can make robust and consistent ratings of the key audio features in hearing aids which can lead to a product description of the key features (e.g., level of Sharpness).

In this joint research project between DELTA, Oticon and GN ReSound, a hearing impaired listening panel of 17 persons was established for evaluating the potential use in hearing aid sound quality evaluations. The listeners were all selected to have a given hearing loss profile (N3 according to IEC 60118-15) to establish a certain level of homogeneity in the panel. Apart from the audiometry requirements, a number of pre-screening tests were conducted to evaluate the potential skills for sensory evaluations.

The main research questions were related to:

- The performance of a hearing impaired panel vs. a normal hearing listening panel
- Bias in sound quality preference assessments from pre-adaptation
- Performance potential of the hearing impaired expert listener panel

Listening tests were performed with commercially available hearing aids recorded in different sound environments. The hearing aids were all fitted with default settings for the given N3 audiogram. Evaluations were made on Preference of the overall quality for the different products in the test. And this was followed by a consensus attribute development process that lead to the identification of the most dominating audio characteristics of the hearing aids. Based on these attributes a set of scales and definitions where developed which were used for in final assessments of the products.

The results showed differences on overall sound quality preference between a hearing-impaired panel and a normal hearing panel. It was also found that there was no overall bias effect from the pre-adaptation. Finally, the panel performance as seen from a sensory analysis perspective (discrimination and panel agreement) showed encouraging results.

B21
Modulation enhancement improves perception of interaural time differences in vowels with combined acoustic and electric stimulation
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The success of cochlear implants (CIs) has resulted in relaxed implantation criteria, which in turn has led to a growing number of implantees with residual hearing in the other ear. These CI users can benefit from binaural input by wearing a hearing aid (HA) in the non-implanted ear. This combination is called bimodal stimulation.

Binaural input can provide access to binaural cues, which are important for localization and understanding speech in noisy environments. Despite the binaural input, bimodal listeners still have poor localization ability and experience problems understanding speech in noise. Since current clinical devices function separately and are not synchronized, binaural cues are not optimally transmitted. The transmission of interaural time differences (ITDs) is particularly limited, because modulations are not always retained in
the electrical stimulus and often they are not synchronized across channels. Previously we have found, using computer-controlled stimulation, that bimodal listeners were able to perceive ITDs in the stimulus envelope when signals were synchronized between the ears and contained clear modulations. This shows that for many bimodal listeners the limitation is not entirely at a perceptual level but rather due to the design of the devices.

We developed a signal processing strategy named modulation enhancement strategy (MEnS). The strategy detects the peaks occurring at the fundamental frequency in the acoustic signal and introduces modulations in the electric signal at these time instants. Modulations are provided synchronized across frequency channels, and synchronized with modulations in the acoustic signal.

ITD detection performance was assessed using vowel stimuli, processed with MEnS or the clinical standard processing strategy ACE, with five bimodal listeners. Additionally, subjects performed a lateral position judgment task using the same stimuli. ITD detection performance was significantly higher with MEnS than with ACE. With MEnS four out of five subjects had just noticeable differences in ITD in the order of 100-250us. Preliminary results from the lateralization judgment task show lateralization of the stimuli based on ITD. Overall results encourage the further development of the bimodal MEnS strategy, which ensures clear modulations synchronized between the acoustic and electric signal to improve perception of binaural temporal cues in speech. [This study was supported by IWT-Vlaanderen project 080304 and Cochlear Ltd. Anneke Lenssen was sponsored by the European Commission within the ITN AUDIS, grant agreement number PITN-GA-2008-214699, Tom Francart was supported by a Post-Doctoral Fellowship of the Fund for Scientific Research of the Flemish Government and a Marie Curie International Outgoing Fellowship, grant agreement number PIOF-GA-2009-252730.]

Assessing the effect of different hearing-aid microphone polar patterns on speech intelligibility in noise
M. Samantha Lewis, Frederick Gallun, Jane Gordon, Jeffrey Shannon and Gabrielle Saunders
National Center for Rehabilitative Audit

Rationale/Purpose: This work had two main objectives: 1) assess the impact of different hearing-aid microphone polar patterns on speech intelligibility in noise and 2) determine whether or not speech-intelligibility performance could be accurately predicted by using the Speech Intelligibility Index (SII; ANSI 1997) and two different measures of hearing-aid directional-microphone performance.

Methods: Fourteen subjects were fit bilaterally with in-the-ear hearing aids that had the capability of being programmed in one of the four microphone polar patterns: 1) omnidirectional, 2) bidirectional, 3) cardioid, and 4) hypercardioid. Speech intelligibility in noise was assessed with each of these polar patterns using the Hearing In Noise Test sentences (Nilsson, McCaw & Soli, 1996) and uncorrelated speech spectrum shaped noise. The sentences were presented from a loudspeaker located at 0° azimuth using an adaptive procedure, while the noise competition was presented at a constant level from four loudspeakers located at 45°, 135°, 225°, and 315° azimuths. Performance was related both to SII scores and to direct measures of hearing-aid microphone directivity. The direct measures of hearing-aid microphone directionality included a front-to-back ratio using real-ear measures completed using the Frye Fonix 6500 and 2 dimensional-directivity index (D-DI) calculations made, in part, by completing polar plot test-box measures with the Frye Fonix 8000.

Results/Discussion: Analyses were completed on 12 subjects. A repeated-measures ANOVA revealed a significant difference in benefit (over unaided performance) between microphone conditions (F(3,33)=4.25, p=0.012). Pair-wise comparisons, using a Bonferroni correction, revealed that the benefit obtained with the hypercardioid microphone polar pattern was significantly different (better) than that obtained with the omnidirectional microphone polar pattern (p=0.010). No
other conditions were significantly different from one another. Although SII and microphone directionality measurements were not strongly related to raw performance, it was found that individual improvements in performance (relative to unaided performance) were related to changes in the SII relative to unaided SII scores. The clinical implications of these results will be discussed. [This work was supported by the National Organization for Hearing Research and the VA Rehabilitation Research and Development Service. The Frye Fonix 8000 used in this investigation was loaned to the National Center for Rehabilitative Auditory Research for research purposes].

References:

Large scale objective assessment of preferred gain and frequency response for multiband compression hearing aids
Adrian Lister and Drew Dundas
Starkey Hearing Technologies

Background: Hearing aid fitting targets are typically generated from audiometric data based on various goals including normalization of overall or specific loudness, maximization of the speech intelligibility index, and preservation of high-level input comfort. It is commonly accepted, however, that patients will request deviations from fitting targets to address individual perceptions and sound quality preferences. This study sought to investigate the deviations from common prescriptive targets that occurred as a result of subjective fine-tuning after the initial calculation of the fitting from the audiogram. A review of data stored on devices that were received and read for repair and or return for credit was conducted to compare modeled responses based on the audiogram to fine-tuned settings retained after at least two follow up visits, in the interests of determining whether gain/frequency responses based in different fitting prescriptions converged after subjective fine tuning.

Purpose: The purpose of this study was to investigate the final gain/frequency responses achieved via subjective fine tuning after starting fittings with prescriptive targets calculated based on the NAL-R, NAL-NL1, DSL-V and eSTAT fitting formulas.

Research Design, Sample, Data collection and Analysis: Gain/frequency response data and audiometric data were extracted from devices that were received for inspection or repair over a three-year period. Modeled target responses for the audiograms associated with the devices were calculated and compared to the final frequency response contained with the devices as a function of the fitting target selected by the clinician for the fitting.

Results: Comparisons suggest that patient-directed fine-tuning resulted in convergent responses wherein the preferred gain/frequency shaping was similar despite variant starting fitting algorithms. The preferred fittings resulted in a modeled real ear aided response that provided significantly less patient-desired low frequency audibility than prescribed by the NAL-NL1, DSL-V or NAL-R prescriptions for average level inputs, and yet for soft inputs, desired increased low to mid frequency audibility, and increased output for louder speech shaped inputs.

Conclusions: The results of this study suggest that compression ratios commensurate with the severity of the hearing loss is desirable to listeners with a wide range of SNHL, while flatter overall gain shaping is also considered desirable. These results support the idea of tailoring compression ratios according to the severity of the hearing loss, while limiting maximum output to levels commensurate with loudness experienced by normal hearing listeners. Large scale analysis of preferred fitting parameters may yet provide useful data that may lead to improved subjective outcomes for individuals with SNHL who are fit with modern amplification systems.
Evaluation of binaural cue preservation in hearing aids: a block-based ITD and ILD estimate method
Guilin Ma, Fredrik Gran and Tobias Piechowiak
GN ReSound Research

Spatial hearing has attracted more and more attention in the hearing-aid industry. To present the users with preserved binaural cues, ITD and ILD should not be distorted.

In order to evaluate the effects of hearing aid as a system or individual modules on ITD/ILD, a block-based ITD/ILD estimate method using input and output measured in an anechoic environment has been developed. The method extracts two time-varying filters representing the processing in the left and right channel respectively, and cross-correlate the two filters at low frequencies to estimate ITD. The two filters are then transformed to the frequency domain to calculate the magnitude difference for ILD.

To derive the ITD/ILD error, true ITD/ILD is calculated from measured head-related transfer functions (HRTFs). ITD is obtained by fitting the excessive phase component of HRTF with a linear curve and ILD is extracted by calculating the magnitude difference between left and right HRTF in each frequency bin.

The results show that the metric can estimate ITD/ILD very well and can track the changes of ITD/ILD on the order of millisecond.

How compression affects the ability to use onset, rise time and decay time differences for hearing out one tone sequence in the presence of another
S.M.K. Madsen and B.C.J. Moore
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A person listening to music often tries to hear out one melodic line from other lines. The use of fast-acting amplitude compression makes it more difficult to hear one voice in the presence of another voice for speech processed to retain only temporal envelope cues in different frequency bands (Stone et al., 2009), and the use of fast compression may also make it harder to hear one melodic line in the presence of others. Fast compression may also distort the onsets of sounds, by introducing "overshoot" effects, making it harder to hear the slightly different onset, rise, and decay times of notionally "simultaneous" tones that are used as cues for segregation of different melodic lines (Rasch, 1978).

We assessed how fast- and slow-acting five-channel compression, using a 2:1 compression ratio, affects the ability to use onset, rise, and decay time cues to detect one (signal) complex tone when another (masking) complex tone is played almost simultaneously. Five normal-hearing subjects were tested. The signal either started slightly before the masker (onset asynchrony 0 to 40 ms) or the signal and masker started and stopped at the same time but the signal had a faster rise or decay time than the masker.

The signal threshold decreased (performance improved) with increasing onset asynchrony, and this effect was significantly greater with than without compression. Compression speed had no significant effect. Performance also improved with increasing difference between signal and masker rise/decay time, but the improvement was not influenced by whether or not compression was applied.

The results indicate that envelope distortions introduced by compression do not adversely affect the ability to use small onset asynchronies to improve detection of a signal tone in the presence of a masking tone. Rather, compression leads to a small increase of the signal level relative to the masker level, and this improves performance. [Supported by Starkey (USA) and the MRC (UK).]

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Rasch R.A. 1978. The perception of simultaneous notes such as in polyphonic music. Acustica, 40, 21-33.
Environment Classification and Adaptation
Martin McKinney, Anil Shankar, Matt Kleffner and Tao Zhang
Starkey Hearing Technologies

The diversity of acoustic situations encountered by typical hearing-aid wearers precipitates a need for hearing aids (HA) that automatically adapt to their surroundings. Attempts have been made in HA technology to automatically classify and adapt to the surrounding environment but performance results and user appreciation are mixed. Current approaches typically account for single audio classes only and do not easily handle audio class mixtures that listeners commonly encounter. Additionally, computational constraints in HAs pose a significant challenge to complex algorithms for audio classification. We address these issues in three ways: 1) through ground-truth audio labeling of acoustic sound mixtures for robust model training; 2) through classification and adaptation system architectures designed for multiple and simultaneous sound classes; 3) and by systematically evaluating trade-offs in classification performance vs. computational intensity.

The integrity of a training database is paramount for machine-learning applications. Incomplete or inconsistent labeling of training data can severely degrade classification performance, especially when the elements (i.e., sound recordings) belong to more than one class. We developed and assessed a robust method for labeling an audio training database using a large number of labelers. We employ a crowd-sourcing mechanism (Amazon’s Mechanical Turk) to collect audio class labels for a number of audio recordings. Two separate tasks were run on the same set of audio recordings: one asked listeners to label only the most prominent sound; the second asked listeners to label all sounds they heard in the recording. Results show that the relative strength of an audio class in a recorded mixture is similar across subjects as it is within subjects. Results suggest that consistent and reliable ground-truth labels for audio mixtures can be obtained in this efficient manner.

The architecture of an audio classification and adaptation system can be designed to inherently address combinations of sound classes. We show that by using a set of parallel single-class detectors with appropriately coupled adaptation responses, the system can handle simultaneous sound classes. In addition, the detector framework allows for optimal feature selection specific to individual sound classes. Together, these design attributes enable the system to better respond and adapt to combinations of sound classes.

Finally, we examine how computational intensity, as defined by frequency resolution, trades off with classification performance. For some audio classes, clear performance advantages can be seen by subtle increases in frequency resolution.

Applications of binaural beamformer technology in assistive listening devices
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In recent years, a new kind of binaural beamformer algorithm has been developed by the HEARing Co-operative Research Centre, Australia. The binaural beamformer was implemented in a real-time processor and evaluated in different listening assistive device applications, which included behind-the-ear (BTE) hearing aids, active earmuffs, and bone-conductors. The algorithm was optimised to suit each individual device and listeners’ needs. The optimisation included changes to the beamwidth and depth of side-lobe cancelation. Speech reception threshold (SRT) in multi-talker noise measures were obtained for different conditions that included 2-to-22 interfering talkers and mild (T60 = 0.3 sec) to moderate (T60 = 0.45 sec) amounts of reverberation at distances exceeding the critical distance. In two successive tests, normal hearing listeners were presented with off-line head-related-impulse-response renderings over head-
phones and, in the second test, listeners were fitted with BTEs and the beamformer in a real sound field. The SRT scores were significantly better for the beamformer presented under headphones than in real-time listening conditions \( [p < 0.01] \). During the real-time assessments, the beamformer SRT scores were also poorest for a narrower beamwidth than with a broader beamwidth. In a final assessment, hearing-impaired listeners were fitted with real-time BTEs set up to examine the beamformer performance relative to the conventional directional microphones. The conventional directional microphone processing was adjusted to produce a fixed cardioid polar response. In addition, sound amplification was provided according to NAL-NL2. The beamformer SRT scores were, on average, 2.6 dB SNR better than directional microphones in the 22-talker, and 3.2 dB better in the 2-talker condition \( [p < 0.001] \). Furthermore, the degree of benefit increased with the degree of hearing loss. When normal hearing listeners were fitted with bone conductors, the beamformer SRT scores were 5.2 dB better than directional microphones. Finally, across all devices examined, all participants had 100% preference for the beamformer in difficult listening situations (noise level greater than the target level) whereas the directional microphones and the beamformer were equally preferred in easy listening situations (noise level less than the target level). Detailed results of these experiments will be discussed.

**The potential for using the Hagerman-derived signal-to-noise ratio with individual fittings**

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Despite improvements in hearing aid (HA) technology, there remain many instances of minimal improvement in speech understanding in noise. One hypothesis is that aided speech intelligibility is partly dependent on the signal-to-noise ratio (SNR) at the output of the HA. The rationale behind this belief is based on the observation that the presence of noise degrades important level and temporal cues in the central auditory system (e.g., Billings, Tremblay, Miller 2011). Cortical neurons have been shown to be sensitive to the relative SNR, rather than stimulus level (Billings, Tremblay et al., 2009). A recent development by Hagerman and Olofsson (2004) has facilitated our ability to effectively separate speech and noise post-HA processing. Using this technique, several researchers have suggested that HA algorithms do modify the SNR (Souza et al., 2006; Naylor and Johannesson, 2009). The potential to measure the SNR at the output of a HA could be of great benefit to clinicians. Previously, researchers have used HA simulations, or one programmed to a flat frequency response. The purpose of this investigation was to estimate the amount of variability expected in (1) HA output SNR given individual fittings, using HAs and algorithms currently on the market, and (2) the error generated from using Hagerman’s phase-inversion technique with these algorithms in individualized fittings.

Twenty-five participants with sensorineural hearing loss no worse than 70 dB HL were included in this study. Three HAs were programmed to match a NAL-NL1 65 dB target for a digital speech signal. Algorithms were activated to manipulate the SNR at the output of the aid (linear, compression, noise reduction). The HA was placed on KEMAR with sentences and 6-talker babble presented from 0° azimuth. Hagerman’s phase-inversion technique was used to measure the short- (30 ms) and long- (120 ms) term SNR at the HA output. Results show that the algorithms modified the SNR by +3 to -6 dB, and varied across individuals. Characteristics of the participant, such as PTA and input SNR, could not explain the variability in HA output SNR. The difference between short- and long-term SNR calculations was approximately 0.5 dB, on average. The error from using the Hagerman technique with the algorithms in this study was estimated on an individual level and results are presented. Clinical implications will be discussed. [Work supported by grants from the University of Washington’s Speech and Hearing Department and NIH (T32-DC005361 & P30-DC04661).]

References:


**B29**

**Evaluation of real time spectral enhancement system based on individual frequency selectivities of sensorineural hearing**

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The objectives of this study were to evaluate the spectral enhancement method in hearing aids, whose parameters were adjusted based on the degree of frequency selectivities measured for individual hearing-impaired listeners.

Spectral enhancement method is basically performed on 64 points FFT; therefore the unit of frequency resolution is 250 Hz (sampling frequency 16 kHz). This method works on 20 points out of 32 points that correspond to the frequency range of 625-5125 Hz. Concretely speaking, the groups of frequency points; the lower range (3 points), the mid-range (5 points) and the higher range (12 points), are working independently.

The method enhances ±6 dB at the peaks and valleys of the magnitude spectrum of the original signal. To measure reduced frequency selectivities, a system that can measure individual auditory filters of hearing-impaired was developed in 2007. Hearing aid fitting software can measure frequency selectivities and stimuli are presented via hearing aid witch is connected the software.

This system can measure auditory filter at 1, 2 and 4 kHz.

Fourteen subjects with reduced frequency selectivities (eighteen ears) and two subjects with normal frequency selectivities (three ears) were participated. The Japanese monosyllabic recognition test (SNR 10dB) was performed under the 2 conditions (Spectral Enhancement condition was set at Off (SEOFF) and On (SEON)). In subjects with reduced frequency selectivities, the parameters were configured in SEON as follows; after the auditory filters of the subjects were measured at 1kHz, 2kHz and 4kHz, when the auditory filter at 1 kHz (2 kHz or 4 kHz) was 3 times broader than the auditory filter of normal hearing at the frequency, the spectral enhancement was applied respectively. For subjects with normal frequency selectivities, two conditions (SEOFF and SEON) test were performed. In SEON case, spectral enhancement was set at ON for all frequencies.

In subjects with reduced frequency selectivities, the results of the monosyllabic recognition scores in SEON tended to be improved than SEOFF. The other way, in subjects with normal frequency selectivities, the results in SEON tended to be got worse than SEOFF. However, there were no significant differences of the results of consonant between SEON and SEOFF.

These results suggested that this spectral enhancement method was beneficial and the effect of the method might increase by the optimization towards frequency selectivity. Spectral enhancement was not suitable for subject with normal frequency selectivities. Furthermore, considering of frequency selectivity for adjusting spectral enhancement was useful.

**B30**

**Exploring narrative effects in hearing aid fittings**

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The sequence of actions in a clinical meeting forms a ‘narrative’ or story. The purpose of this exploratory study was to demonstrate the power
of the narrative to affect the client’s perception of the hearing-aid fitting. To isolate narrative effects, we contrasted dispensing processes having divergent narratives but identical acoustical results. In a balanced cross-over design, we carried out fittings with these processes on hearing-aid clients, and administered a range of self-report outcome measures after two weeks of use. The two carefully rehearsed dispensing process narratives were:

- ‘Diagnostic narrative’. The client remains passive, and the dispenser makes a number of diagnostic hearing assessments. Unknown to the client, the actual fitting is based on hearing thresholds only.
- ‘Interactive narrative’. The client is made to believe that he/she has adjusted the HA settings to his/her own preferences. Again, unknown to the client, the fitting is based on hearing thresholds only.

Thus the take-home hearing aid settings were identical for both narratives.

Experiment 1, 24 experienced hearing-aid users: No order effects were observed. 20 of the 24 subjects had a preferred fitting, and preference judgments were relatively secure. All subjects except one gave exclusively sound-related reasons for their preferences (“sounds more clear” etc). This is surprising, since the two fittings were acoustically identical. We must suppose that it is the subjects’ perception of the fitting process which determined their preferences and self-report of sound quality.

Experiment 2, 16 new users: Opinions were (as usual) more vague for this group, but a clear order effect was observed (13/16 preferred the second fitting). Recall of the sequence of events in the fittings was poor. These results suggest that acclimatisation is the over-riding factor for this user group.

The sum of results will be discussed in relation to habitual fitting practices and research methods.

Fast preference-based individualization of hearing aid sound: An interactive probabilistic modeling approach

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Modern digital hearing aids (HAs) offer an almost infinite number of possible combinations of adjustable parameter settings. They also contain several advanced features that adapt to the sound environment and this adaptation changes the sound of the HAs. The parameter setting of the HA can be adjusted to make the sound more or less comfortable, audible, intelligible etc. Nevertheless, the subjective preference of the individual user is not systematically taken into account when optimizing the HA sound.

In the present study, a preference-based individualization system is investigated, where the setting is optimized based on a minimum of subjective preference assessments. In principle, though, other listening strategies than preference are also applicable. The preference assessments are performed in an iterative 2AFC-like paradigm, where the user subjectively grades his/her degree of preference between two settings while listening to a particular sound stimulus through the HAs. Following each preference grading, an estimate of the unknown and user-specific internal representation of preference as a function of parameter settings is updated using sophisticated statistical modeling. Conceptually, the internal representation associates a value to each parameter setting. Each value describes the perceived quality of the particular setting. Besides the estimate, the statistical modeling approach provides the uncertainty in the estimate as a function of parameter settings. In the present work, the estimate and the uncertainty are combined in a principle statistical manner to provide an information-gain criterion as a function of parameter settings. Based on this criterion, the two best next test conditions are identified by the algorithm. After a pre-defined number of iterations, the parameter setting for which the resulting internal-representation estimate has the highest value, reflects the subjectively best feature setting. Ideally, this approach thus promises fast HA optimization to any individual listening strategy in a given listening situation.

In the present work, estimates of the internal representation of preference for individual hearing impaired users are found. The results indicate
that the procedure converges to a stable and realistic estimate of the internal representations for individual users within reasonable time. The estimate differs from user to user and may show both local and global optima. A subjective post-evaluation furthermore shows an overall preference for the global optimum. Finally, an impression of the reproducibility of the subjectively optimal parameter setting is provided by comparing test/re-test results.

B32

**Proposed 12-item form of the Speech, Spatial and Qualities of Hearing scale (SSQ12)**

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The SSQ (Gatehouse & Noble, 2004), version 5.6, is serviceable a complete 49-item inventory. A recent factor analysis (Akeroyd et al., in preparation) broadly confirms its three main subscales. Gatehouse & Akeroyd (2006) derived 10 “pragmatic” subscales, determined on the basis of commonality of themes. We undertook independent scrutinies to identify 12 items constituting a short form that would have usefulness in clinical and research communities. The choice of 12 provides a size match with the 12-item “handicap scale” devised by Gatehouse & Noble (2004). Care was taken to have items from the three main subscales, and to cover as many as possible of the “pragmatic subscales” — nine of the ten have been represented. Our presentation shows how the 12 items were selected, and provides preliminary indications of how the SSQ12 performs with data extracted from studies using the full SSQ. Relations to prospective short-form SSQ inventories from other research groups are also discussed.

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**Objective analysis of reconstructed sound-fields using higher-order Ambisonics for hearing aid applications**

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The performance of hearing aid technologies is typically assessed by conducting either listening tests in a laboratory or real world field-studies. In the laboratory, a few loudspeakers are usually used to simulate single target and noise sources inside a low-reverberant playback room. Such studies are highly controlled and easy to realize, but the results often do not reflect real world performance. In contrast, field studies provide ecologically valid results but are time consuming and difficult to control. Hence, in order to conduct more ecologically valid listening tests inside the laboratory, the reproduction of realistic sound environments is required.

The concept of higher-order Ambisonics (HOA) has been widely used for loudspeaker-based sound field reproduction. Applying such an approach, it is possible to accurately reproduce a simulated or recorded sound field in a certain area inside a loudspeaker array. However, the size of this “sweet spot” decreases with increasing frequency but increases with increasing Ambisonics order, which in turn demands an increasing number of loudspeakers. Considering this limitation, it is unclear how far HOA can be applied for evaluating hearing aid technologies.

In the present study, a set of objective measures was utilized to determine the minimal required Ambisonics order for evaluating hearing aid applications. A set of head related transfer functions (HRTF) to both the in-ear microphones of a dummy head and the microphones of behind-the-ear (BTE) hearing aids were measured from 1784 points covering a sphere. Those transfer functions were then used in a simulation framework to calculate the average response errors introduced by HOA of different orders. The directivity patterns of the BTE hearing aids, which are highly phase-sensitive, were also compared between an ideal sound field case and its equivalent Ambisonics representation. For an anechoic playback room it was found that for a desired bandwidth B the required Ambisonics order M is about B ≈ M*600Hz, which for hearing aid ap-
applications is only practical for 2D sound reproduction. However, since the free-field case provides the worst-case scenario and is rather unrealistic, the behavior of the Ambisonics reproduction of simulated room responses was additionally investigated. By taking into account the natural variation of room responses across different source receiver locations it was found that the required Ambisonics order is significantly reduced. The results suggest that HOA can be applied for evaluating hearing aid technologies, at least when the sound reproduction is restricted to the horizontal plane.

Pitch and melody perception of simultaneous harmonic sounds in normal and impaired hearing
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Many everyday situations involve hearing out harmonic sounds, such as voiced speech or musical notes, and following them over time in the presence of other harmonic sounds. Despite decades of research on pitch perception, it remains unclear whether the ability to hear out the pitch of one harmonic sound in the presence of others is limited by peripheral frequency selectivity or by other factors, such as phase-locking to the temporal waveform of sounds. In this study a direct test of this hypothesis was undertaken by examining the relationship between measures of frequency selectivity and measures of performance in pitch- and melody-discrimination tasks in normal-hearing and hearing-impaired listeners. Although some measures of frequency selectivity, such as spectral ripple discrimination and forward masking by harmonic complexes, predicted melody discrimination in the presence of competing harmonic sounds, other more traditional measures, such as filter shapes derived from simultaneous masking using notched noise, did not. The results provide insights into the role of frequency selectivity in hearing pitch against complex backgrounds. [Work was supported by NIH grant R01DC05216; subject recruitment was facilitated by Starkey Laboratories, Inc.]

Detection of hearing aid directionality with sound field speech tests
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Two types of speech tests available in Norway were evaluated in their performance to discern between hearing aids with omnidirectional and directional microphones.

One of the deployed speech tests was the Norwegian version of the standard HINT test. The HINT tests use natural sentences. Sound field measurements were performed in the traditional HINT conditions: quiet, noise front, noise right, noise left plus a special condition where the listener was placed between the two speakers in order to accomplish a measurement with noise back.

The other speech test deployed was part of “HiST speech audiometry”, the Norwegian speech audiometry test developed at our institution. The test deployed in this study uses three-word utterances of the structure number-adjective-substantive. The test is administered with standard surround sound DVD equipment deploying lists of 10 three-word utterances where each new utterance has subsequent reduced level until it becomes unintelligible. The thresholds can be calculated from the number of words recognized in each list similar to the QuickSIN test. The five speakers are arranged with the following azimuth positions: 0°, ±45° and ±135°. Six sound fields conditions were used for this test: quiet, noise front, noise ±45° and ±135°, noise ±45°, noise ±135° with the speech administered through the centre speaker (0°) and finally a noise back condition with speech from the front left speaker (-45°) and noise from the surround right speaker (+135°).

The measurements were performed in three different conditions: unaided, with a set of Oticon Epoq XW RITE hearing aids programmed with omnidirectional microphones and with the same hearing aids programmed with directional microphones. The hearing aids were used with open domes. Automatic features of the aids were deselected. The simulated gain of the aids was approximately +10 dB for frequencies greater than
1000 Hz. The measurements were performed in audiometric cabins. The measurements were done by 17 of our second year audiologist students as a laboratory exercise and they altered between the role as a tester and a test subject.

The results as signal-to-noise ratios and spatial release from masking will be presented. Naturally the measurements with noise back conditions are best to discern between omnidirectional and directional microphones. The results are used to discuss how suitable this test method is to classify the directionality of each test subject’s hearing aid.

ASSR stimuli were generated using MATLAB (sample rate 32 kHz) and presented through ER3A insert earphones. Response amplitude curves (RACs) were recorded over the frequency range of 500 Hz to 4000 Hz to 2 kHz carrier tone, 100% exponentially amplitude modulated (modulation rate of 81 Hz) presented in the TEN noise with spectral notch (600 Hz bandwidth) swept across the spectrum. The signal level was 70 dB HL. RACs were recorded twice (on separate two sessions) from ten normally hearing subjects at signal-to-masker ratios subject-specific based on normative data. We identify the point that gives the largest auditory steady state response (ASSR) amplitude to an amplitude modulated pure tone. Preliminary analysis suggests that the new auditory test could aid both the speed of data collection and ease of DR identification.

References:

Characterization of IHC loss and its relevance to hearing aid gain prescription rules
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Sensorineural hearing loss (HL) like the common age-related hearing loss typically results in elevated thresholds and steepened loudness growth with level (recruitment phenomenon). These observations are typically associated with loss of sensitivity and loss of compression, significantly conditioned by loss or dysfunction of outer hair cells (OHC). In hearing aids, amplification and dynamic range compression aim at compensating the described deficits. Nevertheless, success usually shows large inter-individual variability and hearing-impaired (HI) listeners generally still have considerable problems in complex acoustic communication situations including background
noise, reverberation, and multiple speakers. One reason might be related to loss or damage of inner hair cells (IHC) including subsequent retrocochlear neuronal processes at early stages of the auditory pathway. Such damage would affect the “neural coding fidelity” of the sound waveform resulting in loss of temporal fine-structure information at low frequencies and temporal envelope information at higher frequencies. It appears plausible that such deficits could persist even if audibility and loudness perception are restored by suitable compensation measures. Moreover, seemingly appropriate hearing aid gain in one frequency region might even swamp temporal information in a remote frequency region due to spread of excitation. Therefore, it seems reasonable to distinguish between damage of OHC associated with low-level gain loss and reduced compression, and IHC damage associated with linear gain loss and supra-threshold processing disorders in order to find an optimal compensation strategy. However, proven diagnostic measures are neither at hand nor are the outcomes of suggested measures easily comparable without proper auditory models. In a recent study, Ewert and Grimm (2011) predict gain loss (HL caused by OHC loss) from audiometric thresholds combined with the steepness of the categorical loudness scaling function. The assumption was that the total HL is the sum of OHC and IHC induced HL. IHC induced HL was deducted from the OHC loss estimates. Here, a series of four psychoacoustic measurements is presented that aim at directly quantifying IHC damage in a mixed group of young and elderly NH and HI listeners involving temporal fine-structure detection and discrimination. The experiments are conducted at absolute and masked (supra) threshold levels. The data is analyzed on the basis of the pre-processing of an auditory model (Jepsen et al., 2008). The aim of the study is to improve OHC and IHC loss diagnostics and hearing aid gain prescription based on these measures.

References:

B38

Speech perception problems of the hearing impaired could arise partly from their inability to recover envelopes from temporal fine structure
Jayaganesh Swaminathan, Charlotte M. Reed, Louis D. Braida, Lorraine A. Delhorne and Joseph G. Desole
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Listeners with sensorineural hearing loss have a deficit in their ability to use acoustic temporal fine-structure (TFS) cues. A straightforward interpretation of this deficit is that it is due to degradation in the ability of auditory-nerve fibers to phase lock to TFS; however, neurophysiological evidence indicates that phase locking is not degraded following noise-induced hearing loss. Alternatively, the loss of recovered envelopes could also contribute to the reduced ability to use acoustic-TFS cues. When a broadband signal is filtered through a set of narrow analysis filters, the TFS component gets converted into recovered envelopes at the output of the filters. Broadened cochlear tuning, often observed with sensorineural loss, will result in a reduction in envelope recovery from TFS. The present study examined the role of recovered envelopes in the perception of TFS with normal-hearing (NH) and hearing-impaired (HI) listeners.

Speech materials (16 consonants in /a/-/C/-/a/ syllables) were first filtered into 16 adjacent frequency bands. Within each band, the TFS component was retained and the envelope component was discarded. The band signals were then recombined to create narrowband TFS speech and presented to NH and HI listeners for identification. To assess the role of recovered envelopes on the perception of narrow-band TFS, an envelope-vocoded speech version of the narrow-band TFS speech was also created (RSPEECH). The effect of training on the identification of narrow-band TFS and RSPEECH was assessed with NH listeners.

Preliminary results showed that: 1) without training, both NH and HI listeners performed very poorly with narrow-band TFS speech; 2) after training, the identification scores of narrow-band TFS speech for NH listeners improved moderate-
ly to about ~45%; 3) after training, the performance with RSPEECH was comparable to narrow-band TFS speech for NH listeners suggesting that performance with narrow-band TFS speech may be accounted for by recovered envelopes rather than acoustic-TFS per se.

These results suggest that the interpretation of previous studies that have used TFS speech may have been confounded with the presence of recovered envelopes. Hence, the inability of HI listeners to use acoustic-TFS cues may have risen, at least partly from their inability to recover envelopes from TFS. A framework for interpreting these results, as well as their implications for the design of auditory prostheses, will be discussed. [Supported by the NIH/NIDCD.]
Posters for Session C should be put up by 8:00 AM Saturday, August 11, and taken down after 10:00 PM Saturday, August 11 or before 7:00 AM Sunday, August 12. Presenters should be at their posters from 10:10 AM – 11:25 AM; 8:30 – 10:00 PM.

**POSTER SESSION C**

*Saturday 10:10 AM to 11:25 AM*

**C1**

Impaired spatiotemporal coding of vowels is exacerbated by noise

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Hearing aids can restore some degree of hearing to many patients. It has been shown that both neural rate-place and temporal-place vowel coding are degraded with impairment and that these representations can be improved with appropriate amplification. However, the ability to understand speech in noisy conditions remains impaired relative to normal hearing. More recently, we have shown that spatiotemporal coding is also degraded with noise-induced hearing impairment. This is seen as an increase in correlated activity across cochlear place and an associated decrease in traveling wave delay. Although spatiotemporal coding is believed to be important for a number of perceptual phenomena, the effects of amplification on spatiotemporal coding of speech have not been explored.

The current study aims to quantify the degradation in spatiotemporal coding of a vowel in quiet and in noise. We measured auditory nerve responses to the vowel /e/ in chinchillas with both normal hearing and noise induced hearing loss. Through the use of a spectro-temporal manipulation procedure, we predicted the responses of nearby nerve fibers. Spatiotemporal coding was quantified in terms of cross-fiber correlation and characteristic delay near the cochlear best places for the first two formants of the vowel. Vowels were presented between the threshold and saturation levels for each nerve fiber, and speech shaped noise was adjusted to either an equal rate response (approximating equal sensation level) or an SPL equal to that of the vowel (0 dB SNR).

Consistent with previous research, our results indicate that sensorineural hearing loss reduces the traveling wave delay between nearby fibers. Furthermore, our results suggest that normal traveling wave delays may be somewhat larger (i.e., enhanced spatiotemporal cues) when noise is added, but this enhancement was not observed in the impaired data. These results indicate that spatiotemporal coding is degraded with hearing impairment, particularly so in background noise. We are currently collecting data to measure the effects of hearing-aid amplification on spatiotemporal coding, and preliminary results suggest that amplification does not improve spatiotemporal coding of vowels in noise. Future work will focus on further quantifying the effects of amplification, and developing a signal processing strategy to improve the spatiotemporal representations of speech. Such a strategy may be useful for improving speech perception in noise for hearing-impaired listeners. [Supported by NIH F31-DC010966 (JB) and NIH R01-DC009838 (MH).]

**C2**

Evaluation of a health literacy workshop on Hearing Health in Older Adults

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In a previous investigation (Winsor, Jenstad, Purves, & Sims-Gould, in prep), we found that lack of access to unbiased information about hearing and hearing aids was a barrier to older adults seeking hearing help and obtaining hearing aids. In response to this need for hearing health information, we worked with a seniors’ organization (Council of Senior Citizens’ Organizations; COSCO) to create a workshop on “Hearing Health in Older Adults” that could be presented by seniors to their peers. The workshop content was developed from an evidence-based review of the literature, modified by discussions with inter-
disciplinary groups of health-care professionals and students, adult clients of hearing clinics, and older adults in the community. The workshop format was based on COSCO’s suggested template. It is COSCO’s mandate to conduct health literacy transfer seminars for seniors that include the key principles of peer-teaching-peer and participatory action learning, with the intent of creating change in seniors’ behaviour (Grosjean, Pither, Kube & MacLeay, Researching Transitions in Lifelong Learning, 2009). There is a need to investigate whether this format of learning will lead to a change in hearing health behaviour by older adults (i.e., seeking help for hearing loss or obtaining hearing aids).

The purpose of this study was to investigate the impact of the educational workshop “Hearing Health in Older Adults,” in terms of both its efficacy as an information-sharing tool and as an impetus towards taking steps for one’s hearing health. The impact is being investigated through minimally-led focus-group discussions and thematic analysis of the discussion transcripts to provide insight into the following specific areas: (1) What works and what needs to be changed in the workshop content or format to best meet the audience’s needs?; (2) What are the next steps people report workshop attendees would be willing to consider or do for their hearing health as a result of attending this workshop?; and (3) What are the reported barriers and facilitators for taking the next steps? We will present preliminary findings of these outcomes.

The international outcome inventory for hearing aids for the significant other: Results from partners of Veterans

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Rationale/Purpose: The International Outcome Inventory (IOI-HA) is a brief questionnaire that can be used as a supplement to hearing-aid studies to evaluate the effectiveness of treatment. It is also routinely used in the VA as a hearing-aid outcome measure. In 2002, Noble extended the IOI-HA to significant others (IOI-HA-SO). The purpose of this analysis is twofold: 1) assess the psychometric properties of this questionnaire using partners of Veterans with hearing loss and 2) determine the utility of IOI-HA score for predicting the IOI-HA-SO score for these same couples.

Methods: Subjects included Veteran patients with bilateral hearing loss that were fit bilaterally with hearing aids in the Portland VA Medical Center Audiology and Speech Pathology Service and their live-in partners. The subjects with hearing loss completed the IOI-HA, while their partners completed the IOI-HA-SO. Administration of these questionnaires typically occurred 4-6 weeks after the hearing aids were fit.

Results/Discussion: Data were analyzed on 80 couples, of which all patients with hearing loss were male and all live-in partners were female. The mean age of the patients with hearing loss was 67.4 years, while the mean age of the partners was 64.6 years. The mean four-frequency pure-tone average for the patients with hearing loss was 36.4 dB HL for the right ear and 39.2 dB HL for the left ear.

The principal factor method was used to extract factors for the IOI-HA-SO, followed by varimax rotation. A scree test indicated two useful factors, which accounted for 82% and 21% of the common variance. Scale reliability was assessed using Cronbach’s alpha. The overall scale reliability was 0.7. Items 2, 3, 4, and 7 belonging to Factor 1 had a relatively higher Cronbach alpha of 0.78. Cronbach alpha for items 1 and 6 belonging to Factor 2 had a considerably lower value (0.43).

The utility of the IOI-HA score for predicting the IOI-HA-SO score was evaluated by fitting a linear regression model with the total IOI-HA-SO score as an outcome variable and the total IOI-HA score as a predictor. The coefficient for the IOI-HA score effect was highly significant (estimate = 0.68, SE = 0.1; p<0.0001), giving an $r^2$ of 0.37. 95% prediction limits based on the fitted model indicated that, within the observed range of IOI-HA total scores, partners can be expected to score within about ±5 points of the patient. [This work was supported by the VA Rehabilitation Research and Development Service]
A study on effective gain of hearing aids in cochlea dead region
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A sensorineural hearing loss (SNHL) occurs when the cochlea in the inner has functional problem. The region in the cochlea with no (or very few) functioning inner hair cells or neurons called ‘dead regions’. Amplification using hearing aid over a frequency range corresponding to a dead region may not be beneficial. Therefore, it was compared with various gain of hearing aid for the impaired who had dead region.

This paper had two purposes. First, it was aim to compare speech recognition with different location of dead region, such as low frequency dead region, middle frequency dead region and high frequency dead region. Second, it was for comparing which was effective gain for hearing aid with dead region in each frequency.

There were conditions of experiment. We used mel-filter bank, the number of filter bank was 35, and reported evaluation of location and gain of dead region. Subjects were 6 people who had normal hearing ability. The signal sound was mixed white noise and babble noise (SNR=5dB). And the test signal was applied to hearing loss in dead regions. In addition, dead regions were divided three types by frequency. Low-frequency dead region was defined under 1100Hz in frequency. Mid-frequency dead region was between 1350Hz and 2900Hz. High-frequency dead region was over 3100Hz. It was compared the result as various types of gain. Each gain dB was 6 dB and 20 dB.

We needed to considerate two points of view in this paper. The first thing was the result of test different with location of dead region. As the result of WRS, scores of low-frequency dead region was higher than mid-frequency dead region or high-frequency dead region. The second was comparing a gain in dead region. A score of WRS was high, when a gain was low in low-frequency dead region, high in mid-frequency dead region and high-frequency dead region.

[This work was supported by grant No. 10031764 from the Strategic Technology Development Program of Ministry of Knowledge Economy and grant No. SS100022 by Seoul R&B Program.]

Evaluating critical hearing abilities for Army retention standards: Estimating speech-in-noise performance
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Hearing is a critical skill for the survivability, success, and health of active duty service members. Hearing loss and the use of communication devices or hearing protectors that amplify/attenuate sounds can alter a service member’s functional hearing abilities. Current hearing evaluations for fitness for duty, which include a pure-tone audiogram and the speech-reception-in-noise test (SPRINT), are inadequate to predict speech understanding in various mission-relevant listening environments. The primary goal of this project is to develop and evaluate clinical tests that provide relevant information regarding speech communication capabilities of service members.

In addition to a computerized version of the SPRINT, modified versions of three speech-in-noise tasks were presented to listeners. (1) mWIN: modified Words-In-Noise (Wilson, 2003) task with different maskers (vehicle noise and a competing female talker). (2) mCAT: modified Call-sign Acquisition Task (Rao and Letowsky, 2006). (3) mQSIN: modified Quick-SIN test (Killion et al., 2004) in eight different listening conditions.

Young (≤50 years) listeners with H1, H2 or H3 hearing profile (U.S. Army 40-501) were tested. SPRINT scores for listeners with the most hearing loss (H3) were significantly lower than those for normal-hearing (H1) and near-normal-hearing (H2) listeners. All listeners obtained benefit of masker modulations and spatial release from masking in both mWIN and mCAT tests. H2 and H3 listeners showed less benefit of masker modulations compared to H1 listeners. As expected, the closed-set mCAT test resulted in low-
er SRTs and greater masking releases compared to the open-set mWIN test. In the mQSN test, all listeners obtained binaural masking release and benefited from visual speech cues. Most listeners found reverberant and time-compressed speech conditions to be the most difficult. On average, H3 listeners showed worse performance compared to H1 and H2 listeners. H3 listeners also chose a relatively higher SNR, when asked to judge the maximum masker level that they can tolerate without losing any speech information. Speech thresholds obtained in these tasks showed weak or no correlation with audiogram or SPRINT scores.

These results suggest that the speech perception abilities of service members depend on the types of speech materials they use in their missions and the types of maskers they encounter in their operational environments. Work is currently underway to determine a minimum set of clinical tests capable of predicting speech-in-noise performance in various military-relevant tasks. Once validated, these tests could also be used to measure functional benefit of hearing aids, hearing protection and communication systems.

**C6**

Signal-processing model of human cochlear suppression

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Suppression is a salient feature of cochlear mechanics and is a major contributor to psychophysical simultaneous masking. Suppression is evident in measurements of distortion-product otoacoustic emissions (DPOAEs) in subjects with normal hearing and in subjects with mild-to-moderate hearing loss. A time-domain model of suppression has been implemented by applying time-varying gain to each output of a gammatone filter-bank before subsequently summing these outputs. The filter-bank was designed to have a group-delay of approximately 4 ms at all frequencies. The specified influence of suppression on each time-varying gain is dependent on the instantaneous level of every filter-bank output in a manner based on measurements of DPOAE suppression tuning curves. Suppressive influences are evident in single-tone input/output level functions when cross-channel contributions are included in the gain calculation. When suppression threshold (ST) for a low-frequency suppressor is plotted as a function of ST for an on-frequency suppressor the slope of this function provides an unreliable estimate of the inherent compression ratio. Simulated STs plotted as a function of frequency resemble measured DPOAE suppression tuning curves.

**C7**

Comparison of Adaptive Versions of the CCT and the NST

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Starkey Laboratories, Inc.

The audibility of high-frequency information can impact a hearing-impaired listener’s ability to understand speech. Therefore, whenever a new technology emerges that claims to improve access to high-frequency sounds, it is important to be able to quantify the impact that the new technology has on speech intelligibility. Two tests that have been used for this purpose are the California Consonant Test (CCT) and the Nonsense Syllable Test (NST).

The CCT and the NST were both designed to be conducted at a fixed presentation level. Selecting a single presentation level that is appropriate for all participants and test conditions is challenging because there is likely a range in participants’ performance; and choosing a single test level means that some participants may perform at a floor or ceiling level for some of the test conditions. This is undesirable because it limits our ability to demonstrate improvements in speech recognition that are associated with different technologies.

In order to work around the challenge of choosing a single fixed presentation level that is appropriate for all participants and test conditions, we modified the CCT and the NST so that they could be performed in an adaptive manner. The goal of this study was to evaluate which of these two adaptive tests was better in terms of repeatability, variability and effect size. To allow both tests to
be used for a variety of applications—including differentiating the high-frequency benefit that directional hearing aids provide—we presented the stimuli to the listener in the sound field. The speech stimuli played out the 0° speaker, and noise played out the other 7 speakers (at ±45°, ±90°, ±135° and 180°).

Twelve normal-hearing individuals participated in this study. All participants were fitted bilaterally with directional hearing aids that had a particularly good Directivity Index (DI) in the high frequencies. Hearing aids were programmed for a mild, flat, sensorineural hearing loss. Participants were tested unaided and aided with the hearing aids in the omnidirectional and directional microphone modes. Multiple iterations were performed unaided to look for learning effects and to examine the repeatability and variability. Performance in the omnidirectional and directional microphone modes was examined for effect size. Recommendations will be made regarding the future use of adaptive versions of these tests.

C8
A model with compression for the estimation of speech intelligibility in quiet
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In Rhebergen et al. (2010), a perceptually motivated speech intelligibility model was introduced as an alternative to the standard ANSI SII model (the Speech Intelligibility Index) to predict sentence intelligibility in quiet and in noise for normally-hearing and hearing-impaired listeners. The novel aspect of this model was the introduction of cochlear compression in the estimations of the internal excitations from which the speech audibility was derived. In the present study this model is used to predict CVC scores in quiet as a function of the audiogram for a large clinical population (21676 ears). For this, predicted intelligibility functions were transformed to psychometric functions for CVCs using the scores of a normally-hearing reference group (197 ears). We compared predicted SRTs (Speech Reception Thresholds, i.e., 50% correct points) of the compressive model to those of the SII and of a linear version of the model. The results show that the novel model predicts the 50-percent correct points with greatest accuracy for normally hearing ears, moderately impaired ears, and intermediate ears, while the linear version of the model is the more accurate alternative for severely impaired ears. This indicates that the present model may still include too much cochlear compression for severe hearing losses.

C9
Investigation into the effects of caffeine on the auditory cortical response (N1-P2)
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Currently, auditory threshold detection is measured by behavioral audiometry which requires the active and honest participation of the patient. However, it can lead to inaccurate assessments owing to some patients having difficulty understanding the task (e.g., dementia). An alternative method of hearing sensitivity measurement is cortical electric response audiometry (CERA). CERA is an established technique that has been used since 1960’s. It is an objective method of identifying the brain’s response to auditory stimuli. One of the problems with CERA is the measurements are dependent upon the attentiveness of the individuals. There have been several attempts to improve the attentiveness for example reading or watching a movie.

We hypothesize that an overall increase in neural cognition would result in improved amplitude of the N1-P2 auditory response, resulting in more reliable measurements. It is known that caffeine interacts with neurotransmitters in the brain to promote mental alertness. The aim of this study was to investigate whether caffeine enhanced the CERA and improved the precision of CERA hearing sensitivity estimation.
A double-blind, placebo controlled crossover study was conducted, in which volunteers were instructed to abstain from caffeine for 12 hours prior to testing. Caffeine was given in the form of an orange flavored drink (Quick Energy, 59ml, 175mg of caffeine). The placebo was orange cordial mixed with water (60ml). Twenty-four volunteers attended the clinic on two separate occasions. Both conventional behavioral and CERA measurements were recorded at 1kHz, 3kHz, and 8kHz. The identification of CERA threshold was obtained using 20dB down, 10dB up and 5dB down protocol. Behavioral thresholds were obtained using the procedure recommended by the British Society of Audiology.

Linear regression analysis was carried out to determine statistical significance. Comparisons of the CERA and behavioral thresholds were made and the results showed that, with the step sizes used in the study, caffeine did not affect the behavioral thresholds. The difference between the CERA and behavioral threshold was measured for placebo and caffeine; it was found that there was not a statistically significant change when caffeine was introduced. When the amplitude of the CERA was analyzed it was found that caffeine increases the size of the response at threshold by 11% (p=0.025). Although this is a statistically significant increase it is deemed too small to justify employing caffeine when using CERA tests in the clinical environment.

Data were compiled from studies of normal and pathological middle ears, dating back to Voss and Allen (1994). AR measurements were taken using a middle ear acoustic power analysis system such as the MEPA3 (Mimosa Acoustics), or similar. This instrument provides complex acoustic reflectance and impedance measurements over the frequency range of 0.2 to 6.0 kHz via a probe placed in the ear canal. Acoustic data were then analyzed using pole-zero fitting, and compared to existing middle ear models.

Pole-zero fits were achieved with an RMS relative error of less than 5%. These fits typically had impedance characteristics in agreement with the data, and were well-behaved between the impedance and reflectance domains. Good (low error) fits were typically accomplished using 10 to 30 poles, with zeros of equal or 1 less order. Good fits were also achieved for reflectance data with the ear canal delay extracted, which allowed for better comparison across ears. It was observed that the locations of the poles and zeros in the s-plane differ between normal and pathological middle ears. Additionally, it was possible to see the effects of individual variation. For instance, individual AR magnitude variations for normal middle ears in the 1 to 4 kHz range (Allen et al., 2005, Rosowski et al., 2011) corresponded to pole-zero variations within a specific region of the s-plane.

This study establishes a methodology for examining the physical properties of pathological middle ears based on pole-zero analyses of the acoustic reflectance. Pole-zero modeling shows promise for characterizing AR data and identifying middle ear pathologies using a relatively low cost measurement system.
Effects of compression on amplification of one’s own voice: Investigation of real ear to coupler differences for varied degrees of hearing loss

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Comparisons of real ear to coupler differences (RECD) for one’s own voice while speaking (live speech) and in listen-only conditions (immediate replay of the live speech recording) were made using a hearing aid (HA) that was programmed for linear processing (LIN) or wide dynamic range compression (WDRC). As part of a larger study, five HA users produced sequences of the syllable /pa/ at a comfortable level and pace. Participants listened to the speech (both live and replay) using a laboratory HA programmed for LIN with all features disabled, and coupled to the participant's own earmolds. The participants had moderately severe sloping to severe (N=2) or severe sloping to profound (N=3) sensorineural hearing loss. A lapel microphone was placed above the left HA to capture the produced speech. A signal-processing device (SPD) then delivered the speech signal to the participant through direct audio input. Average speech levels at the lapel microphone varied across participants between 71 and 78 dB SPL. When using the prescribed gain required for the degree of hearing loss, spectral levels could exceed the saturation sound pressure level (SSPL). Therefore, the presented signal (live and replay) was attenuated by the SPD to avoid HA saturation. Real ear (RE) recordings of live and replayed speech, along with the speech microphone recordings, were stored and analyzed using Pulse (B&K). The same speech samples with the attenuated input levels, as well as the actual speech levels, were used to make the coupler recordings for both LIN and WDRC programs. The following comparisons were made: (i) RE-replay vs. Coupler LIN; (ii) RE live vs. Coupler LIN; (iii) Coupler LIN vs. Coupler WDRC; and, (iv) RE live vs. Coupler WDRC. In addition, comparisons were made between our obtained RECD values and the standard values. We investigated the accuracy of standard RECD corrections for deriving real-ear SPL of one’s own voice while speaking. Implications of compression on relative loudness of one’s own voice and consequently, the subject’s satisfaction with their voice quality will be discussed.

Sensitivity to dynamic range compression is predictable by distances between modulation spectra

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Dynamic range compression is a signal processing technique that is widely used to reduce the difference in level between the loudest and quietest portions of a signal. Such compression distorts the shape of the temporal (amplitude) envelope of a signal. Here we attempted to characterize the sensitivity of young adults with normal hearing to these distortions. Listeners were asked to distinguish between compressed and uncompressed speech that had first been passed through a single channel noise vocoder. To discourage listening strategies based on overall loudness, all stimuli were scaled to the same rms amplitude and the overall presentation level was randomized over a 6 dB range. All listeners were given a practice session that included feedback. On average, listeners were very sensitive to severe amounts of compression (> 90% correct at the most extreme parameter values), and performance decreased monotonically as the parameter values became less extreme. This sensitivity was well predicted (r = 0.93) by the Euclidean distance between the 6-band modulation spectrum of the uncompressed signal and that of the compressed signal. This correlation increased when distance computation only considered modulation in the 4 Hz band, likely owing to the fact that the speech signal had a peak in its modulation spectrum at that frequency. On the individual level, sensitivity to compression was well predicted (r = -0.87) by performance on a modulation depth discrimination condition using a 4 Hz sinusoidal modulator and a speech-shaped noise carrier. This correlation indicates that sensitivity to com-
pression can be predicted by performance on a simple psychoacoustic task. Overall these results indicate that a relatively straightforward model can be used to identify the compression parameter values that are perceptually distinct (assuming loudness is held constant). This model can be used to reduce the overall compressor parameter space when the goal is to manipulate timbre, or to identify the parameter changes that would not be perceivable when the goal is to preserve the timbre.

**C13**

**Aiding consonant identification with frequency-compression hearing aids for listeners with high-frequency cochlear dead regions**

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Listeners with extensive dead regions (DRs) in the cochlea often fail to get benefit from amplification above 1.7 times the edge frequency of the DR (fe). Frequency-compression (FC) hearing aids could convey speech information occurring above 1.7fe to people with high-frequency DRs. The aim of this study was to determine the best FC settings for consonant identification.

Listeners with high-frequency DRs completed a consonant identification task. The stimuli were VCVs made of one of three vowels (/a/, /i/, or /u/) and one of twenty-one consonants spoken by a female native British English speaker. The VCVs were recorded from a custom-fitted Phonak prototype FC hearing with a cutoff frequency equal to 1.7fe. FC only occurred if the short-term spectrum was dominated by the high frequencies. The FC kneepoint was either equal to fe (‘high’) or at 0.75fe (‘low’). The compression ratio was chosen to convey speech information occurring up to either 5 (‘low’) or 10 ERBN numbers above 1.7fe (‘high’). This gave four FC conditions (FC1, FC2, FC3, and FC4) which were tested weekly in a Latin-Square design. Some control lists (CA: conventional amplification) were also tested each time.

Results are available for four out of eight subjects. A two-way ANOVA with factors ‘knee-point’ and ‘compression ratio’ showed no significant effects.

In a second analysis, scores for each FC condition were compared with CA scores measured on the same day, by performing a one-way ANOVA. Subjects performed better for FC1 (low kneepoint, low compression ratio) However, the effect just failed to reach the significance level (F(1,3)= 9.08, p= 0.057). The other FC conditions gave overall scores similar to CA. Analysis of the confusion matrices showed that FC did change the pattern of confusions, even though this was not reflected on the overall scores.

Separate one-way ANOVAs for each subject showed that FC1 gave significantly better performance than CA for subjects 2 (F(1,7)= 8.10, p= 0.029) and 4 (F(1,7)= 24, p= 0.003). Indeed, three out of four subjects obtained the biggest difference with respect to CA with FC1. FC4 (high kneepoint, high compression ratio) gave significantly worse performance than CA for subject 4 (F(1,7)= 11.27, p= 0.015). Sequential information analysis showed that FC1 was the condition offering the greatest overall benefit compared to CA. [Supported by Action on Hearing Loss.]

**C14**

**What makes different ears sound different?**

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GN-Resound

Virtual sound is a technique, where sound is presented via headphones using HRTF (Head-Related-transfer function) based processing and perceived by the listener as coming from a loudspeaker located within a room. Using a small database of 7 test subjects HRTF’s and corresponding headphones to ear drum responses it is thus possible to listen through the outer ears of the database test subjects.

In this study, it is investigated if there is a correlation between a test person’s perceptual auditory impression of different ears in the database and the corresponding difference in a selection of spatial cues between the ears. The perceptual impression was evaluated by a virtual sound paired comparison test with reference, implemented in
Matlab, where all combinations of different virtual pairs of ears were tested at angles [0, 40, 90, 130, 180, 230, 270 and 320] degrees. The excitation signal was band limited white noise presented at 67 dB SPL.

The spatial cues under investigation were: Interaural level difference (ILD), interaural time difference (ITD) and monaural spectral difference.

It is found that ILD and ITD are not correlated with the results of the perceptual testing. Only the monaural spectral difference is significantly correlated at 4 out of the 8 presentation angles. This indicates that the most important cue describing the difference between pair of ears is the monaural spectral cue.

C15

Training effects in a German speech-in-noise test with original and fast speech
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Listeners can adapt to different hearing situations whether speaker, background noise or other factors change. This is also true for speech-in-noise tests which are useful tools for the evaluation of hearing aids and their processing algorithms. In this application, speech tests are frequently repeated and training effects have to be considered. Research shows decreasing speech recognition thresholds (SRTs, signal-to-noise ratio of 50% intelligibility) for the first presented test lists, that is why two training lists are recommended for a German speech-in-noise test (Oldenburg sentence test, OLSA, Wagener et al., 1999). Furthermore, Golomb et al. (2007) showed that participants also accustom to time-compressed fast speech. There are two assumptions regarding these training effects: They always occur during the first presentations of test lists within a session (intra session training). Also, training effects are more pronounced in the beginning but remain effective throughout all sessions of a study (inter session training). The present study investigates these training effects in the German speech-in-noise test OLSA with original and time-compressed speech. The fast speech was compressed to 30% of its original length. Two groups of young listeners with normal hearing repeated the test six times during five different sessions either with original or time-compressed speech. In general, the time between the sessions was two days. Additionally, a group of older hearing impaired listeners performed the measurements with the original speech. All participants had no experience with the test before. The results show inter and intra session training effects. For example, SRT values of the first two test lists in the first session are at higher values than all following results. Also, participants understood less in the last list of the first session than at the end of the fifth session. Training effects are largest for normal hearing participants, who listened to time-compressed speech, compared to normal hearing and hearing impaired participants, who listened to the original speech. Only the inter session training effects, measured with original speech, are larger in normal hearing than in hearing impaired participants. In summary, training effects can be reduced with two training lists at the beginning of testing and with randomization of the test conditions across sessions or groups.

References:

C16

Realistic signal-to-noise ratios
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Recently, the appropriateness and limitations of adaptive speech tests as outcome measures when evaluating hearing-device features have been discussed. At IHCON 2010, Naylor gave a presentation which outlined the main reasons why such testing could be problematic. The obvious alternative to adaptive speech testing is to test at a fixed signal-to-noise ratio (SNR), but there are
issues that have to be solved. For example: 1. Which SNRs should be used? Unfortunately, there is a lack of studies where realistic SNRs have been investigated. 2. Which type of test (procedure and test material) should be used? The reason why adaptive speech tests have become so popular is that they are easy to perform. Floor and ceiling effects are avoided, and the researcher “always” gets a result that can be used for further analyses.

The current project deals with the first of these issues, the estimation of realistic SNRs.

In a previous study (Wagener et al., JAAA 19, 2008), 20 hearing-aid users made binaural recordings in everyday environments. In a subsequent laboratory study, the participants performed subjective assessments of their own recordings. These assessments showed that the participants had recorded common and typical situations, which they judged to be important.

The data by Wagener et al. were used in the current study. Only recordings where speech was the target were included and recordings were excluded if they contained recording artifacts, uncommon noise situations, or situations where the noise characteristics changed considerably over time. In total 72 situations were analyzed.

After abandoning automatic noise estimation procedures, a manual estimation method was used. For each situation, speech-plus-noise segments were matched (by listening) with similar segments of noise-only. For each ear separately, SNRs were estimated based on power estimates of these segments. Overall RMS levels were also calculated and an accuracy measure, developed within the project, was used. Each estimated SNR can therefore be presented together with the corresponding RMS level and the associated accuracy. Frequency-specific SNRs (1/3-octave bands) were also calculated. In the presentation, the methodological considerations, in particular the assumptions underlying the estimation method, will be presented.

Results, both overall and frequency-specific SNRs, will be presented for the following noise categories: “quiet”, babble, noise from cars and public transport, kitchen noise, music, and radio/TV. The range of SNRs found in the material was large. The estimation accuracy was generally good, but got worse at negative SNRs.

C17

A speech enhancement method using modified IMCRA based on noise classification

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In speech signal processing for hearing aid and speech codec, a noise power estimation algorithm is crucial part to enhance the speech signal under noisy environments since the noise power have been used in many speech enhancement algorithm (MMSE, MS, IMCRA, and so on.). Among them, IMCRA algorithm has better performance of the noise power estimation while the complexity is lower. In this paper, we propose a novel method to improve the performance of the improved minima controlled recursive averaging (IMCRA). The conventional IMCRA algorithm efficiently estimate the noise power by averaging past spectral power values based on a smoothing parameter that is adjusted by the signal presence probability in frequency subbands. Since the minimum of smoothing parameter is defined as 0.85, it is difficult to obtain the robust estimates of the noise power in non-stationary noisy environments that is rapidly changed the spectral characteristics such as babble noise. For this reason, it is proposed that the modified IMCRA, which adaptively estimate and update the noise power according to the noise type classified based on Gaussian mixture model (GMM). To classify the noise environment, feature vectors of GMM consisted of spectrum centroid (SC) and spectral entropy (SE) and order of Gaussian mixture used 16. The motivation for using Gaussian mixture densities for the noise environment classifier stems from the observation that GMM can efficiently represent the SE and SC based on the statistical distribution by a position (mean vector) and an elliptic shape (covariance matrix) through the use of a discrete set of Gaussian functions (mixture weights).

The performances of the proposed method are evaluated by the ITU-T P.862 perceptual evaluation of speech quality (PESQ) and composite
measure under various environments (babble, white, car). The noise were added to the clean speech signal at 0, 5, 10, 15 dB SNR. From the various test, it is verified that the proposed algorithm based on noise classification yield better results compared to the conventional IMCRA-based scheme. In particular, in the non-stationary noise environment, the proposed method produced a remarkable improvement compared to the IMCRA. [This work was supported by grant No. 10031764 from the Strategic Technology Development Program of Ministry of Knowledge Economy.]

C18
Evidence of degraded amplitude modulation detection due to exposure to intense sounds

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Prolonged exposure to moderately intense sounds may produce damage mainly to outer hair cells (OHC), but exposure to very intense sounds may produce damage to the inner hair cells (IHC) as well. Absolute threshold, the conventional clinical measure of hearing function, gives no indication as to possible sites of damage and may not reflect early damage. Additional measures, such as otoacoustic emissions, may indicate OHC dysfunction while the TEN(HL) test gives an all-or-none measure of IHC function. There is no accepted method for detecting early/mild IHC dysfunction.

OHC dysfunction tends to produce increased sensitivity to amplitude fluctuations, because of the loss of cochlear compression, while IHC dysfunction leads to noisier coding and hence may lead to reduced sensitivity to amplitude fluctuations. In a test of the ability to discriminate envelope fluctuations of signals presented at low Sensation Levels (SLs), subjects exposed to high noise levels, such as rock musicians, showed poorer performance than control subjects (Stone et al., 2008). This was interpreted as reflecting reduced IHC function. However, due to the long training required to obtain stable results, the test of Stone et al. was impractical for clinical use.

Little training is required to give stable detection of sinusoidal amplitude modulation (AM). Here, 32 young normal-hearing subjects (mean age 21 yrs) were recruited into one of two gender-balanced groups. One group had low noise exposure and the other had at least bi-weekly exposure at rock/clubbing events. Absolute thresholds and AM detection (25-Hz rate, SLs of 10, 25 and 40 dB) were assessed at 3, 4 and 6 kHz, frequencies commonly associated with evidence of noise-induced hearing loss in the audiogram. Only at 6 kHz was the absolute threshold significantly higher for the noise-exposed group. At 10 SL, noise-exposed subjects were generally poorer than control subjects at AM detection, the largest difference occurring at 3 kHz. The difference across groups also approached significance at 3 kHz at 25 SL. This pattern of results, consistent with the earlier report, suggests that high-level exposure at rock/clubbing events leads to IHC dysfunction, and that this dysfunction can be identified via the detection of AM at low SLs, even when the audiogram is unaffected.

Reference:

C19
Design and evaluation of a bio-inspired compression system

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We investigated possible benefits from using a hearing-aid compression system that shared features in common with the normally functioning auditory system. The proposed system used fast-acting compression and compression channels that were non-uniformly scaled in terms of center frequency and bandwidth. Furthermore, this novel compressor used level-dependent channels, i.e., channels that widened with increasing sound level.
Benefits of this system were explored in three separate experiments including normal-hearing (NH) listeners and aided as well as unaided hearing-impaired (HI) listeners. Performance for the bio-inspired compressor was compared to that obtained with more conventional compressors. The first experiment tested psychoacoustic loudness summation as a function of noise bandwidth and noise level at 1 and 3 kHz by matching the loudness of narrowband and wideband noise stimuli. The second experiment assessed subjective quality using pairwise comparisons and the last experiment tested identification of consonants and vowels in quiet, in speech-shaped noise, and in a three-talker background.

The unaided listeners in the first experiment showed less loudness summation than the NH listeners, presumably due to reduced cochlear compressive gain and degraded frequency selectivity. The aided results indicated that both loudness summation as a function of noise bandwidth and noise level could be restored close to normal only by the bio-inspired compression system. However, performances in the subjective quality and consonant and vowel identification experiments did not differ significantly between the new system and the more conventional compressors tested. The present results inform the development of future loudness models and advanced compensation strategies for the hearing impaired.

**C20**

**Identification of conflicting-cue vowels in listeners with an electrically stimulated ear and an acoustically stimulated ear**

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There is a growing number of “bimodal” listeners who have a cochlear implant in one ear and receive acoustic input from the contralateral ear (typically from a hearing aid). This arrangement raises interesting questions regarding the mechanisms listeners use to integrate potentially disparate information from the acoustic and electrical ears. In this study we used “conflicting cue” vowels to determine whether bimodal listeners identify vowels responding primarily based on acoustic cues, electrical cues, or different combinations of both kinds of cues. Stimuli included three synthetic vowels (/i/, /a/, and /u/) as well as all six possible “conflicting cue” vowels, where one vowel is presented to the acoustic ear and a different one to the electrical ear. There were nine possible responses representing the most common monophthongal English vowels. When presented with conflicting cue vowels, some listeners responded based largely on the acoustic input, and others responded based largely on the electrical input. There were also some listeners whose response patterns were more complex, e.g., consistently responding /u/ when the acoustic stimulus was /i/ and the electrical stimulus was /a/. This result is reminiscent of the McGurk effect, obtained when the audio and visual parts of a stimulus encode different phonemes, except in our case it is the electrical and the acoustic stimuli that encode different vowels. We use these data, combined with mathematical modeling of vowel identification by bimodal listeners, to help determine how acoustic and electrical information are integrated by individuals with different amounts of residual hearing. The present study may provide insight into the different cognitive strategies bimodal listeners may use for electroacoustic speech perception.

**C21**

**Predicted effects of sensorineural hearing loss on across-fiber coding of temporal fine structure**

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Listeners with sensorineural hearing loss (SNHL) have been shown to have a deficit in their ability to use acoustic-temporal fine structure (TFS) cues (e.g., Lorenzi et al., 2006). With respect to monaural processing, such a deficit has been hypothesized to arise from both within channel/fiber and cross-channel/fiber processing of TFS information. A straightforward interpretation of a within fiber deficit is that it is due to degradation in the ability of auditory-nerve fibers to phase lock to TFS; however, neurophysiological evidence indicates the fundamental ability of...
AN fibers to phase lock is not degraded following noise-induced hearing loss (Kale and Heinz, 2010). Moreover, the loss of recovered envelopes has also been shown to contribute to the reduced ability to use within channel acoustic-TFS cues (Swaminathan et al., 2012). Alternatively, recent neurophysiological evidence suggests that there are across-fiber degradations in temporal coding following noise induced hearing loss that may underlie the perceptual deficit in using TFS cues with SNHL (Heinz et al., 2010; Kale, 2011). The goal of this study was to systematically evaluate the predicted effects of differential hair-cell dysfunction on across-fiber TFS coding.

Spike trains were generated from a physiologically based auditory-nerve model responding to broadband noise and a speech sentence. Across-fiber TFS coding was quantified in terms of a neural cross correlation coefficient and a characteristic delay (which provides an estimate of the traveling-wave delay). Correlations were also computed for hearing-impaired (HI) model versions that included selective outer-hair-cell (OHC) and inner-hair-cell (IHC) damage. The effect of sound level on across-fiber TFS coding was also quantified.

For both broadband noise and speech, neural predictions showed: 1) no predicted degradation in within-fiber phase locking for OHC damage, whereas a slight reduction was predicted for IHC damage; 2) broadened tuning associated with OHC damage was predicted to increase significantly the range of fiber separations over which correlated activity exists; 3) IHC damage was also predicted to produce a slightly broader correlated region, due to the higher sound level necessary to overcome the IHC loss, as indicated by the similar results from a normal-hearing fiber at the higher sound level; 4) broadened tuning was also predicted to reduce the traveling-wave delay between different CF fibers.

Thus, consistent with previous neurophysiological findings, OHC damage was predicted to degrade the normal spatiotemporal response pattern by producing a more coincident pattern across fibers. These results further highlight that the degradation in across-fiber coding could contribute, at least partly, to the inability of HI listeners to use acoustic-TFS cues. [Supported by the NIH/NIDCD.]

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**A study of dichotic-listening filters in binaural hearing aid**

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Increased spectral and temporal masking in the peripheral auditory system is assumed to be one of the causes of reduced speech intelligibility for sensorineural hearing impaired people. Dichotic listening, defined as listening to complementary filtered speech signals in each ear, has been proposed to cope with this problem. In a previous study, it was reported that dichotic listening was effective under conditions that a speaker was in front of the listener and background noise was not so loud. In the present study, we proposed a new dividing filter for dichotic listening, and examined the effectiveness of this technique on speech intelligibility under conditions that a speaker placed in oblique directions and a listener was surrounded by speech-like noise.

A listening test was conducted for 19 subjects with mild to moderate high-frequency sensorineural hearing loss. Subjects were placed on the center of six concyclic loudspeakers. Forty VCV syllables were presented from either in front of the subject (0 degree), right side (60 degree), or left side (300 degree). At the same time, speech shaped noises were presented from every six speakers (SNR were 20, 10, 0dB). Subjects wore the signal processor simulating hearing aids. Binaural signals were processed by five different filters as follows: [APF+APF] same as existing binaural hearing aid; [LPF+HPF] dividing signals into two bands using HPF and LPF; [APF+HPF] replacing LPF of Simple filter with APF; [Partial-1] dividing 300Hz to 3kHz of signals into two band, other bands are stereophonic; [Partial-2] suppressing 300Hz to 1kHz of signals in only one ear.

The results are as follows: 1) Overall, every dichotic filters improve consonant intelligibility. (Intelligibility was assessed with Ryan’s multiple comparison procedure); 2) When stimulus is presented form front or “HPF side” speaker, most filters are effective. When stimulus is presented from “LPF side” speaker, only “Partial-2” filter is effective; 3) When SNR is 20dB, every filter is
effective. When SNR is 10dB, “Partial-1” and “Partial-2” are effective. When SNR is 0dB, only “Partial-2” is effective.

C23

Predicting the quality of artificial and real nonlinearly distorted speech and music as perceived by hearing-impaired people

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The goals of this study were to characterize and model the perception of nonlinearly distorted speech and music by hearing-impaired listeners. Hearing-impaired listeners were asked to rate the perceived quality of speech and music that had been subjected to various forms of artificial and real nonlinear distortion. Some of the artificial distortions are inherent to hearing aid designs, and they include (1) hard and soft, symmetrical and asymmetrical clipping; (2) center clipping; (3) “full-range” distortion, produced by raising the absolute magnitude of the instantaneous amplitude of the signal to a power (alpha not equal to 1), while preserving the signal of the amplitude; (4) automatic gain control (AGC); (5) output limiting. These artificial distortions were implemented in both broadband and band-limited conditions. The real nonlinearly distorted speech and music were recorded at the outputs of 3 different compression hearing aids using a “KE-MAR” dummy head with different input level, compression ratio, compression speed and output limiting settings. Previous recordings at the outputs of 4-5 different mobile telephones were included. The recorded outputs were each digitally filtered so that the long-term spectrum of the output matched that of the input as closely as possible. To match the amplitude-frequency response of both real and artificial nonlinear distortions as closely as possible, all stimuli were bandpass-filtered between 300 and 5000 Hz. Stimuli were subjected to frequency-dependent amplification as prescribed by the “Cambridge formula” before presentation via Sennheiser HD580 earphones. For the artificial distortions in broadband and band-limited conditions, the pattern of the ratings was reasonably consistent across subjects and was similar to that for normal-hearing listeners. However, the mean ratings were not lower with increasing amount of soft or center clipping or when the compression ratios of the AGC and output limiting were increased. The deleterious effects produced by these nonlinear distortions may have been offset by the beneficial effects of improving audibility and compensating for loudness recruitment. We combined some of the well-controlled artificial distortions and real distortions by hearing aids and other communication devices to refine the prediction of perceived quality of our developed models and examined the efficacy of the model in accounting for the reduced frequency selectivity that is typically associated with hearing impairment. Further work on the model will be focused on accounting for the perception of band-limited distortion by hearing-impaired people. [Work supported by Deafness Research Foundation.]

C24

Children with hearing aids show diminished spatial masking release

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Problem statement: Spatial segregation of a speech target and masking noise improves intelligibility for listeners with normal hearing. In this experiment the phenomenon was examined for children with hearing loss who used hearing aids or cochlear implants.

Method: Second-grade children participated: 28 with normal hearing (NH), 6 with hearing aids (HAs), 15 with bilateral cochlear implants (CI-CI), and 9 with a unilateral cochlear implant (CI). Listeners repeated 180 unique consonant-vowel-consonant words presented in 0 dB masking white noise in a sound isolation booth. In one condition, the speech and noise signals were presented from the same speaker at 0° azimuth. In another condition, the speech signal was presented at 0° azimuth and the noise was presented at 90° azimuth on the side of the left ear for chil-
Children with NH, HAs, and CICI who had simultaneous implantation. Noise was presented on the side of the second CI for children with sequential implantation and the unimplanted ear for children with one CI. Test sessions were audio/video recorded and scored later for percent correct phonemes and whole words. Additionally, differences in segregated and same signal/noise presentation for correct phonemes and whole words were used as metrics of the magnitude of spatial masking release (SMR).

Results: Children were compared across four groups: NH, HA, CICI, and CI. For phoneme and whole words, order of correct recognition from best to worst was: children with NH, children with HAs, children with CICI, and lastly children with CI. Children in all groups recognized more phonemes and complete words correctly in the spatially separated condition than the spatially same condition. However, the magnitude of the SMR difference between conditions depended on the listening group. Children with NH and CICI showed a larger SMR: 9.4% and 9.3% differences for phonemes and 15.1% and 9.1% differences for words, respectively. The SMR for children with HAs and CI was diminished in comparison: 4.1% and 5.4% differences for phonemes and 5.6% and 5.6% differences for words, respectively.

Conclusions: Although HA listeners had better phoneme and complete word recognition than listeners with cochlear implants, their SMR was smaller than children in the CICI group and similar to children in the CI group. [This work was supported by a grant from the National Institutes of Health, National Institute on Deafness and Other Communication Disorders, R01 DC-006237.]

C26
Perceptual testing with virtual sound
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Virtual sound is a well-established technique where external sound sources up to approximately 6 kHz can be reproduced with headphones. If the technique is based on the user’s own head related transfer functions (HRTFs) it is possible to create a complex realistic virtual sound environment, using only a PC and a pair of headphones.

We present the first test results based on virtual sound where hearing aid algorithms can be tested in a virtual hearing-in-noise-test (HINT) environment. The method is based on measurements of horizontal plane HRTF’s which takes approx-
imimately 20 minutes for each test subject when the angular resolution is 10 degrees.

First, it is tested if the virtual HINT has the same performance in terms of speech-reception-threshold (SRT) values as a normal HINT using real speakers. Four speakers at [-90,0,90,180] degrees are used where the center speaker at 0 degrees is transmitting speech sentences from the DanishHINT developed at the Technical University of Denmark. It is found that the real HINT and the virtual HINT are not significantly different in a statistic sense and that the deviation in mean performance is on the order of 1 dB. Furthermore, it is found that the lack of visual cues (there are no speakers present) in the virtual HINT, does not affect performance more than 1 dB.

Secondly, the virtual HINT is repeated on three normal hearing subjects using either: a) the test subjects natural hearing (personal HRTFs), b) a virtual BTE device in omni mode (only the front microphone), c) a virtual BTE device in directionality mode. The virtual HINT is extended to also test for target speech at [0, 90, 180] degrees. The corresponding mean SRT values with standard deviations are: a) -4.0 +/- 1.2 dB, b) -3.2 +/- 1.9 dB, c) -3.2 +/- 1.4 dB. These results are not statistically different but they suggest that the test persons perform better in a noisy environment when using their own natural hearing.

**Study design:** After a review of literature data indicating a strong link between noise reduction and channel selection in cochlear implants we will present data from a study investigating the effects of noise on performance of normal hearing vs cochlear implant subjects. In this study we prepared stimuli in a Matlab model simulating a CI speech processor and either streamed these stimuli to CI recipients via the NIC™ streaming interface, or to normal hearing subjects via a CI vocoder simulation played on a loudspeaker. Sentence in noise testing was used to determine the SRT in different conditions. The conditions include simulations where noise was only allowed in gaps in the speech, either broadband or per channel, conditions where noise was only allowed during speech periods and conditions when noise was always present.

**Results:** We saw large difference in the effect of the different types of noise distortion for both CI subjects and normal hearing and most interestingly we saw that the results were very different for the two subject groups. The results will be discussed in detail and reviewed against other noise reduction and CI coding strategy data.

**Conclusions:** We conclude that the requirements for noise reduction algorithms of cochlear implants are significantly different than for normal hearing subjects. In contrast to what is known from hearing aid noise reduction, rejection of (near) noise only channels can have a very large impact on the performance of CI subjects, while rather severe distortions of the speech envelope modulations can be tolerated. These findings are important for the design of noise reduction strategies for cochlear implants.

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**Understanding the effect of noise on cochlear implants**

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**Objectives:** To characterize and understand the effect of noise on the performance of cochlear implant recipients and to understand what the similarities and differences are between the requirements for noise reduction for cochlear implants vs hearing normal hearing.

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**Wireless transmission delay time estimation method for active delay algorithm development in binaural hearing aids**

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Binaural hearing aids consist of two hearing apparatus, one for each ear. An intermediate device with an audio device to send audio data to two hearing aids simultaneously using wireless technology has already been developed. Unfortunately the intermediate device suffers from several inconveniences in terms of carrying and usage. Our research proposes a new concept of binaural hearing aids: a Bluetooth dongle substituting the intermediate device is attached to the master hearing aid. The master hearing aid receives stereo audio signals via the Bluetooth device and one channel signal is sent to the slave hearing aid by a 2.4 GHz RF GFSK transmission method to create the binaural hearing aid effect. However, the problem with this system is the processing necessary for the signal transmission and reception in the two hearing aids, which creates a time delay causing the precedence effect. Therefore, our primary objective is to estimate the delay time in wireless transmission in order to develop an active delay algorithm to synchronize the two hearing aids. In this paper, a method is proposed for delay time estimation.

Transmission and reception evaluation boards consist of ADC, DAC, microcontrollers and wireless communication chips. Two boards are each fixed to represent the left and right ear of a subject to mimic the real binaural hearing aid environment. Professionally designed software for real-time sound system measurement was used to estimate the delay time. The software is setup on a laptop and connected with an audio I/O device by USB to constitute the estimation system. The right-side output signal from the audio I/O device is sent through the wireless test boards, and the corresponding output signal from receiver is fed into the audio I/O device for measurement. The left-side output of the audio I/O device is directly connected to the input of the audio I/O device for reference. The delay time is measured by comparing the signal’s input into the system to its corresponding output signal in the software.

The estimated delay times are 7.25ms and 13.04ms that each estimated when latency of the wireless transmission was setup each in 7ms and 12.5ms. The delay time is assumed according to the combined total of the latency of wireless communication, signal processing and acoustic propagation time. The results of our experiment indicate that by using the proposed estimation system it is possible to estimate the delay time accurately in binaural hearing aids.

C29
Release from masking through spatial separation in distance
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It is widely accepted that speech intelligibility improves as a speech signal and interfering masker are separated spatially in azimuth. However, little attention has been given to the effect of spatial separation in distance. In the current study speech reception thresholds (SRTs) were investigated using two different measures, the Listening in Spatialized Noise – Sentences Test (LiSN-S) and the Coordinate Response Measure (CRM). Both corpuses are characterized by providing a large degree of informational masking. To compare the results with pure energetic masking, the SRT was additionally measured in a reference condition using speech-shaped noise (SSN). Two different setups were realized. In a first setup the target was presented at a distance of 0.5m from the center of the listener’s head and the interferer at a distance of 0.5 m, 1 m, 2 m or 10 m. In a second setup the interferer’s location was fixed and the target’s location was varied accordingly. Spatial synthesis was realized through convolution with binaural room impulse responses (BRIRs) recorded in an auditorium. In order to compensate for the distance dependent change in overall intensity, the total signal power was equalized and the long-term frequency spectra of the interferers were adjusted to the average target spectrum. In the case of the SSN interferer the SRT was unaffected by changes in distance. However, for the speech interferer the results revealed a substantial release from masking as the target and interferer were separated in distance. This effect was consistent for both the LiSN-S and CRM measure as well as for both target-masker setups. The strongest release from mask-
ing effect of about 10 dB was observed for the CRM corpus when the target talker was at 0.5 m and the distractor was moved from 0.5 m to 10 m. In this configuration the SRT was similar to the one observed for the SSN interferer. This study suggests that distance related cues play a significant role when listening in complex environments and that the ability to use such cues could relate directly to how normal hearing and hearing impaired listeners function in complex scenes.

**C30**

**Speech understanding in spatially separated noise with bilaterally linked compression**

**I.M. Wiggins and B.U. Seeber**

MRC Institute of Hearing Research, Nottingham, UK

Dynamic-range compression is used in hearing aids and cochlear implants to compensate for the reduced dynamic range of the impaired auditory system. Recently introduced devices allow compression to be coordinated at the two ears through a wireless link. This study investigated in normal-hearing listeners how linking compression across the ears might provide a benefit for speech intelligibility in the presence of a spatially separated noise.

Sounds were filtered with head-related transfer functions to simulate speech from directly in front and a steady speech-shaped noise from an azimuth of 60°. Fast-acting compression with a 3:1 ratio was applied in two independent frequency channels (100–2000 Hz and 2000–5000 Hz). The compression operated either independently at each ear (unlinked condition) or in such a way that identical gain was applied at both ears at all times (linked condition).

An analysis of the compressors’ behavior showed that unlinked compression disturbed interaural level differences (ILDs) and that linked compression achieved a more favorable long-term apparent speech-to-noise ratio at the listener’s better ear than unlinked compression. The perceptual relevance of these effects was assessed in a speech intelligibility experiment. Intelligibility was significantly better with linked than with unlinked compression, and was almost as good with linked compression as with no compression. The benefit of linked over unlinked compression was the same for binaural and for monaural better-ear listening, indicating that the benefit arose from changes to the signal at the better ear rather than from the preservation of ILDs. This is probably because, in this type of speech-in-steady-noise scenario, the binaural interaction benefit is conferred mostly through interaural time differences, which are not directly affected by compression.

The experimental results were modeled by using the $I_3$ measure [Kates and Arehart, JASA 177, 2224–2237 (2005)] to predict intelligibility for monaural better-ear listening. The same model was used to predict intelligibility for a hypothetical hearing-impaired listener with a symmetrical moderate hearing loss using bilateral in-the-canal hearing aids. The modeling results suggest that the potential benefit of linked compression would not be overcome by a counteracting reduction in audibility (because the linked system applies less overall gain at the better ear) for the particular condition considered, and that such a listener could potentially benefit from an improvement in intelligibility of up to about 10 percentage points.

**C31**

**A method for hearing aid fitting verification with phoneme audiometry**

**Alexandra Winkler¹, Inga Holube¹, Nicola Schmitt², Martina Wolf² and Michael Boretzki²**

¹ Jade University, Institute of Hearing Technology and Audiology, Oldenburg, Germany
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Hearing aid fittings can be verified by using various speech intelligibility tests. Unfortunately, most of those tests, especially when using sentences, are not sensitive for high frequency amplification which is needed for a typical sloping hearing loss configuration. Tests which are sensitive for high frequency information need to include specific phonemes while excluding redundancy and context effects. A phoneme test was developed in order to measure audibility, discrimination ability and recognition performance of high frequency speech information. A female speaker was recorded speaking logatoms of the form /a/-consonant-/a/. The phoneme /a/ at the beginning and at the end of each logatom was
partly replaced by the same recording using the so-called cross fading method to focus on the differences between the logatoms to the mid consonant. The consonants /sh/ and /s/ where modified to generate two versions with different mid frequencies (3 and 5 kHz for /sh/, and 6 and 9 kHz for /s/). The results showed that the phoneme audibility seemed to be based at least partly on low frequency information still detectable within the stimuli. Consequently, the phonemes /sh/ and /s/ as well as their transition to the preceding and following phoneme /a/ were modified to reduce low frequency information such as vowel-transients and aspirations. The results show an improved suitability of the phonemes for the verification of hearing aid fittings at high frequencies.

The importance of temporal neural cues for predicting speech intelligibility
Michael R. Wirtzfeld and Ian C. Bruce
McMaster University

Researchers have made several attempts to use perceptual cues to establish physiologically-based speech intelligibility predictors that would be beneficial for hearing aid evaluation and development. However, establishing neural correlates between speech perception and the cues of envelope (ENV) and temporal fine structure (TFS) has been difficult to realize. The spectrotemporal modulation index (STMI) metric\(^1\) has proved useful for predicting the effects of presentation level on intelligibility in normal-hearing and hearing-impaired listeners\(^2\) and the effects of different hearing aid compression schemes\(^3\). The STMI metric, however, cannot explain speech intelligibility for “auditory chimaeras”\(^4\) in which the speech information is primarily in the TFS\(^5\). This motivates the use of predictors that can incorporate TFS information, like the Neurogram SIMilarity (NSIM) metric\(^6\). In this study we investigate how well the NSIM metric can explain chimaera data.

To analyze the importance of ENV and TFS cues, five chimaera types are considered: speech TFS and white-Gaussian noise ENV; speech TFS and spectrally-matched noise ENV; speech TFS and flat ENV; speech ENV and spectrally-matched noise TFS; speech ENV and white-Gaussian noise TFS. Five normal hearing listeners subjectively graded a corpus of 1,750 sentences, 350 for each chimaera type. Using sentence pairs, a chimaera and its respective original NU 6 sentence, model auditory nerve responses\(^7\) are simulated across a set of cochlear characteristic frequencies. The sets of spike trains are re-binned and filtered to produce a pair of spectrotemporal characterizations called “Neurograms” for each sentence: the first has large time bins and thus only represents the ENV cues, while the second retains the TFS cues via the use of small time bins. Using the NSIM metric, these pairs of neurograms are compared with their respective clean sentence neurogram pairs to establish a bounded value capturing the degradation in the chimaera neurograms. In this poster we will describe progress in obtaining optimized predictions of the chimaera intelligibility data using the NSIM metric. In particular, we are interested in determining whether a linear or nonlinear combination of NSIM values obtained from the ENV and TFS neurograms is required. [Supported by NSERC Discovery Grant 261736.]

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On the factors affecting speech reception in a multi-talker listening scenario
Bill Woods, Sridhar Kalluri, Shareka Pentony and Nazanin Nooraei
Starkey Hearing Technologies

Hearing aid users often report dissatisfaction when using aids in noisy scenarios. It is unclear to what extent this complaint arises due to deficits beyond that expected from their loss of audibility. In this study auditive and cognitive influences on speech perception were investigated in listeners with normal hearing (NH) and listeners
with hearing loss (HL) who heard speech presented in a laboratory simulation of a multi-talker scenario. The speech was the nonsense-syllable response measure (NSRM), similar to the coordinate response measure (Bolia et al., 2000) but with the coordinate replaced by a nonsense syllable. This allowed computation of performance predictions using the speech intelligibility index (SII). All listening was done over headphones, and measured impulse responses were used to simulate a condition in which each sentence came from a different location. Listeners heard target NSRM sentences in quiet (Q), in quasi-diffuse background noise (BN), and in the noise plus other NSRM sentences (BNJ; each sentence from different location). Sentence onsets were separated by 800 ms in BNJ. All stimuli were filtered with the equivalent of the NAL prescription for one of the HL subjects. All listeners heard the same stimuli in the same order of presentation. Additional measurements comprised absolute tone-detection, forward and backward digit-span tests, and Trail-Making A and B tests (TMA and TMB). A “proficiency” factor was determined using the SII and percent-correct result in quiet, and used with the SII in predicting results in the other conditions.

Results for all NH subjects were well-predicted in BN and for 8 of 10 NH subjects in BNJ. Predictions were also accurate for the HL group in BN and were accurate for more than half of these 16 listeners in BNJ. The difference (“residual”) between predicted and measured BNJ results for all subjects was found to be uncorrelated with the digit span tests, and in a step-wise regression found to be significantly correlated with results of the trail-making tests: r=0.45 with TMA, and sr=0.57 with TMB. As a third step in the regression age was found not significantly correlated with residuals when the TMA and TMB effects were factored out.

These results are consistent with audibility and quiet-speech-processing factors explaining the difficulty experienced by a significant proportion of hearing-aid users in complex, multi-talker environments; other factors play a role for a smaller proportion of users. Further work is required to tie these laboratory results directly with real-world hearing-aid satisfaction.

**The effect of hearing aid technologies on listening in automobiles**

Yu-Hsiang Wu, Elizabeth Stangl, Ruth Bentler and Rachel Stanziola
The University of Iowa

**Background:** Communication while traveling in automobiles is often difficult for hearing aid users. This is because the automobile/road noise level is usually high, and listeners/drivers often do not have access to visual cues. Since the talker of interest is not usually located in front of the driver/listener, conventional directional processing that places the directivity beam toward the listener’s front might not be helpful, and could be detrimental to speech recognition. Recently, technologies have become available in commercial hearing aids that are designed to improve speech recognition in noisy automobiles. These technologies include (1) a directional microphone system that uses a backward-facing directivity pattern (Back-DIR processing) and (2) a technology that transmits audio signals from the ear with the better signal-to-noise ratio (SNR) to the ear with the poorer SNR (Side-Transmission processing). The purpose of the current study was to determine the effect of (1) conventional directional microphones and (2) newer signal processing schemes on listener’s speech recognition performance and preference for communication in an automobile.

**Methods:** Twenty-five adults with bilateral symmetrical sensorineural hearing loss aged 44 through 84 years participated in the study. The automobile/road noise and sentences of the Connected Speech Test (CST) were recorded through hearing aids in a standard van moving at a speed of 70 miles/hour on a paved highway. The hearing aids were programmed to omnidirectional microphone, conventional adaptive directional microphone, and the newer schemes. CST sentences were presented from the side and behind the hearing aids, which were placed on the ears of a manikin. The recorded stimuli were presented to listeners via earphones in a sound treated booth to assess speech recognition performance and preference with each programmed condition.

**Results:** Compared to omnidirectional microphones, conventional adaptive directional pro-
cessing had a detrimental effect on speech recognition when speech was presented from the side or behind the listener. Back-DIR and Side-Transmission processing improved speech recognition performance (relative to both omnidirectional and adaptive directional processing) when speech was from the back and side, respectively. The participants’ preferences for a given processing scheme were generally consistent with speech recognition results.

Conclusions: The finding that performance with adaptive directional processing was poorer than with omnidirectional microphones demonstrates the importance of selecting the correct microphone technology for different listening situations. The results also suggest the feasibility of using hearing aid technologies to provide a better listening experience for hearing aid users in automobiles.

Tracking semantic information derived from multiple talkers: The influence of spatial separation
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The effect of hearing loss on the processing of speech in complex acoustic scenes has typically been studied experimentally using tasks in which listeners repeat a sentence or respond to a set of keywords from one of multiple talkers. Such studies may underestimate the difficulty for listeners with hearing loss because they fail to address two aspects of understanding speech in realistic multi-talker scenarios -- 1) Listeners not only want to detect or identify simple targets, but they also want to monitor multiple simultaneous sound sources; 2) listeners typically have comprehension as their goal, not just reception of the phonetic components of the speech. Our research aims to address these issues by asking listeners to track semantic information derived from multiple talkers.

Our initial step, with normal-hearing listeners, involves characterizing the influence of spatial separation on source segregation and recruitment of attention. In a simulated realistic environment, the subject hears multiple talkers from different directions, each presenting different streams of natural speech (stories). A stream of questions is derived from the stories, along with a pair of potential answers, and they are presented on a visible screen for response with push buttons. Semantic processing of information is emphasized by writing the answers in a way that maintains the intrinsic meaning of a passage while replacing its keywords. With subject paid for correct answers, attention is controlled by telling him/her that the majority of the questions come from the “primary talker”. Results show a strong limitation of semantic processing in terms of number of talkers. We observe 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 fall into the focus of attention on the primary.

Other studies underway examine effects of age in normal and hearing impaired listeners, reflexive shifts of attention to non-primary talkers, and the time to switch attention when the primary talker is redefined.

Perceptual confusions in combined electric and acoustic stimulation
Yang-soo Yoon and Qian-Jie Fu
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In bimodal hearing, there are two possible underlying mechanisms: “integration” and “better-ear listening”. “Integration” refers to combining the acoustic information provided by the hearing aids (HAs) with the electric information provided by the cochlear implants (CIs), while “better-ear listening” refers to processing speech information exclusively provided by a better ear alone. It is widely believed that bimodal users who receive the benefit have the ability to integrate the information, while those who receive little benefit have the inability to integrate the information;
thus, “better-ear listening” is dominant. In this study, we aimed to elucidate whether this general belief is true in terms of perceptual confusions in consonant and vowel recognition.

Fourteen adult bimodal listeners were grouped according to the difference in their performance across ears. In the benefited group, the performance difference across ears is smaller than that averaged over all subjects; in the non-benefited group, the performance difference across ears is greater. Confusion matrices were measured for the consonant and vowel recognition in noise and in quiet with a HA alone, a CI alone, and a combined CI+HA when speech and noise were presented from the front.

Results showed that the benefited group received the greater bimodal benefit for a subset of consonants and vowels, while the non-benefited group did not receive the benefit for any of consonants and vowels presented. The further analyses showed that for the non-benefited group bimodal interference (bimodal performance is poorer than better ear alone) occurs. If “better-ear listening” is a primary factor in bimodal benefit, then this interference should not occur. The results also showed that for both groups the structure of phonetic cues between the bimodal and the CI alone conditions was completely different. Specifically for the benefited group, a HA helped resolve the most confused sounds, leading to the greater bimodal benefit, caused by a CI alone, while for the non-benefited group, a HA failed to resolve the most confused but did help resolve other minimally, but multiply confused sounds, leading to little or no benefit. If “better-ear listening” is a primary factor, the structure of the cues should be similar between the bimodal and the CI alone conditions.

The results suggest that integration alone is a primary underlying mechanism, while there is no evidence for “better-ear listening mechanism”. The results also suggest that integration process is maximally facilitated for a certain subset of consonants and vowels, not for all sounds. [Work supported by NIH grant 5R01-DC004993.]

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Acoustic analysis of consonant-vowel-consonant syllables (CVCs) from the California syllable test (CaST)

E. William Yund1, Tanya L. Arbogast1, Marc Ertlinger1 and David L. Woods1,2

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2 UC Davis

We developed the CaST to test the perception to 20 initial and 20 final consonants in speech-spectrum noise (SSN) with each consonant presented at signal-to-noise ratios (SNRs) where it would be identified at about 67% correct. The stimulus set includes 40 exemplars of each initial and final consonant for each of the three vowels and six talkers. Details of the test and normative results for young-normal-hearing (YNH) and older-normal-hearing listeners have been published [JASA 127: 1609-1623 (2010); JRRD in press (2012)]. We are beginning an acoustic analysis of these syllables to identify the acoustic features that distinguish initial consonants from each other. The first phase is to separate each initial consonant (including any formant transition) from the remainder of the CVC. Time-normalized spectrograms are computed for each exemplar by varying the time-spacing of the FFT spectral lines in proportion to the exemplar duration. Next we compute the average spectrogram for each consonant for each vowel and talker. In preliminary comparisons of the average spectrograms of /da/ versus /ba/ for one female talker, the two can be discriminated by the higher-frequency plosive burst components of the /da/ [Li, Menon & Allen, JASA 127: 2599-2610 (2010)] as well as the difference in the formant transitions. When voice-specific SSN is added to the /da/, at the three SNRs used for YNH listeners, the /da/ cues identified in the quiet spectrograms are clearly seen in the spectrogram for the highest SNR, become less clear for the intermediate SNR, and are not apparent for the lowest SNR. When the spectrogram is given a sloping, high-frequency hearing loss, by suppressing spectral components that would be below threshold, the high-frequency /da/ burst is absent, but some of the formant transition remains. When the spectrogram is given the same hearing loss plus hearing aid amplification, most of the burst and all of the formant transition returns. However, the
aided /da/ cues are more sensitive to added SSN and are obscured at a lower SNR than for the YNH listeners, above. The long-term goal is to apply this analysis to understand the effects of hearing loss and HAs on consonant perception and to predict consonant confusion patterns obtained with the CaST with normal-hearing and hearing-impaired listeners.

The effect of hearing aid shell and wind shield designs on wind noise levels
Justin A. Zakis and Daniel J. Hawkins
Wolfson Dynamic Hearing

Wind noise attracts some of the lowest hearing-aid satisfaction ratings, with only 58% of surveyed respondents reporting some degree of satisfaction in wind noise (Kochkin, 2010). However, despite the significance of the problem, there are few published studies on wind noise with hearing aids. Wind noise can be reduced at the microphone input through hearing-aid shell and wind shield design, and at the hearing aid output with signal processing approaches. Reducing wind noise at the microphone input is advantageous since it can: a) increase the speech-to-wind-noise ratio; b) avoid clipping distortion in the microphone and hearing aid circuits; c) reduce the need for aggressive wind noise reduction algorithms and the chance of negative interaction with other algorithms; and d) reduce the amount of fitting optimisation required in the clinic. Most previous studies have compared wind noise spectra at one microphone across different hearing aids. We are aware of one study that investigated the benefits of one wind shielding approach in hearing aids (Grenner et al., 2000), although this was with an improvised foam wind shield at one wind speed and angle (7 m/s, 0° azimuth). The current study investigates the efficacy of different commercial hearing-aid shell and wind shield designs within and across hearing aids at different wind speeds and angles. Three behind-the-ear hearing aids with dual microphones were used, and three wind-shielding approaches were evaluated in isolation and/or in combination. The hearing aids were mounted on the right ear of a Knowles Electronics Manikin for Acoustic Research, and stereo recordings of both microphone signals were made at two realistic wind speeds that avoided microphone saturation (3 and 6 m/s) and 36 wind angles (0-350° azimuth) to reveal average trends. The recordings were analysed to evaluate whether the different approaches provided a clinically significant reduction in wind noise levels, and the implications for hearing-aid designers and clinical practice are discussed.

References:

Applications of an annoyance perception model to noise reduction for hearing aids
Tao Zhang, Srikanth Vishnubhotla, Jinjun Xiao and Martin McKinney
Starkey Hearing Technologies

It is well known that hearing aid wearers have a low tolerance for high level ambient noise and can easily become fatigued as a result of prolonged exposure. Noise reduction algorithms can be used to improve listening comfort and reduce fatigue in such environments. Traditional noise reduction algorithms are typically designed to reduce the ambient noise by minimizing the noise power. However, this may not be the most effective way to improve listening effort in noisy environments because what really matters is the noise perception instead of noise power. A more effective way to reduce the noise is to reduce the associated annoyance. Annoyance perception by hearing impaired listeners has been studied systematically and a computationally efficient model has been proposed to account for the annoyance perception as a function of hearing loss (Vishnubhotla et al., 2012). In this study, we incorporate the annoyance perception model for hearing impaired listeners into a traditional noise reduction algorithm. The model takes an audiogram and input signal as input and generates an annoyance score as output. For a given input signal, the
proposed algorithm minimizes the annoyance score instead of the noise power of the output signal. To evaluate the benefits of the proposed algorithm, eight different real-world noise recordings are used. The recordings are processed by the proposed algorithm and a traditional noise reduction algorithm that minimizes the overall noise power. Ten listeners with different degrees of hearing loss are asked to listen to the processed recordings by both algorithms and rate the corresponding annoyance and sound quality. The subjective and objective evaluation results will be presented and discussed.
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