IHCON 2010

International Hearing Aid

Research Conference

• -- • -- •

August 11 – 15, 2010

Granlibakken Conference Center
Lake Tahoe, California

IHCON 2010

Sponsors

● -- ● -- ●

**National Institute on Deafness
and Other Communication Disorders**

The International Hearing Aid Industry

The House Ear Institute

IHCON 2010

Planning Committee

● -- ● -- ●

Technical Chair

Stefan Launer

Technical Co-Chairs

Michael Akeroyd • Benjamin Hornsby

Organizational Co-Chair

Sigfrid Soli

Steering Committee

Sigfrid D. Soli, Organizational Co-Chair
House Ear Institute, USA

• -- • -- •

Stefan Launer, Technical Chair (2010)
Phonak AG, Switzerland

• -- • -- •

Ben Hornsby, Technical Co-Chair (2010)
Vanderbilt University, USA

• -- • -- •

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

• -- • -- •

Kathryn Arehart
University of Colorado, USA

• -- • -- •

Ian Bruce
McMaster University, Canada

• -- • -- •

Brent Edwards
Starkey Hearing Research Center, USA

• -- • -- •

Larry Humes
Indiana University, USA

Peggy Nelson
University of Minnesota, USA

• -- • -- •

Graham Naylor
Eriksholm Research Center, Denmark

• -- • -- •

Todd Ricketts
Vanderbilt University, USA

• -- • -- •

Gabrielle Saunders
Nat'l Center for Rehabilitative Auditory Rsch, USA

• -- • -- •

Brian Moore, Technical Chair (2008)
Cambridge University, UK

• -- • -- •

Judy Dubno, Technical Co-Chair (2008)
Medical University of South Carolina, USA

• -- • -- •

Jan Wouters, Technical Co-Chair (2008)
Leuven University, Belgium

Student Scholarship Recipients

● -- ● -- ●

Name	Institution
Penny Anderson Gosselin	Institut universitaire de gériatrie de Montréal, Canada
Iris Arweiler	Technical University of Denmark, Denmark
Jonathan Boley	Purdue University, USA
Marc Brennan	University of Washington, USA
Foong Yen Chong	University of British Columbia, Canada
Bram Cornelis	Katholieke Universiteit Leuven, Belgium
Jamie Desjardins	Syracuse University, USA
Rachel Ellis	University of Manchester, United Kingdom
Andrew Hines	Trinity College Dublin, Ireland
Julie Ladiges	University of Western Ontario, Canada
Ariane Laplante-Lévesque	University of Queensland, Australia
Anjali Menon	University of Illinois at Urbana Champaign
Martin Müller	University Hospital and Federal Instutitue of Technology, Switzerland
Manasa Panda	Essex University, United Kingdom
Gurjit Singh	University of Toronto, Canada
Thomas Weddell	University of Sussex, United Kingdom
Ke Xu	West China Hospital of Sichuan University, China

Daily Schedule

● -- ● -- ●

Wednesday, August 11, 2010

5:00 PM	Welcome Social
6:00 PM	Dinner
7:30 PM	Welcome Remarks
7:45 PM	Keynote Address
8:45 PM	Discussion
9:00 PM	Evening Social

Thursday, Friday & Saturday, August 12-14, 2010

7:00 AM	Breakfast
8:00 AM	Morning Session A
9:45 AM	Poster Session
11:10 AM	Morning Session B
12:20 PM	Lunch
5:15 PM	Evening Session
7:00 PM	Dinner
8:20 PM	Social

Sunday, August 15, 2010

7:00 AM	Breakfast and Checkout
8:15 AM	Morning Session
9:35 AM	Break
10:00 AM	Morning Session Continues
11:00 AM	Conclusion (buses leave for airport with boxed lunches for passengers)

Program Summary

● -- ● -- ●

Wednesday, August 11

Welcome and Keynote Address

7:30 PM – 8:45 PM

Welcome Remarks: Sig Soli, Stefan Launer

Keynote Address

Harvey Dillon	Mild hearing loss is serious business

Thursday, August 14

Session One

8:00 AM – 9:45 AM

Mild Hearing Losses I: Physiology, Models

Moderator: Brian Moore

Steve Neely	Measurements and models of normal and impaired cochlear responses
Michael Heinz	Temporal fine-structure and envelope coding in auditory-nerve fibers following noise-induced hearing loss
Volker Hohmann	Application of auditory models to assessing the sound quality of hearing aid signal processing algorithms

Poster Session A 9:45 AM – 11:10 AM

Session Two

11:10 AM – 12:20 PM

Mild Hearing Losses II: Psychoacoustics

Moderator: Andrew Oxenham

Anne Marie Tharpe	Consequences of minimal and mild hearing loss in children
Brian Moore	Perceptual consequences of mild hearing loss and benefits of extended bandwidth in hearing aids

Session Three

5:15 PM – 6:50 PM

Mild Hearing Losses III: Hearing Instrument Technology

Moderator: Nikolai Bisgaard

Todd Ricketts	Adults with mild and moderate hearing loss: Are they really hearing aid candidates?
Brent Edwards	Aided psychoacoustics and binaural perception with hearing aids
Gabrielle Saunders	Evaluation of auditory training with veterans

Friday, August 15

Session Four

8:00 AM – 9:45 AM

Speech Intelligibility in Complex Environments

Moderator: Todd Ricketts

Michael Vorländer	Sound fields in complex listening environments
Birger Kollmeier	Speech intelligibility prediction in complex listening environments
Graham Naylor	Limitations of Speech Reception Threshold as an outcome measure in modern hearing-aid research

Poster Session B 9:45 AM – 11:10 AM

Session Five

11:10 AM – 12:20 PM

Individual Differences, Psychoacoustics

Moderator: Michael Akeroyd

Charles Watson	Individual differences in auditory abilities
Andrew Oxenham	Exploring the coding of temporal fine structure in normal and impaired hearing

Session Six

5:15 PM – 6:55 PM

Algorithms & Algorithm Performance

Moderator: Pamela Souza

Bram Cornelis	Speech intelligibility improvements in hearing aids using adaptive bilateral and binaural multichannel Wiener filtering based noise reduction
Jing Chen	Enhancement of spectral changes over time – method and initial assessment
Karolina Smeds	Predictive measures of the intelligibility of speech processed by noise reduction algorithms
Jay Desloge	Impact of wearer's physical characteristics on first-order hearing aid directionality
Tobias Neher	Pinna cue-preserving hearing-aid fittings: Field-testing of benefit

Saturday, August 16

Session Seven

8:00 AM – 9:45 AM

Outer and Middle Ear Acoustics

Moderator: Harvey Dillon

Sunil Puria	High-frequency biomechanics, diagnostics, and hearing aids for the middle ear
Morten Nordahn	The potential of acoustic modeling
Daniel Warren	Pressure lies

Poster Session C 9:45 AM – 11:10 AM

Session Eight

11:10 AM – 12:20 PM

Tinnitus Management

Moderator: Tim Trine

Rich Tyler	Present and future directions for tinnitus treatment
Ole Dyrlund	Advanced hearing instrument technology for tinnitus management

Session Nine

5:15 PM – 6:55 PM

Plasticity and Implantable Hearing Systems

Moderator: Kevin Munro

Anu Sharma	Auditory neuroplasticity in children with hearing impairment
Tom Francart	Signal processing for combined hearing aid and cochlear implant stimulation
Norbert Dillier	Speech intelligibility in various noise conditions with the Nucleus 5 sound processor

Sunday, August 15

Session Ten

8:15 AM – 9:35 AM

Cognitive Aspects, Rehabilitation

Moderator: Susan Scollie

Michael Fisher	Speech Referenced Limiting (SRL): a new approach to limiting the sound level from a hearing aid
Christian Füllgrabe	Relationship between cognitive and speech-perception performance for elderly listeners with normal hearing
Piers Dawes	Older adults, hearing aids and listening effort
Thomas Lunner	Factors affecting help-seeking, hearing-aid uptake, use and satisfaction – what do we know?

Break 9:35 AM – 10:00 AM

Session Eleven

10:0 AM – 11:00 AM

The Legacy of Stuart Gatehouse

Moderator: Graham Naylor

Jerker Rönnberg	Cognitive hearing science: The legacy of Stuart Gatehouse

CLOSE OF CONFERENCE

ORAL PROGRAM

Wednesday, August 11

Keynote Address

7:45 PM **Mild hearing loss is serious business**

Harvey Dillon
National Acoustics Laboratory

Mild hearing loss is extremely common, but only a very small proportion of people with mild hearing loss obtain and wear hearing aids. Even fewer continue using them. This talk will address several questions relating to the potential of hearing aids to help people with mild hearing loss. The characteristics of mild loss include reduced frequency selectivity, reduced temporal resolution, reduced spatial hearing ability, and increased speech reception threshold in noise. The latter can be used to assess hearing loss over the telephone. Although mild loss decreases the audibility and intelligibility of soft speech, difficulty listening in noise is the problem most commonly troubling people with mild loss.

Hearing aids, by contrast, provide the greatest benefit for soft speech in very low noise levels. Hearing aids with directional microphones do provide benefit in noisy places, but the benefit is typically only a few decibels improvement in speech reception threshold, and room acoustics and open fittings often cause the benefit to be less that that, or even zero. The potential wearer is therefore offered devices that can provide a large benefit where the wearer has a small need and a small benefit where he or she has a large need. The decision to acquire hearing aids consequently hinges on a finely balanced cost-benefit analysis by the mildly hearing-impaired person. Factors likely to affect the decision include the person's beliefs about likely benefits, practical disadvantages (whistling, reliability, discomfort), hearing status, expected impact on self-image and the perception of others, level of disability and handicap experienced, and financial cost. Beliefs about these issues, often strongly affected by the reports of other people who have tried hearing aids, are probably more important than objective facts. Once people come to a decision that they need help and that hearing aids will help them, success with the resulting fitting is almost guaranteed. Conversely, those that are begrudgingly coerced into trying hearing aids rarely find them satisfactory.

Technology hugely affects the objective benefits of hearing aids, and their disadvantages. Technology therefore also shapes the beliefs in society about hearing aids as people use them and tell others of their experiences. Future advances to lower the cost of hearing aid provision include self-fitting and trainable devices. Future advances that will make hearing aids more effective in noise include binaural-processing, super-directional hearing aids and active occlusion reduction. In combination, these features should allow people with mild hearing loss to hear better than people with normal hearing in noisy places. If so, society will come to regard hearing aids as visible indicators of super-human hearing, rather than as badges of disability.

Thursday, August 12

SESSION ONE

Mild Hearing Losses I: Physiology, Models

Moderator: Brian Moore

8:00 AM **Measurements and models of normal and impaired cochlear responses**

Stephen T. Neely and Michael P. Gorga
Boys Town National Research Hospital

The cochlea, the sensory organ for hearing, prepares auditory signals for neural conveyance to the brain by mapping the signal's frequency content and by amplifying low-level signals more than high-level signals. The cochlear amplifier, powered by outer hair cells (OHCs), provides this level-dependent gain only in its normal state. Evidence of this compressive gain is observed in cochlear responses. Because OHC malfunction is the most common cause of hearing loss, an "ideal" hearing aid would need to re-supply gain with level-dependent characteristics that resemble normal cochlear gain. Estimates of normative gain for tones may be derived from measurements of categorical-loudness scaling. However, because the cochlea distributes frequency content, normal gain at each frequency depends on surrounding frequencies. Measurements of two-tone suppression of otoacoustic emissions may provide the information needed to design hearing aids with normative gain that has distributed frequency dependence. While amplification of low-levels sounds create its own issues in patients with hearing loss, it remains the case that optimizing restoration of function will likely include the provision of compressive gain that that takes into account interactions across frequency.

8:40 AM **Temporal fine-structure and envelope coding in auditory-nerve fibers following noise-induced hearing loss**

Michael G. Heinz and Sushrut Kale
Purdue University

Recent perceptual studies have suggested that listeners with sensorineural hearing loss (SNHL) have a reduced ability to use temporal fine-structure cues for speech and pitch perception, particularly in degraded listening conditions. These results have fueled an active debate about the role of temporal coding in normal and impaired hearing, and have important implications for improving the ability of hearing aids (and cochlear implants) to restore speech perception in noise. However, these implications depend critically on the physiological bases for these results.

Spike trains were recorded from auditory-nerve (AN) fibers in chinchillas with either normal hearing or a noise-induced hearing loss that produced elevated thresholds and varying degrees of broadened tuning consistent with mixed outer- and inner-hair-cell damage. Within- and across-fiber temporal coding of pure tones, broadband noise, and speech were evaluated using shuffled correlograms. A spectro-temporal manipulation procedure predicted spatiotemporal patterns for characteristic frequencies (CFs) spanning an octave range from the responses of an individual fiber to a stimulus presented with sampling rates spanning an octave range. Cross-correlogram analyses quantified across-

CF fine-structure coding in terms of both a neural cross-correlation coefficient and a characteristic delay that estimates traveling-wave delays.

Envelope coding was enhanced in noise-exposed AN fibers, most strongly in fibers with high thresholds and very steep rate-level functions that were likely associated with severe outer- and inner-hair-cell damage. CF-tone responses suggest altered temporal dynamics in noise-exposed fibers, which may also contribute to enhanced envelope coding. Degradation in fine-structure coding strength was not observed; however, the transition region from fine-structure to envelope coding occurred at lower CFs following SNHL. The most significant degradations were observed in terms of spatiotemporal coding. The range of CF separations over which significant correlated activity existed was wider and estimated traveling-wave delays were shorter for noise-exposed fibers.

Overall, these data suggest that SNHL does not degrade the fundamental ability of AN fibers to phase-lock to either temporal fine-structure or envelope; however, several other effects of SNHL may contribute to perceptual deficits in temporal processing of complex stimuli. For example, SNHL had a greater effect on across-fiber coding of fine-structure than on within-fiber coding. This result has important implications for hearing-aid design according to spatiotemporal theories of speech and pitch coding. Furthermore, it appears to be important to consider the relative effects of SNHL on envelope and fine-structure coding, as well as the effects of both outer- and inner-hair-cell damage. [Supported by NOHR and NIH-NIDCD].

9:15 AM Application of auditory models to assessing the sound quality of hearing aid signal processing algorithms

Volker Hohmann
Medizinische Physik, Universität Oldenburg, Germany

Advances in systems technology allow for increasingly complex processing algorithms in hearing systems addressing increasingly complex acoustic conditions. These developments have the potential of improving the rehabilitation of hearing impairment, but establishing reliable measures of benefit is quite difficult for these complex algorithms and conditions. Being the "gold standard" for algorithm evaluation, subjective testing of hearing-impaired subjects has some limitations in this context. It is very time-consuming and the long acclimatization time needed when listening with new devices in complex acoustic conditions cannot easily be accounted for. Therefore, objective methods for estimating the benefit of an algorithm in a certain acoustic condition are desirable. They allow for identifying promising candidate algorithms and the acoustic conditions in which the algorithms might be applicable and thus identify critical acoustic conditions to be tested subjectively. In addition and in combination with technical measures like segmental signal-to-noise ratio and distortion measures, perceptual measures based on auditory models might be useful for developing meaningful objective measures. This contribution presents recent applications of auditory models to objective algorithm evaluation. Because of the growing importance of binaural and multi-microphone processing in hearing instruments, binaural/multichannel models for speech intelligibility and quality will be emphasized[1]. [Funded by DFG, BMBF and DAAD.]

Emiya, V., Vincent, E., Harlander, N., Hohmann, V. (2010). Multi-criteria subjective and objective evaluation of audio source separation. AES 38th International Conference on Sound Quality Evaluation. Piteå, Sweden: 9.

Thursday, August 12

SESSION TWO

Mild Hearing Losses II: Psychoacoustics

Moderator: Andrew Oxenham

11:10 AM **Consequences of minimal and mild hearing loss in children**

Anne Marie Tharpe
Vanderbilt University

Twenty years ago, audiologists, physicians, and other professionals were largely unconcerned about the impact of unilateral and minimal bilateral hearing loss on children. Audiologists would advise preferential seating for these children when they started school and assured parents that that would take care of any potential problems. However, as more and more parents expressed concern about their child's academic progress, professionals began to pay closer attention to this "minimal" problem. Today we have a significant amount of information about the psychoeducational and behavioral outcomes of children with minimal degrees of hearing loss. For example, 30-55% of children with unilateral and mild bilateral hearing loss have academic difficulties, with many requiring grade retention.

Despite widespread knowledge of this academic risk, children with minimal and mild losses are less likely to be fitted with hearing aids than those with greater degrees of loss – a situation that is even more pronounced in children with unilateral hearing loss. Furthermore, even when hearing aids are recommended, children with these losses are less likely to wear them consistently than children with greater degrees of loss.

This presentation will review past and current findings on the consequences of minimal degrees of hearing loss on children.

11:50 AM **Perceptual consequences of mild hearing loss and benefits of extended bandwidth in hearing aids**

Brian C.J. Moore, Christian Füllgrabe, Brian R. Glasberg, Michael A. Stone and Thomas Baer
Department of Experimental Psychology, University of Cambridge, UK

People with mild hearing loss over the frequency range that is most important for speech intelligibility (roughly 0.5 to 4 kHz) rarely use hearing aids, although they do seem to have difficulties in understanding speech in situations where background noise and/or reverberation are present. In the first part of this presentation, I will review evidence suggesting that these difficulties stem partly from a reduced ability to process temporal envelope and/or temporal fine structure information, as revealed in psychoacoustic tests using narrowband low-level stimuli, such that restricted regions in the cochlea are assessed. The perceptual deficits are not compensated by hearing aids, which may partly account for the limited use of hearing aids by people with mild hearing loss.

A second possible reason for the limited use of hearing aids is the limited bandwidth of most hearing aids. People with mild hearing loss may be candidates for extended-bandwidth hearing aids. We have developed a method for fitting such hearing aids called CAMEQ2-HF (Moore *et al.*, 2010). Several laboratory and field studies using this fitting method for subjects with mild hearing loss have shown: (1) An increase in bandwidth from 5 to 7.5 kHz was clearly audible for most subjects and an increase from 7.5 to 10 kHz was usually audible; (2) An increase in bandwidth from 5 to 7.5 kHz improved the ability to detect word-final /s/ and /z/; (3) Under conditions simulating listening via multi-channel (fast or slow) compression hearing aids, there was a small but significant benefit for speech intelligibility of increasing bandwidth from 5 to 7.5 kHz, when the target came from one side of the head and the interferer from the other. There was no significant benefit of increasing bandwidth from 7.5 to 10 kHz.

Moore, B. C. J., Glasberg, B. R., and Stone, M. A. (2010). "Development of a new method for deriving initial fittings for hearing aids with multi-channel compression: CAM-EQ2-HF," Int. J. Audiol. 49, 216-227.

Thursday, August 12

SESSION THREE

Mild Hearing Losses III: Hearing Instrument Technology

Moderator: Nikolai Bisgaard

5:15 PM **Adults with mild and moderate hearing loss: Are they really hearing aid candidates?**

Todd Ricketts
Vanderbilt University

A number of studies have suggested that children can exhibit a number of problems associated with mild and minimal hearing loss. Recent work suggests that adult listeners with mild-to-moderate hearing loss may also exhibit deficits including poorer word recall accuracy and increased cognitive effort (e.g. Tun et al., 2009). It might therefore be concluded that adults with hearing loss have the potential to derive benefits from amplification, even when the degree of hearing loss falls in the mild-to-moderate range. Despite this conclusion, data have shown that only a very small percentage of listeners with hearing thresholds that do not exceed approximately 60 dB HL pursue amplification. If it is assumed that some benefit from amplification is possible in these listeners, it might be concluded that non-pursuit of amplification may result from one of several negatives associated with hearing aids. According to survey data, oft cited reasons for non-pursuit or rejection of amplification include: cost, fit and comfort issues, negative side effects, poor sound quality (both own voice (due to occlusion and external sounds possibly due to limited bandwidth, poor cosmetics, and feedback. Recently a number of technologies have been combined in an attempt to offset many of these complaints. Specifically, open fitted micro-BTE instruments with well performing feedback suppression processing can lead to a cosmetically appealing and comfortable fit. Further, these technologies can be combined with extended high frequencies in an attempt to improve sound quality for external sounds. In this session, recent work specific to open fitting hearing instruments will be reviewed. Specifically, benefits, limitations, optimization of fitting formula, and the effects of extended bandwidth on sound quality in open instruments will be disucssed. Speculation regarding current issues related to open fittings including interaction with other features and processing and current and future challenges with using these technologies to remediate mild to moderate hearing loss in adult listeners will also be offered.

5:55 PM **Aided psychoacoustics and binaural perception with hearing aids**

Brent Edwards
Starkey Laboratories

Psychoacoustic measures of hearing impaired subjects are used to interpret the impact of hearing loss on higher level auditory function: tuning curve measures in hearing impaired subjects, for example, have been used to suggest that poor speech understanding in noise might be due to poor frequency resolution. In a similar manner, aided psychoacoustics can assist in understanding higher level auditory function with hearing aids: any impact of hearing aid algorithms on psychoacoustic measures might explain an improvement or deficit to higher-level aided auditory function.

Recently, the potential for wireless ear-to-ear hearing aid technology to affect binaural auditory function has been suggested. Understanding the impact of hearing loss on binaural ability and also the impact of non-communicating hearing aids on binaural function is necessary in order to determine the potential for benefit to hearing aid wearers from wirelessly-communicating binaural hearing aids. One method of investigating this potential is through aided binaural psychoacoustic measures. This talk will review data from our laboratory and others that provide insight to these issues and also detail the psychoacoustic considerations relevant to these issues.

6:30 PM **Evaluation of auditory training with veterans**

Gabrielle Saunders[1], Theresa Chisolm[2], Rachel McArdle[3], Sherri Smith[4] and Richard Wilson[4]

1 *National Center for Rehabilitative Auditory Research (NCRAR), Portland VA Medical Center*
2 *University of South Florida*
3 *Bay Pines VA Healthcare System*
4 *James H. Quillen VA Medical Center*

Hearing aid outcome varies widely. Recent studies have shown improved outcomes among individuals who have supplemented hearing aid use with home-based computerized auditory training. It is not clear, however, these findings can be generalized to all populations or whether specialized training programs are superior to auditory training through directed listening to, for example, books-on-tape. To examine this a three site, randomized controlled parallel group clinical trial is being conducted comparing four forms of intervention: (1) Auditory Training for twenty sessions (AT20) using the Listening and Communication Enhancement™ (LACE) program, (2) Auditory Training for ten sessions (AT10) using the DVD version of the LACE program, (3) Directed Listening (DL) for twenty sessions in which participants listen to, and answer questions about, books played from a computer and (4) standard-of-care hearing-aid intervention (CTL). Outcomes are assessed immediately following training and six months later. Participants are adults with mild to severe sensorineural hearing loss who have used their current bilateral hearing aid settings for at least 4-weeks. Primary outcome measures are the Words In Noise (WIN) test, Hearing Handicap Inventory for the Elderly (HHIE) and the Abbreviated Profile of Hearing Aid Benefit (APHAB), with secondary outcome measures of the Revised Speech Perception in Noise (R-SPIN), working memory (digit span task), auditory processing speed (accelerated NU6 materials) and selective attention (NU-20 words in the presence of a single talker masker). Analyses of variance of data from 134 participants show no intervention group differences on any measures other than the APHAB Ease of Communication (EC) scale, and the R-SPIN. AT20 participants reported fewer communication difficulties (APHAB EC scale) than subjects in the other intervention groups, while AT20 and DL subjects did not show improved ability to understand words-in-noise presented in isolation or in *low* context sentences. However, AT20 and DL resulted in improved word recognition in noise for *high* context sentences over the CTL condition. Further, auditory speed of processing improved for all participants in the LACE training group, while DL resulted in equivalent improvements for experienced hearing aid users only. Auditory memory improved equally with LACE and DL for experienced HA-users only. These results will be discussed with regard to effectiveness of auditory training and in relation to other published data in the area.

FRIDAY, AUGUST 13

SESSION FOUR

Speech Intelligibility in Complex Environments

Moderator: Todd Ricketts

8:00 AM **Sound fields in complex listening environments**

Michael Vorländer
RWTH Aachen University

The conditions of sound fields used in research, testing and fitting of hearing aids are usually simplified or reduced to fundamental physical fields, such as the free or the diffuse sound field. The concepts of such ideal conditions are easily introduced in theoretical and experimental investigations and in models for directional microphone, for example. This may be interpreted as effect of direct-to-reverberant sound field when one specific sound source such as in the communication situation with one talker is concerned. Noise sources may affect the communication via several complex sound fields as well.

When it comes to real-world application of hearing aids, however, the field conditions are more complex with regard to specific stationary and transient properties in room transfer functions and the corresponding impulse responses and binaural parameters.

In this presentation it is discussed how sound fields can be categorized in outdoor rural and urban and indoor environments. Furthermore, sound fields in closed spaces of various sizes and shapes and in situations of transport in vehicles, trains, aircrafts are compared with regard to the binaural signals at the open ear canal and at microphones of BTE and ITE hearing aids. The room acoustic field is analyzed in its modal decomposition, in its temporal reflection pattern and in the consequence for sound signals in general and for speech in particular. Speech intelligibility is affect by spatially distributed noise sources and by reverberation. It is, thus, to be discussed which information about the spatial sound field is required for best-possible segregation of signal and noise.

8:40 AM **Speech intelligibility prediction in complex listening environments**

Birger Kollmeier, Jan Rennies*, Rainer Beutelmann, Tim Jürgens, Bernd Meyer and Thomas Brand*
*Medizinische Physik, Universität Oldenburg and *Fraunhofer IDMT Hearing, Speech and Audiotechnology, Oldenburg, Germany*

Appropriate models for speech intelligibility in acoustically „difficult“ situations (i.e., background noise from several interfering sound sources, reverberation) are an important tool not only for research, but also to characterize the speech reception deficit of the individual listener as a function of his/her hearing loss in a certain environment. In addition, the models may help to assess and to optimize the performance of a hearing instrument for the individual in everyday life which is characterized by complex listening environments.

This contribution presents an overview on existing and new models that have mostly been implemented in the “Speech Intelligibility Prediction Toolbox” in order to allow for a comparison of model performance and to assess the expected speech intelligibility for a predefined situation. The focus of the presentation is laid on two aspects of speech intelligibility prediction that are currently under development:

a) Binaural interaction: A proper consideration and prediction of the binaural gain in speech intelligibility is a key property for correctly assessing the individual performance in rooms (Beutelmann et al., 2010). However, the relative contribution of early reflections and reverberation is not easy to model in a consistent way.

b) Speech recognition back end: Most models assume a degradable “peripheral processing unit” or “speech recognition front end” that accounts for the deterioration in speech intelligibility caused by hearing loss or imperfect (binaural) noise cancelling. On the other hand, a potentially perfect speech recognition back end is used to model the near-to-perfect “world model” by an optimal detector or a perfect knowledge of the SNR in frequency bands (such as, e.g., the classical prediction methods AI, STI, SII and related measures). If a more realistic back end is adopted from automatic speech recognition (such as, e.g., HMM or DTW), a considerable human-machine gap is typically observed that limits the applicability of this modelling approach (see Meyer, 2009).

In summary, a comparatively exact prediction of speech intelligibility can be done in “easy” acoustical situations for listeners with a mild-to-moderate hearing loss. The current models still show room for improvement as soon as more complex listening situations, hearing loss types and more detailed properties of the speech reception process are considered.

References:

Beutelmann, R., Brand, T., and Kollmeier, B. (2010) “Revision, extension, and evaluation of a binaural speech intelligibility model (BSIM),” J. Acoust. Soc. Am. (in press)

Meyer, Bernd: Human and automatic speech recognition in the presence of speech-intrinsic variations, University of Oldenburg, 2009.

9:15 AM **Limitations of Speech Reception Threshold as an outcome measure in modern hearing-aid research**

Graham Naylor
Eriksholm Research Centre, Denmark

SRT-like measurements for quantifying the intelligibility of speech in noise (with results in dB SNR) were originally developed as efficient diagnostic tools for evaluating speech-reception disability at the start of a clinical treatment process. It is nowadays increasingly common for research into hearing-aid (HA) systems to utilise such measurements when quantifying outcomes: however the validity and relevance of such approaches is open to question, especially with the increasing interest in highly non-linear signal processing schemes in HAs.

This presentation considers the influence of various factors on the suitability of SRT as an outcome measure in modern HA research. The fundamental problem is that SRT procedures do not adequately constrain the SNR at which testing takes place. The behaviour of non-linear systems (even simple ones like dynamic range compression) is highly dependent on the SNR at the input, and thus the results of comparisons between alternative non-linear (i.e. almost any) HA systems may be critically affected by

- aspects of the SRT procedure itself (corpus, scoring, noise type, acoustics etc.) which affect the SNR at which criterion %-correct is achieved,
- the inherent SNR needs (or 'diagnostic SRT') of the listener, which depend on audiometric loss, cognitive abilities, etc.

The issue of ecological validity is also considered; the unconstrained SRT approach implies that 'x dB SRT change' is equally significant, whatever the baseline SRT. Testing should take place in an SNR range relevant for the user (i.e. realistic), and for the HA feature under test.

SRT is an attractive outcome measure because it can be made relatively fast and repeatable, and 'always gives a result' with convenient statistical properties (and no floor or ceiling effects). However the first priority for outcome measures in HA research should be validity and relevance, not convenience.
Based on the foregoing concerns, suggestions for the further development of speech-in-noise outcome measures for HA research are discussed. These include:

- robust documentation of real-life SNRs
- possible constrained SRT procedures (e.g. manipulating task content to control the SNR at which baseline testing occurs)
- procedures based on fixed SNRs, and how to deal with floor/ceiling effects.

FRIDAY, AUGUST 13

SESSION FIVE

Individual Differences, Psychoacoustics

Moderator: Michael Akeroyd

11:10 AM Individual differences in auditory abilities

Charles Watson and Gary Kidd
Indiana University

In early studies of sensory abilities (~1860-1960), the range of individual differences (ID) was the focus of many investigations, while more recently IDs have been commonly treated as though they reflect measurement error rather than any stable characteristics of individual subjects. This view has, of course, been less common among clinical investigators, who have frequently hypothesized deviations from normal sensitivity or acuity as explanations of such conditions as delayed language development or reading disorders. The studies we have been conducting were originally motivated, not by hypothesized associations between clinical conditions and auditory processing abilities, but rather by the observation that IDs in discrimination and recognition are vastly larger for complex stimuli than for the traditional pure tones, noise bursts or clicks of psychoacoustics. Repeatedly confronted with unexpectedly large IDs, we set out to answer a few fundamental questions. The first was whether IDs on auditory discrimination tasks really are reliable properties of so-called normal hearing listeners? The second was, on how many different dimensions do listeners vary or, conversely, how many discrete auditory abilities do we have? We have addressed these questions by administering batteries of auditory tasks to large numbers of listeners. The reliability of performance is sufficient to say with confidence that IDs in auditory abilities are stable properties. The pattern of correlations among the tests in the batteries has suggested a fairly small number of discrete auditory abilities, possibly as few as four. A well-replicated finding in these studies has been that measured spectral and temporal acuity have only very weak associations with the ability to process familiar sounds, such as speech or the sounds of the environment. This familiar-sound- recognition ability is significantly correlated with visual speech recognition (from facial dynamics) but only very weakly associated with general intelligence. More recent studies have focused on the ability to learn to perform difficult auditory tasks. Using the test battery approach, we attempted to identify those individuals in a sample of 1000 young adults who were unusually capable of learning to discriminate and identify novel complex sounds. Neither auditory nor cognitive measures predicted IDs in performance on these tasks. However, despite very large changes over the course of several training sessions, very early performance was a strong predictor of that achieved at the end of training. As would be expected, among pre-training measures, the strongest predictors were those most similar to the training tasks.

11:50 AM **Exploring the coding of temporal fine structure in normal and impaired hearing**

Andrew J. Oxenham and Christophe Micheyl
Psychology Department, University of Minnesota

A band-limited signal can be decomposed into a rapidly varying carrier (temporal fine structure, or TFS) and a more slowly varying modulator (temporal envelope, or TE). For most acoustic signals the TFS can be represented in the auditory periphery by the spatial distribution of stimulation within the cochlea (place code) and/or by the precise timing of spikes within the auditory nerve (temporal code), at least up to the limits of phase-locking. Higher up in the auditory system, phase-locking is observed up to only a few hundred Hertz, suggesting that TFS is probably coded via a tonotopic or population code beyond the cochlear nucleus.

Recent studies in normal and impaired hearing, as well as acoustic simulations of cochlear-implant processing, have suggested a special role for TFS in coding pitch and speech, particularly in challenging acoustic situations. Most importantly, some studies have suggested that many people with cochlear hearing loss may have a selective deficit for the temporal coding of TFS, which results in poorer pitch perception and a reduced ability to understand speech in complex backgrounds. At present, the physiological origins of this deficit remain unclear, and recent data show intact phase locking on auditory-nerve fibers following cochlear damage. Nevertheless, the potential importance of temporally coded TFS in the challenges faced by hearing-impaired individuals has led to the proposal of clinical tests for the diagnosis of TFS coding deficits.

Here we examine various ways in which researchers have attempted to separate TFS from TE processing, and to distinguish between temporal and rate-place (spectral) processing, in the auditory periphery. We show that most, if not all, methods designed to test temporal coding of TFS in speech and pitch perception can also be explained by the alternative mechanisms of TE and/or spectral place coding. Specifically, for listeners with cochlear hearing loss, the deficits that have been ascribed to a failure of temporal TFS processing may instead be explained by broader cochlear tuning that is known to occur with damaged or dysfunctional outer hair cells. Our findings do not rule out the importance of temporal coding in the auditory system, but they suggest that it is not yet necessary to postulate additional TFS deficits to explain the differences in pitch and speech perception between normal-hearing and hearing-impaired individuals. [Work supported by NIH grant R01 DC 05216.]

FRIDAY, AUGUST 13

SESSION SIX

Algorithms & Algorithm Performance

Moderator: Pamela Souza

5:15 PM **Speech intelligibility improvements in hearing aids using adaptive bilateral and binaural multichannel Wiener filtering based noise reduction**

Bram Cornelis and Marc Moonen
Dept. of Electrical Engineering (ESAT-SCD), Katholieke Universiteit Leuve
Jan Wouters, Dept. of Neurosciences (ExpORL), Katholieke Universiteit Leuven

A major problem for hearing aid users is the degradation of speech intelligibility in a noisy environment. Previous studies have shown that noise reduction by the Speech Distortion Weighted Multichannel Wiener Filter (SDW-MWF) can significantly improve speech intelligibility.

In future binaural hearing aids microphone signals will be exchanged between the left and the right device to generate an output signal for each device. As more microphone signals are available in a binaural noise reduction procedure, a further intelligibility improvement can be achieved compared to a bilateral procedure where the hearing aids operate independently. Previous studies have shown that an (idealized) binaural SDW-MWF indeed allows for significant intelligibility improvements compared to bilateral noise reduction.

In this study, a perceptual evaluation of the speech intelligibility obtained with five different bilateral and binaural noise reduction algorithms is performed. In contrast to previous studies which evaluated off-line versions of the SDW-MWF, the algorithms are implemented on a real-time research platform, so that realistic adaptive versions of the bilateral and binaural SDW-MWF can be evaluated. As the SDW-MWF relies on a voice activity detector (VAD), a novel binaural VAD was also implemented. To study the impact of VAD errors, the SDW-MWF algorithms are evaluated both for an ideal VAD (as in previous work) and for this realistic VAD. A bilateral fixed beamformer and an unprocessed reference condition are also included in the study.

The test subjects are seated in a room with livingroom-like reverberation, where four different spatial scenarios are created: two single-noise scenarios, a pseudo-diffuse (three interferers) noise scenario, and a very realistic scenario which is representative for a cafeteria situation. The subjects wear hearing aids of which the microphone signals are processed by the real-time platform. The generated outputs are redirected to the receivers of the hearing aids. Two groups of ten normal hearing subjects and one group of five hearing impaired subjects took part in the study.

The results show that the SDW-MWF algorithms are capable of significantly increasing the speech intelligibility in several scenarios, even for the realistic implementation with real VAD. In some scenarios there is also a significant binaural benefit, i.e. the binaural SDW-MWF outperforms the bilateral algorithms.

5:35 PM **Enhancement of spectral changes over time – method and initial assessment**

Jing Chen, Thomas Baer, and Brian C.J. Moore
Department of Experimental Psychology, University of Cambridge, UK

In previous studies, several attempts have been made to compensate for the effects of reduced frequency selectivity of the hearing impaired by sharpening of spectral peaks or enhancing spectral contrast in individual brief signal segments. However, most information in speech is carried in the spectral changes over time, rather than in static spectral shape per se. Here we describe a new form of signal processing aimed at enhancing spectral changes over time and we present an initial assessment using perceptual tests.

The signal processing strategy was based on the fast Fourier transform (FFT): the input sound signal was segmented into frames, windowed, spectrally smoothed, spectral-change enhanced, and then resynthesized with the overlap-add technique. The spectral change across every two adjacent frames was evaluated by taking the ratio of the magnitude-spectrum values and was enhanced by modifying the magnitude spectrum based on the spectral change of preceding frames, using a Difference of Gaussians (DoG) function. The processing was implemented with four adjustable parameters: b, the width of the DoG function specified in ERB_N number (starting value 1 ERB_N); ξ and m, controlling the effect of preceding frames; and S controlling the amount of enhancement.

Preliminary perceptual tests with both normal-hearing and hearing-impaired subjects were conducted to assess the possible benefits for speech intelligibility of enhancing spectral changes. Ratings of speech quality and intelligibility were obtained using several variants of the processing scheme, with different processing parameters. The results suggest that the processing strategy may have advantages for hearing-impaired people for certain sets of parameter values. [This work was supported by a Newton International Fellowship from the Royal Society and the Royal Academy of Engineering and by the MRC].

5:55 PM **Predictive measures of the intelligibility of speech processed by noise reduction algorithms**

Karolina Smeds[1], Arne Leijon[2], Florian Wolters[1,3], Anders Nilsson[1,2], Sara Båsjö[1], and Sofia Hertzman[1]
[1] ORCA Europe, Widex A/S, Stockholm, Sweden; [2] KTH, Stockholm, Sweden; [3] University of Applied Sciences, Oldenburg, Germany

Noise reduction (NR) is commonly used in modern hearing aids. Previous measurements (Smeds et al. 2009, ISAAR, Helsingør) have shown that hearing aid NR algorithms function in very different ways. To evaluate the effects of various NR algorithms, laboratory or field tests are usually performed with listeners with or without hearing impairment. It would be of great value if computational predictive measures could be used to indicate the effects of NR algorithms prior to testing with listeners.

The aim of the current study was to evaluate a number of potential computational measures related to speech intelligibility to see to what extent they can predict the effect of NR for listeners with impaired and normal hearing.

Twenty listeners with mild-to-moderate hearing loss and ten listeners with normal hearing participated in a laboratory study. An adaptive speech recognition test using sentences in babble noise was used. The speech test produce gives results in terms of physical signal-to-noise ratios (SNR) that correspond to equal speech recognition performance with and without the NR algorithms. This facilitates a direct test of how well the predictive measures agree with the experimental results. A good predictive measure should produce the same result for all test conditions.

Three NR algorithms plus the reference unprocessed condition were compared using preprocessed sound files presented in a sound-treated test booth. For the listeners with hearing impairment, individualized gain was provided using tightly fitted linear hearing aids. Two of the NR algorithms were generic “text book” algorithms based on Wiener filtering and spectral subtraction, and the third NR algorithm was a spectral subtraction algorithm fine-tuned for hearing aid use.

The experimental results were used to evaluate a number of predictive measures, including a standard Speech Intelligibility Index (SII) method (ANSI S3.5 1997) and two extensions of the method, one time-variable SII method (Rhebergen et al. 2006, J Acoust Soc Am 120(6): 3988-97) and one coherence-based SII method (Kates & Arehart 2005, J Acoust Soc Am 117(4): 2224-37). Further, one glimpsing model of speech perception (Cooke 2006, J Acoust Soc Am 119(3): 1562-73) and one information theoretic approach (Stadler et al. 2007, Interspeech, Antwerpen) were used.

The ability of the computational measures to predict the effects of NR will be presented and discussed.

6:15 PM **Impact of wearer's physical characteristics on first-order hearing aid directionality**

Jay Desloge[1], Ivo Merks[2], Tao Zhang[2], Pat Zurek[1]

1 Sensimetrics Cooperation

2 Starkey Laboratories, Inc.

Hearing aid wearers often report limited or no benefit from a directional microphone in a hearing aid. One hypothesis for the lack of benefits could involve the wearer‘s physical characteristics (e.g., body, head, and ear shape/size) and their impact on the directional response of the hearing aid. The purpose of this research was to study the impact of individual-wearer physical differences upon the frequency-dependent directional characteristic of hearing-aids with first-order directional microphones.

Acoustic and physical measurements were taken on 20 human subjects plus the KEMAR manikin. Directional microphone responses were obtained by combining microphone signals from either dual-omni BTE aids or dual-omni ITE aids. A single pair of BTEs and a single pair of ITEs were used for all subjects (and KEMAR). These measurements were then analyzed to identify correlations and dependencies between acoustic directionality and subject physical characteristics. Directivity Indices (DIs) were partitioned into *a body effect* (the directional gain from free-field to front microphone) and a *processing effect* (the directional gain from front microphone to hearing aid output).

The measurements showed remarkably small variations in DIs across wearers. The ITE provided a *DI* of 5-10 dB between 0.5 and 10 kHz. The BTE *DI*s were generally lower than those of the ITE: only 0-5 dB between 0.5 and 4 kHz with a band of slightly higher *DI*s around 2 kHz and less 0 dB above 4 kHz. The body effects were similar for BTE and ITE aids below about 2 kHz. Above 2 kHz, the BTE body effect was negative while the ITE body effect was clearly positive, possibly due to pinna effects. The processing effect across subjects roughly mirrored the body effect, reflecting the fact that the total DI (the sum of body and processing effects) was largely constant across subjects. Finally, the following trends, although weak, were observed:

1. Higher pinna flare was associated with larger DIs in the ITE configuration
2. Presence of eyeglasses was associated with lower DIs in the ITE configuration
3. Higher shoulder and torso size was associated with higher DIs in the BTE configuration
4. Higher ear-canal-to-shoulder distance was associated with lower DI at 2 kHz in both the BTE and ITE configurations

The results of the present study suggested that physical characteristics were not a major contributor to inter-user variability in first-order directional aid benefit.

6:35 PM Pinna cue-preserving hearing-aid fittings: Field-testing of benefit

Tobias Neher, Søren Laugesen, René Burmand Johannesson, Niels Søgaard Jensen and Louise Kragelund
Eriksholm, Oticon Research Centre

It is well established that our pinnae act as direction-dependent acoustic filters, imposing spectral peaks and notches on incoming sound that are important for spatial hearing purposes. Since these pinna cues are best preserved at a listener's ear-canal entrances, it should, in principle, be advantageous to fit listeners with hearing aids having microphones at corresponding locations. Nevertheless, a large number of hearing-aid wearers are, for a number of other reasons, fitted with behind-the-ear (BTE) devices. To investigate the benefit obtainable from preserving pinna cues in hearing-aid fittings, a randomised cross-over study was carried out. Based on the outcome of an initial candidature study (also to be presented at this conference), 20 test subjects with mild-to-moderate, symmetrical, gently sloping hearing losses were selected to participate in a field test. All test subjects were fitted with custom-made, experimental hearing aids that allowed switching in software between a pinna cue-preserving microphone location and a BTE microphone location. Gains were prescribed based on a procedure (also to be presented at this conference) that ensured sufficient audibility of all test stimuli up to 8 kHz. All test subjects were then acclimatised to each microphone location for at least four weeks, after which the following outcome measures were administered: (1) spatial release from speech-on-speech masking along both the front-back and left-right dimension, (2) a localisation test with emphasis on front-back discrimination performance, and (3) a modified version of the Speech, Spatial and Qualities (SSQ) questionnaire that facilitates direct comparison of different hearing-aid fittings. Finally, to provide an objective basis for evaluating benefit from pinna cue-preserving hearing aids, in-situ head-related transfer function measurements were made for each test subject, microphone location, and angle of sound incidence. This presentation will focus on the results from the field test, which at the time of writing is still underway.

SATURDAY, AUGUST 14

SESSION SEVEN

Outer and Middle Ear Acoustics

Moderator: Harvey Dillon

8:00 AM High-frequency biomechanics, diagnostics, and hearing aids for the middle ear

Sunil Puria
Stanford University

The broadest frequency range of hearing among vertebrates is achieved by mammals, whose upper limits vary from 10 kHz (elephants) to 90 kHz (mouse), and even higher for animals employing echolocation. This ability to sense high frequencies provides a critical means for localizing sound (Masterson *et al*. 1969, JASA), and is related to many of the biomechanical features that distinguish mammals from other vertebrates, including the presence of distinct radial and circumferential collagen fiber layers of the tympanic membrane, the presence of three distinct middle-ear bones, the elongation of the basilar membrane, and the motility of the outer hair cells in the organ of Corti. In this talk we argue that the morphometries of the tympanic membrane and malleus-incus complex are specially adapted to one another in mammals so as to optimize high-frequency hearing (Puria and Steele 2010, Hear. Res.), which is consistent with the hypothesis that the radial collagen fibers of the tympanic membrane play an integral role in effective high-frequency sound conduction (O'Connor *et al.* 2008, Laryngoscope). While the coupling of these two systems for high frequency transmission has potential clinical significance, diagnostic information for both sensorineural and conductive hearing pathways is typically unavailable above 6 kHz due to the way bone-conducted hearing is measured. To address this, we report measurements using new magnetostrictive headphones (TEAC HP-F100 and -F200) that enable valid bone conduction measurements up to 16 kHz in subjects with thresholds as high as 80 dB HL (Popelka *et al*. 2010, Hear. Res.). Such high-frequency information could allow conventional and frequency-transposition hearing aids to perform better, and could improve our understanding of the outcomes of middle-ear surgical repairs. An exciting area of development has been in the field of middle-ear hearing aid technologies, where energy is directly coupled to the eardrum, ossicles, or cochlea, but where surgery is often required to implant a transducer. We will review the different technologies in this area, including one non-surgical design with the potential to improve high-frequency performance over conventional hearing aids, in which light is used to wirelessly transmit both the power and signal to a transducer contacting the tympanic membrane (Perkins *et al.* 2010, Hear. Res.). [Work supported in part by R01 DC 005960, SBIR R44 DC008499, and ARRA supplemental funds all from the NIDCD of NIH.]

8:40 AM The potential of acoustic modeling

Morten A Nordahn and Mads J Herring Jensen
Widex A/S, Nymoellevej 6, DK-3540 Lynge, Denmark

One of the cornerstones of hearing instrument (HI) fitting is acoustic accuracy. Without control of the sound at the eardrum, there is a risk that the HI user is insufficiently or overly compensated for their hearing loss. This might cause well researched and opti-

mized hearing instrument signal processing features to become ineffective or even inaudible. Deficient management of the acoustics may thus increase the variation in the HA user's performance and satisfaction. Therefore, it is important for developers to understand the principles behind the resulting sound at the eardrum or around the hearing aid of the individual user.

For this purpose, acoustic modeling is an excellent tool, which enables researchers to analyze the challenges in shaping the geometry of directional microphones or obtaining the intended gain at the eardrum of the individual HI user. The goal of this talk is to review and illustrate different approaches to acoustic modeling, and discuss and visualize their potential and limitations.

The Transmission Line Model (TLM) is an accurate modeling tool for studying plane-wave acoustics in tubes and cavities. The TLM is based on building blocks for each unique part of the acoustic HI system, e.g., the receiver, tubing or the ear canal of the individual user. These blocks are combined to calculate a certain frequency dependent transfer function, e.g., the real-ear insertion gain. The TLM approach is computationally fast and particularly strong in analyzing the influencing factors on the sound pressure at the eardrum of individual users, or for explaining observations in fitting verification measurements.

For sound fields around more complex geometries such as the head or the hearing aid shell, the accuracy of the TLM approach falters, and the more computationally demanding Finite-Element modeling (FEM) approach takes over. With FEM, acoustic calculations are based on a set of general acoustic equations and a complete three-dimensional CAD model of e.g., the hearing aid on a KEMAR head. FEM has numerous applications with regards to HIs, e.g. in the design phases of new HI shapes or in analysis of the feedback path from vent opening to hearing aid microphone.

In combination, the two methods provide researchers with a fast and flexible tool, which may be used to further strengthen the fusion between acoustics, hearing aid design and signal processing algorithms for optimization of the in-situ HI performance.

9:15 AM **Pressure Lies**

Daniel Warren and Charles King
Knowles Electronics

At frequencies below about 5 kHz, the sound pressure to which a hearing aid wearer is exposed is adequately determined by the sound pressure level measured at a single probe position in the ear canal. At higher frequencies, where standing waves are developed in the canal, measured sound pressure levels vary markedly with probe insertion depth. Various methods have been developed to cope with this variation, including extrapolation of sound pressure level from the probe depth to the TM via modeling and simple assumptions, or by first determining the termination impedance of the ear canal, measuring a quantity such as forward pressure level or sound intensity level which is relatively independent of probe depth, then determining sound pressure at the canal. The question remains how relevant any of these measures are, however, since the depth of the TM, which is tilted with respect to the canal axis, is ill-defined. Furthermore, at even higher frequencies (> 8 kHz), acoustic radial modes develop and all bets are off for uniformity of sound pressure with radial position in the canal.

Similar issues exist for measurement couplers used to characterize hearing aids and hearing aid receivers. The 2cc coupler (IEC 60318-5) is much larger in volume than most occluded ear canals and some designs support radial modes at frequencies as low as 6 kHz. The coupler formerly known as the 711 (IEC 60318-4) was well designed to imitate the impedance of a typical adult ear canal at the reference plane, but the designers of B&K's ear simulator admit that correctly reproducing the sound pressure at the TM was only a secondary consideration.

These measurement issues are becoming increasingly pertinent as receivers, hearing aids, and insert earphones for personal audio are being designed to perform at higher frequencies, well beyond the traditional 6 kHz limit for hearing aids. In this study we will use computer models to examine the alternative measurement possibilities and data reporting techniques in search of more robust and meaningful characterization of high frequency performance of ear-canal loaded devices.

SATURDAY, AUGUST 14

SESSION EIGHT

Tinnitus Management

Moderator: Tim Trine

11:10 AM **Present and future directions for tinnitus treatment**

Richard S. Tyler
The University of Iowa, Dept of Otolaryngology-Head and Neck Surgery and Communication Sciences and Disorders

Tinnitus can be devastating in many patients, causing with emotional, hearing, sleep and concentration problems. Psychophysical procedures such as Minimum Masking Level are helpful for treatments focused on reducing the magnitude of the tinnitus. Questionnaires such as the Tinnitus Handicap Questionnaire and a new Tinnitus Activities Questionnaire are helpful for treatment focused on reducing reactions to tinnitus. There are no medications or dietary supplements that have been shown to reduce the magnitude of the tinnitus. Several counseling and sound therapies have been shown to reduce the reactions to tinnitus. We utilize a picture-based Tinnitus Activities Treatment. A variety of new sound therapies show promise. For decades, the use of electricity presented to the cochlea has been shown to reduce tinnitus in some patients, and there are several ongoing trials to suppress tinnitus in the cochlea and in the brain. Magnetic stimulation of the brain can also suppress tinnitus in some patients, but its clinical application is uncertain. Finally, there is now a greater appreciation that there are different subgroups of tinnitus patients, and the careful selection of subgroups is now being applied to new drug trials.

11:50 AM **Advanced hearing instrument technology for tinnitus management**

Ole Dyrlund
GN Resound

One of the more accepted tinnitus treatment options is sound therapy, where ear level amplification of environmental sounds and / or artificial sounds generated by a tinnitus sound generator (TSG) are the focal point. In addition to the ear level device, other sound generating devices, such as sound pillows, radio, TV, etc. can be used to supplement.

Often sound therapy is an integrated part of a more comprehensive treatment plan involving counselling and therapy e.g. Tinnitus Retraining Therapy (TRT).

Administrating sound therapy helps the patient to habituate from the tinnitus and ultimately, over time, the tinnitus is considered less important, and relief is achieved.

As the large majority of patients with tinnitus also have a hearing loss, hearing instrument amplification can be useful in tinnitus treatment, especially when implemented in open fitting instruments. (Del Bo et.al 2006) offering a number of unique benefits:

- High frequency amplified sound is combined with direct natural sound low frequency sound providing more natural environmental sounds without annoying occlusion feeling.
- Due to the size of the instrument and the soft coupling to the ear canal, these instruments offer a high degree of wearing comfort and cosmetic appeal and don't attract the users attention to the instrument and the tinnitus.

A combination of open amplification and a TSG may provide additional benefit as the TSG can provide sound input in situations where the amplified acoustic input from the environment is insufficient. Recent research indicates promising results from testing a prototype open amplification device with sound generator features (Carrabba et. al 2010).

Typically random noise is tailored to the needs of the patient low and high pass filtering to shape frequency spectrum of the noise to optimize listening comfort and/or focusing the frequency range of the noise to the pitch of the tinnitus. Modulation of the noise level may be beneficial creating a more soothing sound experience. Alternatively synthesis of "natural" sounds like the sound of rain or ocean waves may be implemented.

Other common features are manual control of the signal level and multiple listening programs.

Automatic features, which create less focus on the device, may enforce habituation from the tinnitus.

An example is automatic signal level increase in quiet environments, where the tinnitus is more easily recognized, but lowering of the level e.g. in different speech situations to avoid masking.

SATURDAY, AUGUST 14

SESSION NINE

Plasticity and Implantable Hearing Systems

Moderator: Kevin Munro

5:15 PM Auditory neuroplasticity in children with hearing impairment

Anu Sharma
University of Colorado

Sensory deprivation, as in congenital deafness, prevents normal development of the central auditory pathways, eventually resulting in deficits in oral language learning for children with hearing loss. However, early intervention with hearing aids and/or cochlear implants may make it possible to avoid some of the harmful effects of auditory deprivation. Using electrophysiological and brain imaging techniques (such as EEG, CAEPs, and MEG) we have examined the trajectories and characteristics of deprivation-induced and experience-dependent plasticity in the central auditory nervous system of children with sensorineural hearing loss. The findings of sensitive periods for cortical plasticity in humans are generally consistent with those reported from animal models. Plasticity can be inherently both harmful and beneficial. One aspect of plasticity that works against children and adults with long term auditory deprivation is the re-organization of higher order cortex by other sensory modalities such as vision and somatosensation. Recent studies also suggest that cortical re-organization in hearing impaired children and adults result in functional activation of certain cognitive circuits which are predictive of behavioral outcomes. [Supported by NIH/NIDCD]

5:55 PM Signal processing for combined hearing aid and cochlear implant stimulation

Tom Francart, Anneke Lenssen and Jan Wouters
ExpORL, Dept. Neurosciences, K.U.Leuven

Due to the success of cochlear implantation there is a booming population of implantees that use a hearing aid (HA) contralateral to their cochlear implant (CI). This is called bimodal stimulation. However, the signal processing of the CI and HA was developed independently and precludes several important potential advantages of binaural stimulation.

Localization of sound sources and binaural unmasking of speech in noise are in normal hearing listeners strongly dependent on the perception of binaural cues: interaural level and time differences (ILD and ITD). Psychophysical tests with 10 bimodal listeners indicated that they were sensitive to ILD with an average just noticeable difference (JND) of 1.7dB. We also found that loudness growth functions between electric and acoustic hearing could be well approximated by a linear relationship between current in uA and acoustic level in dB.

Psychophysical tests with 8 bimodal listeners indicated that those subjects with an average acoustic hearing threshold at 1000 and 2000Hz of 100dBSPL or better were sensitive to ITD for different single- and multichannel stimuli with JNDs in the order of 100-250us.

As JNDs in both ILD and ITD are within the range that is physically available, these cues should be usable for sound source localization or binaural unmasking if they are properly transmitted by the CI and HA. In current clinical devices there are two main issues to be addressed to enable the transmission of binaural cues: temporal synchronization and loudness balancing. To enable ITD perception, a sound source right in front of the subject should have an ITD of 0us at auditory nerve level. Therefore the devices need to be synchronized and the electric signal needs to be delayed to compensate for the acoustic traveling wave delay. We found this delay to be in the order of 1.5ms.

Both ILD and ITD perception are optimal when the loudness is equal at the two ears for a sound source in front of the subject. Additionally, perceptual loudness growth should be the same at both sides. In actual devices this involves synchronization of automatic gain control circuits, measuring a loudness growth function per subject and compensating for it in the signal processing.

We will present an overview of our basic psychophysical measurements of ILD and ITD perception with bimodal hearing and the first results of a novel bimodal speech processing algorithm that ensures synchronized and balanced stimulation and additionally emphasizes ILD and ITD cues.

6:25 PM **Speech intelligibility in various noise conditions with the Nucleus 5 sound processor**

Norbert Dillier and Wai Kong Lai
Laboratory of Experimental Audiology, ENT Department, University Hospital Zurich, Switzerland

Speech recognition performance of cochlear implant recipients has improved significantly over the last years. Test material and procedures developed for evaluation of hearing instrument algorithms can now often be used for this group of patients as well. This study evaluates two preprocessing options for the recently introduced Nucleus 5 Sound Processor (CP810, Cochlear) suggested to be used in noisy environments. The device contains two omnidirectional microphones which can be configured as a directional microphone combination (called “focused sound”) or as an adaptive beamformer (adjusting the directivity continuously to maximally reduce the interfering noise). Sentence recognition in noise was assessed for the “focus” and “beam” preprocessing options as well as the omnidirectional reference setting for various sound field conditions using a standardized speech in noise matrix test (Oldenburg sentences, OLSA). 8 subjects who previously had been using the Freedom speech processor and subsequently were upgraded to the CP810 device participated in this series of evaluation tests. The SRT50 was determined using sentences presented via loudspeaker at 65 dB SPL in front of the listener and noise presented either via the same loudspeaker (S0N0) or at 90 degrees at either the ear with the sound processor (S0NCI+) or the opposite unaided ear (S0NCI-). The fourth, diffuse noise condition consisted of three uncorrelated noise sources placed at 90, 180 and 270 degrees (S0NDSF). The noise level was adjusted by an adaptive procedure to yield a signal to noise ratio where 50% of the words in the sentences were correctly understood. For the S0N0 sound presentation the median SRT50 values were about -1.7dB irrespective of the preprocessing condition. For spatially separated speech and noise sources the SRT50 for the “beam” option ranged from -11.3dB (S0N+CI) to -7.8dB (S0NDSF) whereas for the “focus” option the resulting SRT50 ranged from -9.5 (S0N-CI) to -2.25 (S0N+CI). These performance levels compare favorably with results obtained by normal hearing subjects in the same test conditions (about -13.5dB for spatially separated target and noise sources).

SUNDAY, AUGUST 15

SESSION TEN

Cognitive Aspects, Rehabilitation

Moderator: Susan Scollie

8:15 AM **Speech Referenced Limiting (SRL): a new approach to limiting the sound level from a hearing aid**

Michael Fisher, Nicky Chong-White and Harvey Dillon
National Acoustic Laboratories

Hearing aid users, with access to a volume control, typically set the volume so that conversational speech is at a comfortable level. When fixed at this volume setting everyday sounds such as a door slam, a telephone ring, music in a shop, an umpire's whistle or the sound of washing up pots and pans can sometimes be too loud.

Conventional methods of sound level limiting in hearing aids normally involve setting the limit to just below the level at which loudness discomfort occurs for the listener. If this limiting level is low relative to the amplified speech level then the quality of the speech will be reduced. On the other hand if this limiting level is high relative to the amplified speech level then sounds of a higher level than speech will be amplified to levels above that of the speech making them less comfortable.

A new approach to sound level limiting is to use the level of the speech that the hearing aid user has recently or is currently hearing as a reference and reduce the level of non-speech sounds with respect to this reference. This novel method is called Speech Referenced Limiting (SRL). The limiting level is adaptive and is set by the loudness of the speech to which the hearing aid user is acclimatised. When done on a frequency specific basis the umpires whistle is reduced to the level of the treble of a recent conversation and the rumble of a truck to the level of its bass. This is achieved by estimating the loudness of speech at different frequencies to produce a speech reference and limiting sound that exceeds this reference. The method may be used in multi-band hearing aids that have a speech program. Details of the SRL scheme and experimental data on the effects of SRL on speech and noise are presented. [The authors acknowledge the financial support of the HEARing CRC, established and supported under the Australian Government's Cooperative Research Centres Program.]

8:35 AM **Relationship between cognitive and speech-perception performance for elderly listeners with normal hearing**

Christian Füllgrabe[1], Brian C.J. Moore[2]
MRC Institute of Hearing Reserach
University of Cambridge

Sensorineural hearing loss leads to degraded speech intelligibility. The standard treatment is via hearing aids with frequency-specific amplification and amplitude compression. However, many hearing-aid candidates are elderly, and some of the speech-perception deficits may be related to age rather than hearing loss *per se*. The potential link between an aging sensory/cognitive system and speech-perception deficits, especially in challenging listening conditions, has been acknowledged (CHABA, 1988) and is supported by increasing empirical evidence (e.g. Pichora-Fuller & Souza, 2003). However, the nature of the age-related deficit (modality-specific changes in sensory processing *vs.* general cognitive slowing of information processing) is still a matter of debate. Also, the effect of aging on speech processing has mainly been studied via group differences between elderly and young listeners with matched audiograms on a given task rather than by administering several tasks to the same group of elderly listeners. Moreover, often only a partial audiometric match between young and elderly listeners was obtained, and no in-depth assessment of the listeners' cognitive abilities was conducted.

The general aim of the present study was to further the understanding of the inter-individual variability in auditory perception/cognition observed in the elderly population (Pichora-Fuller & MacDonald, 2008), and hence the rehabilitative needs of elderly hearing-impaired listeners. Twenty elderly (≥ 60 years) listeners with audiometrically normal (≤ 20 dB HL) hearing thresholds between 0.125 and 6 kHz and six normal-hearing young listeners participated. The study combined the analysis of group differences with a correlational analysis of (i) speech-processing abilities, (ii) cognitive abilities and (iii) subjectively reported listening difficulties. The prevalence of problems in difficult listening situations was assessed using two questionnaires that are normally employed for the assessment of hearing-aid benefit. A battery of cognitive tests (Mini-Mental State Examination; Test of Everyday Attention; digit span test from the Wechsler Adult Intelligence Scale) was also administered. Identification performance will be obtained for speech tokens lowpass-filtered at 6 kHz and presented (i) in quiet, (ii) in a steady speech-shaped background noise, and (iii) amplitude-modulated speech-shaped background noise. **[**Support provided by the RNID Flexi-grant 2010-11 "Effect of aging on auditory temporal processing" and the MRC].

References

CHABA (Committee on Hearing, Bioacoustics and Biomechanics) (1988). Speech understanding and aging. J. Acoust. Soc. Am., 83, 859-893.

Pichora-Fuller, K., & MacDonald, E. (2008). Auditory temporal processing deficits in older listeners: From a review to a future view of presbycusis. In: T. Dau, J.M. Buchholz, J.M. Harte, & T.U. Christiansen (eds.) Auditory signal processing in hearing-impaired listeners, 1st International Symposium on Auditory and Audiological Research (INSAAR 2007). Denmark: Centertryk A/S, pp. 291-300.

Pichora-Fuller, K. & Souza, P. (2003). Effects of aging on auditory processing of speech. Int. J. Audiol., 42, 11-16.

8:55 AM Older adults, hearing aids and listening effort

Piers Dawes[*], Kevin Munro[*], Sridhar Kalluri[§], Nazanin Nooraei[§], Brent Edwards[§]
**Audiology and Deafness Research Group, Department of Psychological Sciences University of Manchester, UK*
[§]Starkey Laboratories Hearing Research Centre, Berkeley, CA, USA

Listeners may report a benefit in reduced 'effort' of listening when using amplification of certain signal processing strategies, although this benefit may not be reflected in traditional speech-in-noise tasks. Research with normally hearing young listeners found that Noise Reduction signal processing significantly reduced listening effort at certain SNRs, as measured by a dual task objective measure of listening effort (Sarampalis et al., 2009). The dual task measure included a primary speech-in-noise task and a secondary visual reaction time task, with reaction times on the secondary task taken as an index of listening effort.

In this study, a dual task was used to assess listening effort in older adult hearing aid users in quiet, +7 and 0 SNR background noise, both with and without hearing aids. Preliminary results suggested that hearing aids provided a significant benefit of reduced listening effort at +7 SNR, but not in quiet or at 0 SNR. Increased effort in background noise and ability to take advantage of amplification to reduce listening effort were examined by correlating listening effort with cognitive skills including performance IQ, speed of processing, working memory and attention switching. Significant correlations were found between reduction in listening effort and attention switching. Additionally, experienced hearing aid users were compared with novice hearing aid users, with the hypothesis that experienced hearing aid users would show a greater reduction in listening effort for aided versus unaided conditions than novice users. [Supported by Starkey Laboratories.]

Reference

Sarampalis, A., Kalluri, S., Edwards, B., & Hafter, E. (2009). Objective measures of listening effort: Effects of background noise and noise reduction. *Journal of Speech, Language and Hearing Research, 52*, 1230-1240.

9:15 AM **Factors affecting help-seeking, hearing-aid uptake, use and satisfaction – what do we know?**

Line Vestergaard Knudsen[1], Graham Naylor[1], Thomas Lunner[1], Claus Nielsen[1], Sophia E. Kramer[1,2]and Marie Öberg[3]

[1] *Eriksholm Research Centre, Denmark*

[2] *VU University Medical Center, Amsterdam, The Netherlands*

[3] *Linköping University, Sweden*

The uptake of hearing aids (HAs) and satisfaction with them has not changed dramatically over recent decades, despite huge advances in HA technology and improvements in objective 'performance' of HA users. It appears prudent to consider the influence of non-technical factors. In order to obtain further insights, we performed a review of the scientific literature. Our focus was on factors relating to people and processes, rather than those relating to HA technology, and the four outcome variables of interest were *help-seeking*, *hearing-aid uptake*, *hearing-aid use* and *satisfaction*.

This review covers papers published between 1980 and 2009. Structured searches in Pubmed and Cinahl databases, and inspection of reference lists of relevant articles, were initially used to identify several hundred publications, which reduced to 40 after the application of inclusion and exclusion criteria. Criteria having most filtering effect related to study quality (e.g. peer review, traceability of methods etc.), types of factors studied (e.g. HA features vs. processes) and domain of outcomes studied (e.g. objective benefit vs. use/satisfaction).

The 40 studies encompassed relations between 31 different factors and at least one of the four outcome variables. There were personal factors (e.g motivation, expectation, attitude); demographic factors (e.g. age, gender); and external factors (e.g. cost, use of counselling).

Only one factor had a positive effect on all outcome variables. This factor was *self-reported activity limitation* (i.e. self-reported hearing problems).

There was agreement that the *degree of objective hearing loss* influenced *HA uptake*, but the literature showed no consistent results with regard to its influence on *HA use* and *satisfaction*. Interestingly the vast majority of published studies showed no relationship of *age* or *gender* with any of the four outcome variables.

Only four factors apparent during the primary HA fitting session were assessed, and only two of these concerned aspects of the fitting process itself.

Factors studied so far explain only small proportions of the variance in outcomes.

We conclude that firm knowledge is almost absent regarding how what the professional does with the patient affects *HA use* and *satisfaction*. Much more research into this is needed.

SUNDAY, AUGUST 15

SESSION ELEVEN

The Legacy of Stuart Gatehouse

Moderator: Graham Naylor

10:00 AM **Cognitive Hearing Science: The legacy of Stuart Gatehouse**

Jerker Rönnberg[1,2,3], Mary Rudner[1,2,3] and Thomas Lunner[1,2,3,4]
1 Department of Behavioral Sciences and Learning, Linköping University
2 The Swedish Institute for Disability Research, Linköping University
3 Linnaeus Centre HEAD, Linköping University
4 Oticon A/S, Research Centre Eriksholm, Snekkersten, Denmark

Stuart Gatehouse was a true pioneer of cognitive hearing science. He forcefully argued that cognition plays a vital role in the field of audiology, perhaps especially when it comes to the interaction between signal processing in hearing aids and cognitive function.

The ease of language understanding (ELU) model was used as a cognitive hearing science background in the two studies reported. Three parameters determine ELU (Rönnberg et al., 2008): the quality and precision of phonological representations in semantic long-term memory (SLTM), phonologically mediated lexical access speed, and explicit, *storage* and *processing* resources in working memory (WM). If there is a mismatch between phonological information extracted from the signal and the phonological information represented in SLTM, WM is assumed to be invoked to infer and construct the meaning of a message.

We present data from a sample of persons with hearing impairment drawn from the Swedish prospective cohort aging study called Betula (Nilsson et al. 2004). The memory data from Betula were used to test the ELU hypothesis which states that in the mismatch situation, less information will be encoded into episodic long-term memory (ELTM, Tulving, 1983). This will lead to an increasing *disuse* of ELTM and a subsequent decline of ELTM function while WM will remain intact because of its explicit use in compensating for the mismatch. A measure of short-term memory (STM) *storage* was used as a proxy for WM. Structural equation modeling demonstrated that (a) ELTM and SLTM are negatively associated with hearing impairment, whereas STM is not, (b) that SLTM is strongly related to ELTM, and (c) age and hearing impairment contribute independently to ELTM deficits. Rival hypotheses are discussed.

In a second study of recall of final (repeated) words in sentences, we show for participants with hearing impairment that the effects of background noise (steady-state noise and four-talker babble) interact with noise reduction schemes (realistic and ideal binary masking; Boldt et al., 2008; Wang et al., 2009) vs. no noise reduction (only linear amplification) conditions, such that the noise reduction conditions show an equal memory facilitation in the four-talker babble noise. No such effect was found in steady-state noise. Serial position analyses of the most recently presented final words (an index of STM) and the pre-recency positions (indexing ELTM) will be presented.

The studies show that both hearing impairment and signal processing influence perception and encoding into learning and memory systems. The results have clinical implications.

Poster Program

Posters for Session A should be put up by 8 AM Thursday, August 12, and taken down after 10 PM Thursday, August 12, or before 7 AM Friday, August 13. Presenters should be at their posters from 9:45 AM – 11:10 A.M.; 4:30 PM – 5:00 P.M.

POSTER SESSION A

Thursday 9:45 AM – 11:10 AM

A1

Correcting for impaired auditory nerve fiber group delays could improve some features of the spatiotemporal discharge pattern

Timothy J. Zeyl and Ian C. Bruce, McMaster University

Auditory nerve fibers in an ear with outer hair cell damage can be conceptualized as filters having a broadened frequency response area, a shallower phase response and a shorter group delay with respect to a healthy fiber, particularly at low stimulus presentation levels. As well, the presence of inner hair cell damage requires increased stimulus presentation levels for restoration of fiber discharge rates, which results in broad auditory filters with shallow phase response and short group delay. As a consequence, the discharge times in the impaired ear in response to a tone stimulus are more coincident across a population of fibers with a range of characteristic frequencies. This behavior resembles the spatiotemporal response pattern in a healthy auditory periphery in response to loud stimuli and has been postulated as a potential correlate to loudness recruitment. The present study evaluates the potential for correction of the altered phase response in the neural firing pattern of the impaired ear by a hearing aid. We implement a version of the spatiotemporal pattern correction scheme presented by Shi *et al.* (J. Speech Lang. Hear. Res. 2006), which measures the instantaneous difference in group delay between a bank of model healthy and impaired auditory nerve fibers and inserts the corresponding delays into an analysis-synthesis gammatone filterbank in the hearing aid. Human testing of the processing scheme showed that listeners preferred unprocessed sounds over processed sounds and that no systematic improvement in speech intelligibility was provided by the processed speech.

We evaluate this processing scheme with a computational model of the auditory periphery (Zilany & Bruce, JASA 2006, 2007) in response to a synthesized vowel for a mild and a moderate-to-severe high frequency sloping hearing loss, both with mixed hair cell damage. Analysis indicates that there are some technical and conceptual problems associated with the processing scheme that need to be addressed. These include: i) a possible non-flat frequency response through the analysis-synthesis filterbank due to time-varying changes in the relative temporal alignment of filterbank channels, ii) group delay corrections that are based on potentially incorrect frequencies due to the spread of synchrony in auditory nerve responses, and iii) modulations of frequency in the processed signal created by the insertion of delays resulting in the presence of abnormal frequencies of auditory nerve synchronization. Despite these issues, evaluation with an error metric derived from auditory nerve response cross-correlations shows that this processing scheme has the potential to improve performance at some sound pressure levels if the technical limitations are addressed sufficiently. [This work was supported by NSERC Discovery Grant 261736 (Bruce) and an NSERC Canadian Graduate Scholarship (Zeyl).]

A2

An analysis of wind noise at the front and rear microphones of hearing aids

Justin Zakis and Frauke Schall, Dynamic Hearing Pty Ltd, Australia

Wind noise attracts some of the lowest hearing-aid satisfaction ratings, with only 58% of surveyed respondents having some degree of satisfaction in wind noise (Kochkin, 2010). Despite the magnitude of this problem, there have been few published studies on the characteristics of wind noise with hearing aids. Most previous studies have shown wind noise spectra measured at a single microphone output at relatively low wind speeds (approximately 2-5 m/s), or at the receiver output over a wider range of wind speeds (approximately 4.5-13.5 m/s). However, at the higher wind speeds the output measurements were affected by limiters in the hearing aid that obscured the input level and wind speed at microphone saturation. Microphone saturation may lead to the masking of high-frequency speech by wind, and affect the ability of hearing-aid algorithms to satisfactorily detect and suppress wind noise. In addition, while many hearing aids switch to the front omni-directional microphone in wind, the relative wind noise levels at the front and rear microphones may vary with frequency, microphone placement, wind speed, wind azimuth, and wind shield design. The current study investigates these issues through an analysis of wind noise at the outputs of the front and rear microphones of different hearing-aid shells at three wind speeds (3, 6 and 12 m/s) and eight wind azimuths (45 degree increments). A mock completely-in-the-canal device and two behind-the-ear (BTE) shells were used: the larger BTE had exposed microphones, while the smaller BTE had mechanical wind shielding. Microphone saturation was achieved and wind noise levels between the front and rear microphones differed depending on the conditions. The results and implications for hearing-aid designers and users will be discussed.

Kochkin, S. (2010). MarkeTrak VIII: Consumer satisfaction with hearing aids is slowly increasing. The Hearing Journal, vol. 63, no. 1, pp. 19-20, 22, 24, 26, 28, 30-32.

A3

Adaptive noise reduction combining MBSS and classification for digital hearing aids

Sunhyun Yoo[1], Jonghee Han[1], Yoon Sang Ji[1], Heepyung Kim[1], Sangwan Seo[1], Sung Hwa Hong[2], Sang Min Lee[3], Dong Wook Kim[4], Sun I. Kim[1] and In Young Kim[1]

[1] *Department of Biomedical Engineering, Hanyang University*

[2] *Department of ORL-HNS, Sungkyunkwan University*

[3] *Department of Electronic Engineering, Inha University*

[4] *Bio & Health Lab, Samsung Advanced Institute of Technology*

In digital hearing aids (DHA), noise reduction is indispensable to improve the speech quality and intelligibility. However, it is still required to develop a robust algorithm in various noise circumstances for DHA. Recently, quite a number of studies have been performed for development of noise reduction algorithms applicable to different environmental conditions.

This study suggests an adaptive noise reduction algorithm which combined an environment classification system and multiband spectral subtraction(MBSS) in order to adaptively subtract noise components. The algorithm consisted of voice activity detection(VAD), classification Systems and 9 band spectral subtraction. For non-speech periods, environmental conditions were classified into four classes such as babble noise, car noise, traffic noise and white noise. From powers of the 9 octave bands, features were extracted and fed to support vector machine(SVM) using radial basis function(RBF) kernel to classify the environmental conditions. It could classify four different environments with 97% accuracy. Depending on the environmental conditions, the weight factor of each 9 bands were determined experimentally and multiplied to the noise reduction factor in MBSS. For the mixture of 1 IEEE-sentence with 35 different environmental noises, we evaluated the algorithm by SNR at -5dB, 0dB, 5dB and mean opinion score(MOS).

The proposed algorithms showed the highest improved SNR at -5dB. When it compared to existing SS or MBSS algorithm, SNR improvement ranged from 1dB to 5dB. The best improvement result of SNR could be seen in white noise and Car noise conditions. In MOS, when it compared to conventional SS or MBSS algorithm, the proposed algorithm got higher point in every condition. [This work was supported by grand No. 10031764 from the Strategic Technology Development Program of Ministry of Knowledge Economy].

A4

Objective longitudinal outcome assessment of hearing impaired Chinese children after early intervention

Ke Xu[1], Yun Zheng[1], Sigfrid D. Soli[2], Zhaoli Meng[1], Gang Li[1] and Kai Wang[1]

[1] *West China Hospital of Sichuan University, Chengdu, Sichuan, China*

[2] *House Ear Institute, Los Angeles, California*

Objectives: Objective longitudinal outcome assessment of deaf and hearing-impaired children after early intervention is necessary to establish evidence-based treatments. This is especially true in China with the largest pediatric population with hearing impairment in the world. A hierarchical battery for assessing speech and language development in young deaf children from 0 to 6 years old has been developed for this purpose. The battery includes measures of speech perception, speech production, and language development. An objective longitudinal study to assess young hearing impaired children fitted with hearing aids using this battery is currently underway at the West China Hospital (WCH) of Sichuan University in China.

Method: A Mandarin hierarchical battery of pediatric speech perception tests for research and clinical use with very young children has been developed. This battery is comprised of the Mandarin IT-MAIS (Zheng et al., 2009a), the Mandarin Early Speech Perception test (MESP) (Zheng et al., 2009b), and the Mandarin Pediatric Speech Intelligibility test (MPSI) (Zheng et al., 2009c). Speech production is evaluated with the Mandarin Speech Production test (MSPT). In addition, language development is evaluated with shortened versions of the Mandarin Child Development Inventory (MCDI), Words & Gestures for use with the youngest children, and the MCDI Words & Sentences, for use with older children. In addition, the revised version of the Revised Preschool Language Assessment test (PLA-R) for school age children.

Recruitment of hearing impaired subjects for this four-year longitudinal study began in early 2009 and will continue for two years, or until 120 subjects are recruited. Children under five years of age with bilateral moderate-profound non-syndromic sensorineural hearing loss who are fitted unilaterally or bilaterally with hearing aids at the WCH Hearing Center are invited to participate. Participants receive normal audiological and medical treatment, and are evaluated with the developmentally-appropriate speech perception tests from the hierarchy at the time of their hearing aid fitting and 3, 6, 12, 24, 36, and 48 months after fitting.

In addition, a sample of 50 normally-hearing 2-year olds will also be followed for the same time period. These subjects are administered the same assessment battery at baseline, 12, 36, and 48 months. Results for normally-hearing children will be used to define the normal developmental trajectory for speech perception, as measured longitudinally. This trajectory provides a reference for evaluation of longitudinal performance by the hearing impaired sample.

Results and Conclusions: To date, both the entire hearing impaired sample and the entire normally-hearing sample have been recruited. Approximately 40 hearing impaired subjects have completed baseline, 3-, 6- and 12-month evaluations. Preliminary analyses of their one-year speech perception results will be reported. These analyses will focus on the initial measures in the speech perception hierarchy, the Mandarin IT-MAIS and the MESP. Longitudinal results for these measures will be compared with published cross-sectional data from normally-hearing children (Zheng et al, 2009a, 2009b). Additional analyses will attempt to identify early audiological measures predictive of longitudinal development of speech perception approaching the normal developmental trajectory. These findings can help to establish evidence-based standards for pediatric audiology in the early identification and treatment of hearing impairment with hearing aids. [This project is sponsored in part by Widex A/S, Denmark, House Ear Institute, USA, and West China Hospital of Sichuan University, China.]

A5

A new American-English speech corpus for testing speech understanding in complex scenarios

William S. Wood and Sridhar Kalluri, Starkey Hearing Research Center, Berkeley, CA

Much current work investigates listener performance in complex scenarios using manipulations of multi-talker corpora such as the Coordinated Response Measure (CRM). The benefits of such corpora are linked to the different ways investigators can signal the identity of a target talker in a multi-talker stimulus. Current corpora are limited in usefulness, however, because their keywords do not span the range of sounds in conversational speech and because there is no simple way to take audibility into account when analyzing results for listeners with hearing loss. The current work introduces a new American-English speech corpus for testing speech understanding in complex scenarios. This corpus combines the flexibility and utility of corpora like the CRM with the benefits of a nonsense-syllable test by recording 12 different talkers uttering sentences with a CRM-like structure ("Ready NAME mark KEYWORD now") and using a full complement of vowel-consonant and consonant-vowel tokens as keywords. We describe the methods used in recording and editing the corpus, and provide data and analysis concerned with testing the accuracy of the Speech Intelligibility Index (SII; ANSI S3.5-1997) in predicting performance in conditions similar to those used by Pavlovic and Studebaker [1984. "An evaluation of some assumptions underlying the articulation index", J. Acoust. Soc. Am. 75 (5), 1606-1612]. Specifically, we measured percent correct with normal-hearing listeners in 5 different contexts (totaling 10 different conditions): 1.) 65 dB SPL speech in 2 levels of speech-shaped noise; 2.) 65 dB SPL speech low-pass filtered at 2 different cutoffs; 3.) 65 dB SPL speech high-pass filtered at 2 different cutoffs; 4.) 2 speech levels in a noise created to simulate a sloping, moderate high-frequency hearing loss; 5.) 2 levels of a high-frequency emphasized speech in the same noise as in the previous condition. These conditions target a range of percent correct (45%-90%) and SII values (0.16-0.60) commonly used in research. We discuss the results from these tests in terms of the accuracy of the SII in predicting the percent correct performance of the subjects.

A6

Speech understanding in noise with two vibrant soundbridge audio processors: A comparison of omnidirectional and directional microphones

Mario Wolframm[1,2], Nadia Giarbini[3] and Christian Streitberger[3]

[1] *Leopold-Franzens-University, Dept. for Linguistics, Innsbruck, Austria*

[2] *VIBRANT MED-EL, Innsbruck, Austria*

[3] *Franz Tappeiner Hospital, Dept. for Otorhinolaryngology, Merano, Italy*

<ins>Background</ins>: The active implantable middle ear implant Vibrant Soundbridge (VSB) may be used with an audio processor using omnidirectional microphone technology or with one using directional microphone technology. The focus of this research is speech understanding in noise by VSB patients using both types of audio processors.

<ins>Objective</ins>: To compare speech understanding in noise with an audio processor using an omnidirectional microphone (AP 404) with one using a directional microphone (Amadé) in four noise conditions.

<ins>Materials and Methods</ins>: 12 experienced and unilaterally implanted VSB subjects with mixed or conductive hearing loss participated. A prospective, within-subjects design was used. The German sentence test Oldenburger Satztest (OLSA) was administered in the following noise conditions: S_0N_0, S_0N_{VSB} (noise on VSB side), S_0N_{cont} (noise on contralateral side), and S_0N_{180} in a 65 dB SPL (A) speech-shaped noise. The test was administered first with the AP 404 and then repeated after fitting subjects with an audio processor in directional microphone mode. No noise reduction or speech enhancement algorithms were used, and subjects were tested immediately after fitting. The mean SNR was used to calculate a directional advantage (DA) for each noise and processor condition.

<ins>Results</ins>: SNRs with the directional microphone Amadé were lower (better) than with the omnidirectional microphone AP 404 in all noise conditions. The mean SNR significantly decreased in the S_0N_0 condition from 6.6 dB (AP 404) to 5.3 dB SNR (Amadé) (DA= 1.3 dB), in the S_0N_{VSB} condition from 6.3 dB (AP 404) to 3.6 dB SNR

(Amadé) (DA= 2.7 dB), in the S_0N_{cont} condition from 4.7 dB (AP 404) to 1.8 dB SNR (Amadé) (DA= 2.9 dB), and in the S_0N_{180} condition from 5.3 dB (AP 404) to 0.6 dB SNR (Amadé) (DA= 4.7 dB). As expected, the largest DA was found in the S_0N_{180} condition. No DA was expected in the S_0N_0 condition, but was present.

Conclusion: The results suggest significantly improved speech understanding with a directional microphone as compared with an omndirectional one in all noise conditions. However, because lower SNRs were observed even when speech and noise were presented to subjects from the same frontal source, a condition not expected to show an advantage for a directional microphone, it is possible that differences between the processors in not only microphone directivity but also in chip and other technology may have contributed to lower SNRs.

A7

The effect of compression on speech-level perception for hearing-impaired individuals

William M. Whitmer and Michael A. Akeroyd, MRC Institute of Hearing Research (Scottish Section)

Current hearing aid technology uses dynamic compression to compensate for hearing impairment. This ought to affect any aspect of perception based on level cues, such as auditory distance. In a previous study, however, Akeroyd (*JASA*, 127:9-12, 2010) found no difference in distance discrimination for speech sounds with or without hearing amplification for hearing impaired individuals. To examine this (lack of) effect more directly, level discrimination thresholds were measured in the current study using words, sentences and speech-spectrum noises, all with and without hearing amplification. Thirty-eight hearing-impaired listeners (mean age = 67.6 years) participated with their own hearing aids (mean compression ratio at 2000 Hz = 1.7:1). Thresholds were measured using an adaptive-tracking, 2AFC procedure. Stimuli were presented from a loudspeaker facing the listener in a sound-dampened chamber. Eight normal-hearing participants performed the same task for comparison.

In both aided and unaided conditions, it was found that level discrimination for different word and sentence tokens was more difficult than for noises: in unaided conditions, mean ΔL thresholds for words, sentences and noises were 2.4, 2.1 and 1.4 dB; in aided conditions, mean ΔL thresholds were 2.9, 2.2 and 1.3 dB, respectively. The mean level-discrimination threshold for noise was significantly higher for hearing-impaired than normal-hearing listeners (0.8 dB). There were no correlations between hearing-impaired performance and various audiometric and hearing-aid measurements. These results demonstrate that there is a clear difficulty in judging the level differences between words or sentences relative to judging level differences for broadband noises. This difficulty was found for both hearing-impaired and normal-hearing individuals, and had no relation to the amount of hearing-aid compression. The lack of a clear adverse effect of hearing-aid compression on level discrimination is suggested to be due to the low effective compression ratios of currently fitted hearing aids. The poorer level-discrimination thresholds for speech sounds are suggested to be due to the method of equalisation. [This work supported by the MRC and CSO].

A8

Binaural de-reverberation based on interaural coherence

Adam Westermann[1], Jörg M. Buchholz[1], Torsten Dau[1] and Kristoffer F. Jørgensen[2]

[1] *Centre for applied hearing research (CAHR), Department of Electrical Engineering, Technical University of Denmark (DTU), Kgs. Lyngby, Denmark*

[2] *Widex A/S, Lynge, Denmark*

When communicating inside a room, the direct sound is accompanied by multiple reflections originating from the surrounding surfaces. Conceptually, the response characterizing the room is often separated between early reflections (containing the first 50-80 ms of the room response) and a late (diffuse) reverberation. Where early reflections often improve speech intelligibility, the late reverberation typically deteriorates speech intelligibility, listening comfort, as well as localization, which is particularly a problem for hearing impaired listeners. In order to reduce late reverberation in hearing aids, different, mainly monaural, algorithms have been proposed recently. Here, a novel binaural de-reverberation algorithm is proposed, which is inspired by the work presented in Allen et al. [JASA, 1977] and utilizes the concept of interaural coherence (IAC). Based on a study of the IAC in different listening environments a number of significant modifications have been introduced to: (i) allow a binaural signal output that maintains the directional properties of the auditory scene, (ii) limit signal artifacts and (iii) provide more control for adjusting the algorithm to the given acoustic environment. In order to assess the potential benefit of the proposed method a subjective evaluation was performed applying binaural impulse responses with very long reverberation times. Sound quality was measured using a Multiple Stimuli with Hidden Reference and Anchor (MUSHRA) test. The change in both reverberation and spatial characteristics were investigated through a multi-parameter sensory evaluation test. The different algorithm configurations were tested on antonym scales between "*Dry – Reverberant*", "*Focused – Extended*", "*Near – Distant*" and "*Artificial – Natural*". The results were compared to a binaural reference system based on a binaural version of the classic spectral subtraction method presented in Lebart and Boucher [Acta Acustica/Acustica, 2001].

A9

Using middle ear prosthesis for simultaneous cochlear stimulation and pressure relief through the round window

Thomas Weddell, Ian J. Russell and Andrei N. Lukashkin, University of Sussex

The round window membrane (RW) functions as a pressure relief valve in conventional hearing allowing structures of the middle ear to move. Investigations in recent years have shown that middle ear implants can be used to beneficially stimulate the inner ear via the RW leading to significant recovery of hearing thresholds. In certain cases of severe conductive hearing loss, including congenital malformations of the outer and middle ear, RW stimulation is the only option. Isolated clinical uses of this technique have been applied but more thorough theoretical and empirical studies are required to make the outcome of this technique more predictable. Using guinea pigs as test subjects we have investigated physiological effects of RW stimulation using a simulation of active middle ear prosthesis (AMEP) (a cylindrical neodymium iron boron disk magnet) placed upon the RW which can be stimulated by an electromagnetic coil positioned in close proximity to the magnet.

Compound action potentials of the auditory nerve (CAP) and mechanical responses of the RW and ossicles to sinusoidal stimuli were measured by electrode and laser interferometry in guinea pigs in response to acoustic stimulation through a calibrated loudspeaker coupled to the ear canal and to AMEP stimulation (movement of the magnet placed on the RW).

The cochlear neural threshold measured as a threshold for the N1 peak of the CAP did not change after placement of the magnet on the RW and during the entire experiment (up to 3 hours after the placement). The coil voltage - magnet displacement relationship is linear and frequency dependent. Magnet displacement threshold curves demonstrate extremely high sensitivity of the cochlea to the RW stimulation in the nm range. At neural threshold levels and above when the cochlea is driven with an AMEP, ossicular movement was only observed at low frequencies <5 kHz and high stimulation amplitudes. Due to the relatively high impedance of the ossicles as seen from the cochlea and the fact that the magnet did not entirely cover the RW, part of the RW, which was not covered by the magnet, could function as a pressure shunt during the RW stimulation. We propose that the basilar membrane is directly driven via near-field particle displacement generated in the vicinity of the RW by vibrations of the magnet. [This research was supported by the MRC.]

A10

Validity of a Filipino adaptation of the Tinnitus Handicap Inventory

Celina Ann M. Tobias, Erasmo Gonzalo D.V. Llanes, and Charlotte M. Chiong
University of the Philippines-Philippine General Hospital

Tinnitus is a symptom that affects millions worldwide. In the Philippines, a review of audiology request forms at the Ear Unit of the Philippine General Hospital from January to July 2009 revealed that out of 1079 patients that underwent audiometry testing, 348 complained of tinnitus—207 of which as a symptom secondary to another problem, and 141 as a chief complaint.

Although there is a need for tools that can be used to evaluate tinnitus patients in the Philippines, there are currently no concrete services or tools available.

The purpose of this study is to validate a Filipino adaptation of the Tinnitus Handicap Inventory (THI), one of the most commonly translated tinnitus questionnaires, in order to help quantify the effects of tinnitus on the quality of life of Filipino tinnitus patients as well as assist clinicians dealing with Filipino tinnitus patients in justifying their decision for therapy or in making the appropriate referrals.

This study is a cross-sectional psychometric validation of the THI by Newman, et al. The Filipino version of the THI was given to 75 patients, aged 18 to 82 with tinnitus, recruited consecutively at the Ear Unit of the Philippine General Hospital after receiving assessment at the Ear, Nose, and Throat-Out Patient Department. Patients with dizziness, vertigo, and mental retardation were excluded. Puretone audiometry was performed and psychoacoustic characteristics of tinnitus (loudness and pitch) were determined.

The THI-PH showed robust internal consistency reliability for the whole questionnaire and the three subscales. It is a valid and reliable tool that can be used to quantify the effects of tinnitus on quality of life of Filipino patients as well as evaluate the effectiveness of hearing aids, maskers, and other treatments for tinnitus.

A11

Measuring and predicting the quality of nonlinearly distorted music and speech as perceived by hearing-impaired people

Chin-Tuan Tan[1], Brian C. J. Moore[2] and Mario Svirsky[1]

1 *School of Medicine, New York Univ., New York, NY 10016*

2 *Univ. of Cambridge, Cambridge CB2 3EB, United Kingdom*

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 nonlinear distortion, some of which are inherent to certain hearing aid designs, including (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. Stimuli were subjected to frequency-dependent amplification as prescribed by the “Cambridge formula” before presentation via Sennheiser HD580 earphones. These artificial distortions were implemented in both broadband and band-limited conditions (the latter simulate the effect of applying AGC or limiting separately in different frequency bands). In our preliminary results with the broadband conditions, the pattern of the rating was reasonably consistent across subjects, with only two of ten subjects not making consistent ratings. 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. Without any modifications to our previous model for normal hearing listeners, we were able to predict the perceived quality of speech and music by hearing-impaired listeners with good correlations of 0.89 and 0.87 respectively. Currently, we are working on the band-limited conditions and we will also be including recordings of the outputs of mobile telephones and compression hearing aids as cases of “real” nonlinear distortions. [Work supported by Deafness Research Foundation.]

A12

Speech enhancement algorithm for digital hearing aids on the basis of auditory scene analysis

Masahiro Sunohara, Kiyoaki Terada, Takatoshi Okuno and Takashi Iwakura, Rion Co., Ltd.

Most of the hearing-impaired people have troubles communicating in social settings. As far as they are in a quiet environment, their audibilities can be improved in most cases by using the hearing aids. However, it is still hard for them to pick up a sound under noisy environments such as conversation in a crowd, attention to one phrase in a music, etc. This problem could be attributed to a lack of their abilities to segregate complexly-mixed sound. On the other hand, the ability is inherent in normal hearing people and has been summarized as the basis of psychoacoustical knowledge called Auditory Scene Analysis (ASA). According to the theory of ASA, the periodicity related to fundamental frequency (F0) and its harmonic components is one of the most significant cues to segregate any mixed sound. In other words, if the periodicity of speech were enhanced somehow, it could be helpful for hearing-impaired people to communicate each other under noisy environments.

In order to help hearing abilities for hearing-impaired people, a speech enhancement algorithm for digital hearing aids is proposed, which is motivated by a concept of the ASA. The algorithm consists of the F0 extraction system from voiced sound and a flexible time-variant comb filter designed to deliver a natural enhanced sound of the F0 and its harmonics with a low computational cost. In addition, an algorithm of controlling the time-variant response of spectral envelope for the comb filter is proposed based on the theory of Wiener filter. To evaluate the enhancement algorithm, the prototype hearing aids in which these methods are implemented are prepared. The word intelligibility test for 6 hearing impaired subjects with the prototypes is carried out using 20 test words mixed with a multi-talker noise so that signal-to-noise ratio is +10dB. From the results of the evaluation, the word intelligibility score for 5 out of 6 subjects are improved or almost the same. [This work was supported by a grant from Association for Technical Aids in Japan since 2008.]

A13

Objective and subjective sound quality measurements of binaural wireless hearing aids

David Suelzle[1], Iman Elsabah[1], Jon Pietrobon[1], Vijay Parsa[1], Margaret Cheesman[1], Ahmed Sameh Farid[2]
[1] *National Centre for Audiology, University of Western Ontario, London, Canada.*
[2] *Faculty of Medicine, Cairo University*

When fitted bilaterally, modern hearing aids with independently-acting multichannel wide dynamic range compression (WDRC) and adaptive directionality may distort salient sound localization cues and impact the naturalness of sound. Binaural wireless hearing aids are designed to alleviate this issue by synchronizing the functioning of WDRC and adaptive directionality, as well as other advanced signal processing features such as digital noise reduction and feedback cancellation. There is limited evidence, however, supporting the effectiveness of binaural wireless hearing aids. Furthermore, studies benchmarking the performance of different models of binaural wireless hearing aids are currently lacking. Therefore, the objective of this study was to measure and compare the performance of two different binaural wireless hearing aid schemes in terms of sound quality. In particular, the study aimed to record the output of binaural wireless hearing aids in response to speech stimuli played in noisy and reverberant laboratory environments, and evaluate the quality of recordings through objective sound quality models and a behavioural rating task by normal and hearing impaired listeners.

The hearing aid recordings were obtained in the anechoic and reverberation chambers (RT60 ≈ 667 ms) at the National Centre for Audiology. Binaural behind-the-ear (BTE) wireless hearing aids were programmed to fit the targets prescribed by DSL 5.0 for each hearing impaired listener, and placed on a Head and Torso Simulator (HATS). The HATS was positioned at the centre of a speaker array, and speech samples were played from 0^o azimuth, while multi-talker babble or traffic noise were played from speakers at 90^o, 180^o, and 270^o azimuth simultaneously. Recordings were obtained for quiet, 0 dB, and 5 dB SNRs with all combinations of omni/adaptive directional microphone configuration and wireless communication on/off. The hearing aid recordings were later played back to the participants and their ratings of speech quality were obtained for each condition. In addition, the recordings were processed through different sound quality prediction schemes based on coherence-based SII, loudness pattern distortion, and modulation spectrum analysis. Preliminary results of this project-in-progress confirm a number of expected results including a preference for the anechoic environment over the reverberant environment, a preference for the adaptive directional mode in the anechoic environment with surrounding noise and a preference for the omni directional mode in the absence of noise. However, no clear preference for binaural wireless communication was observed. Clinical implications of these results as well as correlations with objective metrics will be discussed.

A14

Speech-in-noise testing: The importance of random fluctuations in steady noise

M. A. Stone, B. C. J. Moore, C. Füllgrabe, and R. C. Mackinnon, Department of Experimental Psychology, University of Cambridge, Downing Street, Cambridge CB2 3EB.

Speech-in-noise testing is a mainstay of objective measures of benefit of hearing prostheses. Many studies have shown that, for normal-hearing listeners, speech identification is better when the background is fluctuating than when it is steady, an effect called masking release (MR), which is thought to depend on "listening in the dips". MR is reduced or absent for people with hearing impairments or cochlear implants. An implicit assumption of such studies is that the inherent fluctuations in a steady background noise have little effect. Here, we show that this is far from true, using signals passed through a tone vocoder to promote speech identification based on envelope cues.

In one type of processing, resembling conventional methods, target sentences (Tgt) were mixed with either steady random noise (Nse), the same noise sinusoidally amplitude modulated at 8 Hz (SAM), or an interfering speaker (Jam), all at –7 dB Tgt-to-background ratio. Background sounds were matched in long-term spectrum. The mixture was then tone vocoded using 12, 19 or 30 channels.

A second type of processing, simulated the "ideal" case in which the Nse had no random fluctuations, and the SAM noise had only the imposed 8-Hz fluctuations. The Nse without random amplitude fluctuations was simulated using a constant-level envelope in each vocoder channel. The SAM without random amplitude fluctuations was simulated by modulating the constant-level envelope in each channel. For completeness, the Jam channel envelopes were extracted for the Jam alone. In all cases, the background channel envelopes were added to the separately-extracted Tgt channel envelopes before being used to modulate the channel carriers of the vocoder. The mixing of the Tgt and background therefore occurred in the envelope domain (ENV), in contrast to the mixing in the signal domain used for conventional processing (SIG).

For both methods, intelligibility was measured using 18 young, normal-hearing listeners. For the Jam background, performance was similar (and poor) for the two methods. For the Nse background, performance was very poor (near 5%) for method SIG, but was near 90% for method ENV, indicating that the inherent fluctuations in steady noise have a very large effect. For the SAM background, performance was markedly poorer (20-40%, depending on channel number) than for method SIG (40-63%, depending on channel number). We conclude that steady Gaussian noise introduces low-rate modulations into the channel envelopes that have a large deleterious effect on speech intelligibility. "Steady" noise is therefore a misnomer.

A15

Perception of one's own voice with hearing aids.

Stefan Stenfelt, Linköping University, Sweden

The own voice is perceived by sound transmitted via an airborne path and a bone conduction path to the ear. The air conduction path is rather straightforward with sound exiting the mouth/nose and entering the ear canal. Bone conduction transmission from own voice production is more complex involving transmission pathways to the ear canal, middle, and inner ear.

Using previous knowledge of the air and bone conduction pathways during vocalization with model predictions of the occlusion effect and hearing aid gain, a model of the perception change of one's own voice using hearing aids was devised. This model was verified with measurements on hearing aid wearers from the clinic showing the relation between type of hearing aid (open fitting, mould with ventilation, or in-the-ear design), amplification, and perceived own voice change.

Measurements are done in a clinical sample showing how a hearing aid fitting influences the perception of one's own voice. The same subjects are self-rating their perception of own voice showing a correlation between amplification, hearing aid type, and own voice perception. Also, there is a significant loudness reduction of own voice production when using hearing aids indicating that hearing aids relieve speech production.

A16

Perception of various acoustical dimensions across acoustical situations and hearing instruments

Andre Steinbuss[1], Matthias Latzel[1], Joerg Haubold[2], Michael Schulte[3] and Matthias Vormann[3]

[1] *Siemens Audiological Engineering Group, Erlangen, Germany*

[2] *ciAD – competence in AudioDesign, Dortmund, Germany*

[3] *Hoerzentrum Oldenburg, Oldenburg, Germany*

The functionality of any hearing aid is directed towards satisfying two main listening aspects of the hearing impaired; on the one hand, the aid should allow for a speech intelligibility benefit in both the quiet and noisy situations. On the other hand, the sound of the hearing aid should be assessed as comfortable by the user across all listening situations. In order to achieve the two goals the hearing aids, as well as fitting rationales, are subjected to evaluations on the part of the scientific researchers as well as the commercial stakeholders.

In the field studies, the questionnaires are firmly established as a means of assessing the hearing aid wearer's satisfaction and preferences. However, even an in-depth analysis of the field study data does not always allow for a decision as to the listening dimension which most influenced, or which dominated, the subject's assessment.

As an alternative to this approach and with a goal of determining the impact of various listening dimensions on the hearing aid wearer's satisfaction in mind, the present study was conducted in the laboratory conditions with hearing impaired subjects using the paired comparison method.

The study evaluated five hearing aids, each from a different manufacturer. Each hearing aid's output was recorded individually using probe tubes. Recordings were taken for the three acoustical situations speech in quiet, speech in noise and music. The subjects were asked to assess the recordings with regard to different acoustical dimensions during paired comparisons. The acoustical dimensions speech intelligibility, loudness, naturalness, sharpness and overall satisfaction were investigated in a 4-scaled forced choice manner.

The presentation will show the results of the paired comparisons. Based on the correlations found between different listening dimensions, tentative indications as to their importance will be discussed.

A17

Assessing envelope distortion in clinically-fit hearing aids

Pamela Souza[1], Eric Hoover[1], Frederick Gallun[2] and Marc Brennan[3]

1 *Northwestern University*

2 *National Center for Rehabilitative Audit*

3 *University of Washington*

The speech envelope, consisting of modulations below 50 Hz, provides important cues to speech recognition. The Envelope Difference Index (EDI: Fortune et al., 1994) quantifies distortion to the temporal envelope of a speech signal. The EDI can characterize the effect of signal processing strategies such as wide-dynamic range compression (WDRC) amplification on the speech envelope. In our previous work, EDI values greater than 0.2 were associated with reduced speech recognition, particularly for listeners with reduced ability to detect spectral differences. This indicates that clinically fit hearing aids with an

EDI greater than 0.2 could adversely impact speech recognition for the user. However, it is not known under what circumstances clinically fit hearing aids will have an EDI greater than 0.2. In the present study, the EDI was measured for speech amplified by wearable hearing aids. All recordings were done on a KEMAR manikin. Simulated audiograms ranging from flat to sloping and from mild to severe were assessed. Because the majority of clinicians use the manufacturer's default program settings, all hearing aids were programmed with frequency-gain parameters set with these default settings. The hearing aids were fit utilizing clinically-appropriate real ear targets. We also assessed the effect of adjustments that might be made in response to patient complaints, such as reducing maximum output. Test signals were vowel-consonant-vowel nonsense syllables and sentences. On average, EDI values were within the acceptable range (< 0.2), but this was not the case for all stimuli with all combinations of parameters. EDIs exceeded the acceptable range for severe hearing loss and when the maximum output of the hearing aid was reduced. There was an effect of consonant, with high-frequency fricative consonants (particularly "s" and "sh") demonstrating the most envelope change. The range of EDIs also varied with the default compression time constant of the product. These results can guide clinicians as to when unacceptable envelope distortion might be suspected. It is also suggested that products (or custom parameter settings) which utilize longer time constants may be most appropriate for patients with reduced spectral discrimination. [Work was supported by the National Institutes of Health and the Veterans Administration Rehabilitation Research and Development Service.]

A18

The high performance core with low power consumption for hearing processor

Jongkeun Song and Dongwook Kim, Samsung Advanced Institute of Technology

It has long been developed hearing processor showing high performance but ultra low power consumption to support various kinds of hearing algorithm such as noise reduction, feedback cancellation, sound directivity control and binaural hearing, etc. Several companies recently tried to design multi-core architecture composed of a conventional low performance (<50MIPS) master and slave core. However, in order to implement high performance hearing processor for supporting various hearing aid algorithms, it needs to develop new core with high performance over 100MOPS and ultra low power consumption below 1mW.

The purpose of this study is to verify newly developed core performance and power consumption to integrate hearing processor. A standard cell library of 65nm CMOS low power technology with high threshold voltage (HVT) process was used to reduce the power consumption and the leakage. The simulated system was implemented with the core, a bus controller, four banks of SRAM (64kB x4), and an interrupt controller. The total gate count of the core was 395,000. To estimate core performance, the 16 and 64-tap FIR filter and FFT algorithm was used. The test result shows that total power consumption of FIR filter was 1025uW (16-tap, N=64, 626-cycle) and 1233uW (64-tap, N=256, 7662-cycle) and performance was calculated with 116MOPS/mW. In the case of FFT algorithm, total power was simulated with 1076uW at 256-point FFT of radix-2(1460cycle) and 1156uW at 256-point FFT of radix-4(1140cycle) and performance was calculated with 110MOPS/mW. These results suggest that more complex and larger number of algorithms can be executed in this core with ultra low power consumption. For more powerful performance, we will design dual- and quad-core hearing processor using this low power core. [This work was supported by grant No.10031731 from the Strategic Technology Development Program of Ministry of Knowledge Economy]

A19

Use of the Extended Speech Intelligibility Index (ESII) and the Hearing In Noise Test (HINT) to quantify functional hearing ability

Sigfrid D. Soli[1], Shelley Montgomery[2], Larry Meyers[3], Laurel Alvarez[2], Stacey Fuller[3], Kelly Hunley[3]

[1] *House Ear Institute, Los Angeles, CA*
[2] *Corrections Standards Authority, Sacramento, CA*
[3] *Sacramento State University, Sacramento, CA*

The Extended Speech Intelligibility Index (ESII) is used to predict the speech reception thresholds (SRT) in real-world fluctuating noise environments (Rhebergen et al., 2008). For individuals with normal hearing the ESII at the SRT is typically about 0.34 (e.g., Festen and Houtgast, 2008). Likewise, the ESII for a HINT threshold of -2.6 dB S/N—the norm for the Noise Front test condition (Soli and Wong, 2008), which is most similar to the conditions under which the ESII is calculated—is 0.35. Thresholds above the norm indicate that a larger ESII is needed to achieve the same level of intelligibility. This relationship between the ESII and HINT threshold elevation can be used to quantify the impact of threshold elevation on functional hearing ability, especially speech communication, in real-world noise environments.

This presentation describes research to establish and validate hearing screening criteria for Correctional Officers based on HINT thresholds and ESII analyses of their workplace noise environments. Calibrated recordings of workplace noise environments where hearing-critical tasks are routinely performed were obtained and analyzed in small time frames (Rhebergen and Versfeld, 2005) to obtain 1/3 octave band spectrum levels for use in combination with the speech spectrum levels specified in ANSI S3.5-1997 to calculate the SII in each small time frame. Assuming that typical brief two-way speech communication takes place over a 4-sec interval, the SII values within each 4-sec interval were averaged to produce the ESII for each interval. The frequency distribution of these ESII values for each workplace noise environment was used to determine the proportion of intervals for which the ESII exceeded a criterion value for effective speech communication, assuming a HINT threshold at the norm. This proportion defines the likelihood of effective communication in each particular environment. Likewise, the weighted combination of proportions for the set of workplace environments comprising the typical day of a Correctional Officer defines the overall likelihood of effective communication throughout such a day.

Elevation of a HINT threshold above the norm increases the ESII criterion value and thus reduces the likelihood of effective communication. For example, a 1 dB threshold increase corresponds to an increase in the ESII of approximately 0.03. Using this relationship, it is possible to make quantitative estimates of the impact of threshold elevation on the likelihood of effective communication during hearing-critical activities on the job and, in so doing, to establish and validate hearing screening criteria for specific jobs and workplaces. The results of this effort will be reported, with an emphasis on the broader implications for quantifying the functional benefits of hearing aids and other treatments for hearing impairment.

A20

Study on self assessment of hearing loss in mobile phones

Junil Sohn[1], Dongwook Kim[1], YunSeo Ku[1], Kyungwon Lee[2], Junghak Lee[2]
[1] *Samsung Advanced Institute of Technology*
[2] *Hallym University of Graduate Studies*

A lot of people have mobile phones and needs for better sound quality in mobile phones has increased especially in various noisy environments. The improved sound quality gives elderly people great benefit like enhanced speech intelligibility. For enhancing speech intelligibility, self assessment of user's hearing loss is a very essential process, but it is difficult that the traditional test with high fidelity like using various pure tones is applied in mobile phones. Because users have to spend a long time to respond to stimulation with various frequencies and amplitude repetition, mobile phone is not good system for the hearing test with high fidelity, and users do not have knowledge of audiology.

In this study, we proposed self assessment of hearing loss that can simply test user's hearing loss in mobile phones. Four phonemes, /a/, /i/, /sh/ and /s/ were used as test sounds and each phoneme has different sound levels from 20dB SPL to 70dB SPL with interval of 5 dB SPL. It seems to be reasonable that these phonemes are used as test sounds in mobile phones, because these phonemes are complex signals and have much more information than pure tone. Therefore we can get information of hearing loss from user's response to a few numbers of phonemes.

In order to show the performance of the proposed method, we carried out two stage test. In the first stage, we tested 24 normal ears and 32 ears with mild or moderate hearing loss using audiometer in chamber. Comparison with audiogram showed that average difference was 8.7dB.

In the second stage, we built the proposed self assessment of hearing loss on a mobile phone, Samsung SPH-M8400 and tested on group with mild hearing loss. In this test, we could know that users can check their hearing loss simply but high reliably. [This work was supported by grant No.10031731 from the Strategic Technology Development Program of Ministry of Knowledge Economy]

A21

Pinna cue-preserving hearing-aid fittings: Who might be a candidate?

Niels Søgaard Jensen[1], Tobias Neher[1], Søren Laugesen[1], Louise Kragelund[1] and Thomas Behrens[2]

[1] *Eriksholm, Oticon Research Centre*

[2] *Oticon A/S*

A field test investigating the benefit of preserving pinna cues in hearing-aid fittings has been conducted, the results of which are to be presented at this conference by Neher et al. In a lab study preceding the field test, the aim was to identify a group of hearing-impaired people who expectedly would be able to utilise pinna cues, and thus would be suitable candidates for participation in the field test. This candidature-study also investigated whether the ability to utilise pinna cues as measured in a complex listening test can be predicted by means of more basic measures of auditory and cognitive functioning.

The study included a group of 30 hearing-impaired test subjects (out of which 20 were selected as candidates for pinna cue-preserving hearing aids and included in the subsequent field test). The test subjects' ability to utilise pinna cues was assessed in two complex listening tests conducted with open ears in free field: (1) spatial release from speech-on-speech masking along the front-back and left-right dimension, and (2) a localisation test with emphasis on front-back discrimination performance. The basic auditory measures included two headphone-listening tests: (1) a test of the ability to detect changes in monaural spectral ripple patterns, and (2) estimation of the effective bandwidth of binaural temporal fine structure processing. In all listening tests, sufficient audibility of the various stimuli was ensured, as discussed in a companion poster by Laugesen et al. The cognitive measures included in the study were: (1) the reading span test, and (2) the test of everyday attention.

This poster will present and discuss data from the candidature-study. Focus will be on those data – and their inter-correlations – which are most relevant for the discussion of possible benefits from pinna cues, i.e. the data relating to the front-back dimension in the complex listening tests. The discussion will target the question of how candidacy for a pinna cue-preserving hearing-aid fitting can be assessed.

A22

Uncertainty about the location of a target: Effects of hearing loss and linear vs. nonlinear amplification on auditory spatial attention

Gurjit Singh[1,2], M. Kathy Pichora-Fuller[1,2], Thomas Behrens[3] and Tobias Neher[3]

[1] *Department of Psychology, University of Toronto at Mississauga, Mississauga, Canada*

[2] *Toronto Rehabilitation Institute, Toronto, Canada*

[3] *Eriksholm Research Center, Oticon A/S, Snekkersten, Denmark*

To understand better how hearing loss and different hearing aid fittings affect spatial listening abilities in complex multi-talker situations, we investigated auditory spatial attention in 8 older adults with normal audiometric thresholds below 4 kHz and 8 older adults with bilateral sloping sensorineural hearing losses, ranging from moderate to severe in the higher frequencies. In a condition with real spatial separation, a target sentence from the CRM corpus was presented from one spatial location and competing sentences from two different locations (0°, 45°, 90° azimuth), with pre-trial cues specifying the target's identity and location. In a condition with simulated spatial separation, corresponding perceived spatial locations of the target and competitors were achieved through exploitation of the precedence effect. Seven different probability specifications indicated the likelihood of the target being presented at the three locations (100-0-0, 80-

0-20, 60-0-40, 0-0-100, 20-0-80, 40-0-60, and 50-0-50). All listeners completed 10 hours of baseline testing. Using a within-subject, within-device crossover design with 8-week acclimatization periods, listeners with hearing loss were then fitted with hearing aids providing linear and nonlinear amplification. At the conclusion of each acclimatization period, listeners completed an additional 10 hours of testing. As expected, hearing-impaired listeners performed more poorly than normal-hearing listeners across all listening conditions, but especially under test conditions in which the location of the target was less certain. Furthermore, in conditions where targets were expected to be presented from 0°, but were presented from 90°, providing either linear or nonlinear amplification for the hearing-impaired listeners resulted in significant improvements in performance. Finally, individual differences in performance were observed for the hearing-impaired listeners, with some individuals performing better with linear amplification and other individuals performing better with nonlinear amplification. Implications for clinical practice, hearing aid development, and future research will be discussed.

A23

A nonlinear time-domain cochlear model to explore the effects of cochlear signal processing on speech

Roger Serwy, Andrea Trevino, Jont Allen and Bob Wickesberg
University of Illinois at Urbana-Champaign

A nonlinear, time-domain cochlear model has been implemented using wave digital filter (WDF) methods. The model includes Allen's resonant tectorial membrane (RTM) as well as models for the inner and outer hair cell depolarization via cilia displacement. Simulated neural tuning curves and basilar membrane motion are compared to published data for the chinchilla cochlea. Auditory nerve responses to speech waveforms, taken from the chinchilla, are compared to simulation results as well.

Cochlear nonlinearities play a strong role in speech perception. For example, the roll-over effect describes the reduction in speech perception scores as presentation intensity increases beyond 65 dB-SPL. Such effects are usually attributed to the upward spread of masking (USM) or neural saturation effects. Other masking effects, such as the forward spread of masking (FSM) can mask speech immediately after a high-level burst. These cochlear nonlinearities affect the neural encoding of acoustic cues found in speech waveforms.

In hearing-impaired (HI) ears, outer hair cell damage affects nonlinear cochlear signal processing in an adverse way, namely reducing sensitivity to stimuli. By using a physically-based nonlinear cochlear model, the neural differences between HI ears and normal-hearing (NH) ears can be simulated, making it possible to design better hearing aid algorithms to best undo the effects of cochlear damage.

A24

Behavioral measures of monaural temporal fine structure processing

Sébastien Santurette and Torsten Dau, Centre for Applied Hearing Research, Department of Electrical Engineering, Technical University of Denmark

Deficits in temporal fine structure (TFS) processing found in hearing-impaired listeners have been shown to correlate poorly to audibility and frequency selectivity, despite adverse effects on speech perception in noise. This underlines the need for an independent measure of TFS processing when characterizing hearing impairment. Estimating the acuity of monaural TFS processing in humans however remains a challenge. One suggested measure is based on the ability of listeners to detect a pitch shift between harmonic (H) and inharmonic (I) complex tones with unresolved components (e.g. Moore et al., JASA 125:3412-3422, 2009). However, spectral cues arising from detectable excitation pattern shifts or audible combination tones might supplement TFS cues in this H/I-discrimination task.

The present study further assessed the importance of the role of TFS, in contrast to that of temporal envelope and spectral resolution, for the low pitch evoked by high-frequency complex tones. The aim was to estimate the efficiency of monaural TFS cues as a function of the stimulus center frequency Fc and its ratio N to the stimulus envelope repetition rate. A pitch-matching paradigm was used, such that changes in spectral indices were not useable as a cue. The low pitch of broadband pulse-trains was matched to that of inharmonic transposed tones ($3 \leq Fc \leq 7$ kHz, N=[11.5,14.5]). Resolvability of the stimulus components was assessed, and the contribution of TFS information to individual pitch-matching results was estimated and compared to performance of the same subjects in an H/I-discrimination experiment ($2.2 \leq Fc \leq 9$ kHz, N=[11,13,15]) similar to that of Moore et al. (2009).

Pitch matches revealed an ambiguous low pitch related to the timing between TFS peaks near adjacent envelope maxima, up to Fc=7 kHz for N= 11.5 and Fc=5 kHz for N=14.5. Pitch salience decreased as Fc or N increased, but pitch matches never relied on the envelope repetition rate. Moreover, the results from the component-resolvability experiment indicated an inability of subjects to hear out the lowest frequency components of the stimuli. This strongly suggests that the monaural representation of TFS persists at high frequencies and prevails over envelope or spectral cues for perception of the low pitch of high-frequency complex tones. Individual performance in the H/I-discrimination experiment is therefore expected to show a similar dependency on Fc and N as the corresponding individual pitch matching results. If so, such methods may be useful to estimate the upper frequency limit for monaural TFS processing in individual subjects.

A25

Cues governing listeners' acceptable speech and noise levels

Karrie Recker, Martin McKinney and Brent Edwards
Starkey Laboratories, Inc.

The Acceptable Noise Level (ANL) test is a measure of the amount of background noise that a listener is willing to "put up with" while listening to running speech. The ANL test is significant because it is the only measure demonstrated to be correlated with an individual's success with hearing aids. Individuals who are very tolerant of background noise are generally successful hearing-aid wearers, whereas individuals who are very intolerant of background noise are generally unsuccessful hearing-aid wearers. In order to understand what makes someone successful with hearing aids, we wanted to determine what perceptual mechanism underlies an individual's ANL score. To test this, we performed a series of experiments.

Twenty normal-hearing and twelve hearing-impaired individuals participated in this study. For each individual, we measured ANLs at four fixed-speech levels (50, 63, 75 and 88 dBA). Additionally, we modified the task and had listeners adjust the level of the speech to the minimum, preferred and maximum levels at which they were willing to listen in four different background noise levels (multi-talker babble at 50, 60, 70 and 80 dBA). This was of interest because the literature shows conflicting trends regarding the signal-to-noise ratios (SNRs) at which people are willing to listen. As presentation levels increase, ANLs have been shown to increase (Franklin et al., 2006; Tampas and Harkrider, 2006; Freyaldenhoven et al, 2007), whereas preferred SNRs have been shown to decrease (Munson & Karlin, 1962).

For each of these four measures, we wanted to determine whether either speech intelligibility or loudness was guiding individuals' decisions regarding the SNRs that they were willing to accept. Speech intelligibility was examined by comparing participants' results with the levels that would be expected based on Articulation Index Theory. Loudness was examined by comparing listeners' results to data generated from a loudness model by Chalupper and Fastl (2002). Additionally, listeners performed several loudness-matching tasks in which they matched the level of a target (speech or multi-talker babble) in the presence of a masker (speech or multi-talker babble).

The order of all test conditions was randomized. All test conditions were performed at least twice and averaged. Results suggest that different listeners are using different cues for these tasks. Detailed results, and their clinical implications, will be discussed.

A26

Development of an invisible and removable intra-oral tissue conduction microphone for hearing device applications

T.L. Proulx, Sonitus Medical, Inc.; Gerald R. Popelka, PhD, Stanford University

Tissue contact vibration sensors (contact microphones) have been widely employed in electronic stethoscopes for sensing sounds originating from within the body, such as the heart beat, blood flow or respiration. These transducers are placed in contact with the external surface of skin or soft tissue and generate an electrical signal in response to vibrations propagating through the tissue induced by biophysical processes. Other electronic stethoscopes in common use include those used to detect user speech, such as throat-mounted or head-mounted microphones. These sensors detect vibrations induced by resonances of the larynx and other portions of the vocal tract that propagate through the hard tissue of the larynx or bone of the skull and through the surrounding tissue.

A related application in this area involves a fully implantable hearing aid, where the microphone is installed subcutaneously just above and behind the ear or within the bony wall of the auditory canal. In contrast with sensors designed to detect user speech, the implanted hearing aid microphone detects ambient environmental sounds, and responds to vibration of the skin surface as opposed to internal vibrations propagating through tissue. While an implanted microphone offers numerous advantages compared to an externally worn microphone, the surgical procedure required to install or remove it is a significant drawback. There is interest in a non-surgically installed (i.e. removable), but invisible microphone that offers acceptable sensitivity and fidelity for implantable hearing aids or other concealed hearing device applications.

We report on development of a small intra-oral tissue conduction microphone system capable of detecting external ambient sound that couples to and propagates through the soft tissue of the face and cheek. The microphone is integrated into a removable and unobservable dental appliance that attaches to the upper molars. Prototype intra-oral microphones using polyvinylidene fluoride (PVDF) piezoelectric film were measured in situ with sound transfer functions of 20mV/Pa or higher from 250 Hz to 8 kHz. A conventional single syllable word recognition test (NU6-1A, 50 items) was broadcast at 70dB SPL and recorded by the sensor installed in a volunteer's closed mouth. Mean word recognition accuracy of the recordings was 95.3% with a range of 94% – 98% for 6 normal hearing listeners. Microphone design, modeling and testing, and further developments of this work in progress are reviewed.

A27

Temporal fine structure deficits in perceptual organization

Niels Henrik Pontoppidan, Research Centre Eriksholm, Oticon A/S
Marianna Vatti, Centre for Applied Hearing Research, Technical University of Denmark

Recent experiments show that hearing-impaired listeners have limited access to temporal fine structure[1] (TFS) and indicate that normal-hearing listeners benefit from TFS in difficult situations[2,3]. In the TFS1 test[4], most hearing impaired cannot perceive the rather large difference between the harmonic and the shifted tones; this is despite the fact that all involved parts are audible and that differences are obvious to normal-hearing listeners. The observation raises the question: If hearing-impaired cannot perceive the differences in the TFS1 test, what else can't they perceive? The *Role of Micromodulations*[5] demonstrates how the onset of relatively small amount of frequency modulation (the micromodulation) alters the perceptual organization of the sound, and thus overrides the strong *old + new heuristic*[5]. If hearing-impaired cannot perceive and group according to the micromodulation, it strengthens the claim that TFS is important for speaker recognition and gap listening. In the present study, we will explore the role of micromodulation for sound perception and auditory scene analysis. One initial task is identifying the limits of frequency modulation rate and frequency modulation excursion perceived as natural vibrato in synthetic vowels for normal-hearing and hearing-impaired listeners. Data from this investigation will be presented. The next task will address the link between the micromodulation and grouping directly, and the outline of that experiment and preliminary results will be presented as well.

References:

1. *Hopkins, K. and Moore, B.C.J. Moderate cochlear hearing loss leads to a reduced ability to use temporal fine structure information. JASA 122(2):1055-1068. 2007.*
2. *Hopkins, K., Moore, B.C.J., and Stone, M.A. Effects of moderate cochlear hearing loss on the ability to benefit from temporal fine structure information in speech. JASA 123(2):1140-1153. 2008.*
3. *Lunner, T., Hietkamp, R.K., Andersen, M.R., Hopkins, K., and Moore, B.C.J. Effect of speech material on the benefit of temporal fine structure information in speech for normal-hearing and hearing-impaired subjects. submitted to JASA 2010.*
4. *Moore, B. and Sek, A. Development of a fast method for determining sensitivity to temporal fine structure. International Journal of Audiology 48(4):161-171. 2009.*
5. *Bregman, A.S. and Ahad, P.A. Demonstrations of Auditory Scene Analysis: The Perceptual Organization of Sound. www.psych.mcgill.ca/labs/auditory/bregmancd.html. 1996. MIT Press.*
6. *Bregman, A.S. Auditory Scene Analysis (MIT Press, Cambridge, Massachusetts 1990).*

A28

The effect of probe microphone system on the NAL-NL1 prescriptive gain provided to patients

Erin M. Picou, Todd A. Ricketts and H. Gus Mueller
Vanderbilt University

Clinician's commonly use prescriptive real ear targets as a starting point for assigning hearing aid gain. Probe-microphone equipment systems facilitate this process by generating and displaying prescriptive targets, in addition to the hearing aid gain (or output) values. While research conducted in the 1980s suggested that prescriptive targets were consistent across several probe-microphone systems that were available at that time (e.g., Humes, Hipskind, & Block 1988), both hearing aid and probe-microphone technologies have changed substantially in the last 20 years. With the changes in technology come increases in options, and not all probe-microphone systems may implement NAL-NL similarly. Sources of variability include: accounting for number of compression channels in the hearing aid, conversion factors, level and shape of input signal, and signal analysis bandwidth. For example, it has recently been shown that specific probe microphone systems differ in the spectral shape of their test signals (Keidser et al, 2010). With the new technologies and potential for increased variability, it has become essential to re-examine the consistency and reliability of current probe-microphone systems. Thus, the purpose of this study was to investigate the consistency and reliability of the fitting provided by several popular, commercially available probe-microphone systems, specifically their implementation of NAL-NL. Preliminary data from two participants and three probe-microphone systems has shown that patients can receive significantly different gain depending on the choice of system when fitted exactly to "NAL-NL targets" (Ricketts & Mueller, 2009). To confirm these findings, and further explore the possible variables that led to inter-system variability, 18 participants and four commercially available probe-microphone systems were tested using a round-robin approach. The test- re-test reliability and inter-system deviation from prescriptive targets was examined for hearing aids without adjusting hearing aid settings. Measurements included real ear aided response, real ear unaided response, and real ear aided gain. Real ear insertion gain was also calculated. Results will be discussed in terms of the system factors that appear to lead to differences in fit between systems and how these differences may interact with specific hearing aid factors (e.g., amplitude compression, occluding or non-occluding fitting).

A29

Technology and method of rapid self-hearing test with mobile device

Gyuseok Park, Hongsub An and Sangmin Lee, Department of Electronic Engineering, Inha University, Korea

To be an aging society, many people wanted to save their characteristics of hearing in the mobile phone or audio equipments, and they needed a sound system for their preference. A simple test of own characteristics of hearing in personal device should be different from complete medical test in hospital. First, taking a measurement of hearing could be as fast as possible. Second, the tester and subject is the same person. In regard to those two points, we suggested an algorithm to test the characteristic of hearing and developed a program. We tested the hearing characteristic of subject in a frequency band, and then we use the method that the next frequency was determined based on a result for rapid self-test. Because the subject is the tester, we designed a GUI for double blind. During the test, we estimated the level of first sound in each frequency. In order to estimate the hearing levels, it was used an average and extended slope line of hearing levels in each frequency. Hearing level of subject was tested in 1000Hz and 4000Hz first, and then we decided to test or skip other frequencies. We estimated hearing level by an average which was between 250Hz and 1000Hz or 1000Hz and 4000Hz. If the hearing levels in each frequency were similar or existed in regular range, they were not required to test all frequencies between 1000Hz and 4000Hz, and move to test in other frequency. In case of most people who have difficulty in hearing, the loss of hearing level was shown a constant slope between frequency bands on hearing loss graph. We calculated an extension line of other frequency levels, and we used it in 8000Hz. A pure tone was generated by estimated hearing level at first in each frequency. This method was evaluated how much time and how many steps to finish the test. The test using this method was able to save the whole time as reducing steps to determine the first sound in each frequency. In addition, correlation was high between result by this method and result by the audiometry. [This work was supported by grant No. 10031764 from the Strategic Technology Development Program of Ministry of Knowledge Economy]

A30

Computer models of normal and impaired hearing

M. R. Panda, C. M. Tan, W. Lecluyse and Ray Meddis, University of Essex, UK

This project aims to develop computer models of normal and impaired hearing. These models are computer programs based on published physiological data. This project uses an updated version of a previously published computer model by Meddis (2006) adapted to produce behavioural responses like human data.

Absolute thresholds, psychophysical tuning curves (PTCs), and temporal masking curves (TMCs) of two normal hearing individuals were first simulated using the model. Various physiological parameters of the normal hearing model were varied and their effects on the psychophysical measurements were observed. These results were used as a basis for hypothesizing the consequences of different kinds of cochlear pathology.

Absolute threshold, PTCs and TMCs of a number of hearing-impaired persons were also collected. The causes of the hearing impairments of these persons were hypothesized. One of the normal hearing models was adjusted accordingly for five profiles. Two of these profiles are presented in the poster. The first profile indicated reduced outer hair cell function. The corresponding physiological parameter in the normal hearing model was adjusted to simulate this hypothesis. The second impaired hearing profile showed hearing pattern suggestive of a strial presbyacusis. The endocochlear potential of the normal hearing model was reduced to simulate this profile. These changes to parameters were consistent with known physiology. In both cases, single parameter change in the normal hearing model was able to simulate the hearing- impaired data. In future, it may be possible to use these computer simulations to optimise hearing aid design for an individual client. [This project is supported by the DRF, UK.]

Reference:

Meddis, R. (2006). "Auditory-nerve first-spike latency and auditory absolute threshold: a computer model," J Acoust Soc Am 119, 406-417.

A31

Variability in bone conduction force output due to intersubject differences in mechanical impedance of the skin

Per Östli, Bo Håkansson, Herman Lundgren, Sabine Reinfeldt and Hamidreza Taghavi

Department of Signals and Systems, Chalmers University of Technology

Background: Bone conduction audiometry is frequently used as a diagnostic method to determine the degree of sensorineural hearing loss. Bone conduction audiometry is also an important tool to determine whether a patient is a suitable candidate for BAHA, or not. As bone conductor, the Radio Ear B71 transducer is commonly used and calibrated with a Brüel & Kjær artificial mastoid. At the hearing threshold, the voltage to the transducer is determined by the dial value of the audiometer. The equivalent force threshold level calibrated on an artificial mastoid. This is only an indirect measurement of the real force at the skin and the real force at the skin may differ between subjects even if the voltage to the transducer is the same.

The aim of this study is to estimate the variability in force levels arising from intersubject differences in the mechanical impedance of the skin.

Method: A simple, well known and earlier used model of the transducer has been combined with real measured mechanical impedance at the skin of 30 subjects. The measurements of the mechanical impedance were performed by Cortés 2002, at the position behind the ear, normally used for bone conduction hearing threshold measurements.

Results: The intersubject standard deviation of the force levels at the skin, only related to the variability in mechanical impedance of skin, is generally ranging from 0-4 dB in the interesting frequency range. The difference in mechanical impedance between artificial mastoid and mean of the subjects results in a difference in terms of force of 0-10 dB. The deviation between the mean mechanical impedance of the skin and the artificial mastoid is probably not a diagnostic issue, since the influence is included in the interpretation of normal hearing. In a "magic" frequency range around 1150 Hz it was found that the standard deviation of the force is almost zero. This is due to that the mechanical output impedance of the B71 is low compare to the skin impedance. In general by decreasing the output impedance of the transducer the variability in force levels arising from intersubject differences will decrease.

A32

The Adaptive Logatom Test – sensitive measure for mild hearing losses

Myriel Nyffeler[1], Michael Boretzki[1], Andrea Kegel[2] and Kathrin Meisenbacher[3]

1 *Phonak AG, Switzerland*
2 *Universitätsspital Zürich, Switzerland*
3 *FH Oldenburg, Germany*

Nonlinear frequency compression (NLFC) is a proven technique for improving the ability of people with hearing impairment to detect and recognize high-frequency sounds. As difficulty perceiving such sounds is one of the most common characteristics of hearing loss, the practical success of nonlinear frequency compression is a significant advance in the field of hearing instruments.

The voiceless /s/ is the consonant with the highest acoustic energy in Indo-European languages. For men, it is between 4 and 6 kHz; for women, it is between 6 and 10 kHz. While the /s/ for some men is still within the transmission range of conventional hearing instruments, often the female /s/ can not be made sufficiently audible with conventional amplification, especially when the voice is soft or distant. These facts led to the focus of our study on the benefits of NLFC for mild hearing loss for identifying the /s/ sounds of a female speaker. A test of /s/ identification is not possible with conventional sentence tests and barely possible with a word test. Because maximum consonant specificity was decisive, a special test, the Adaptive Logatom Test, was designed to test whether NLFC provides sufficient benefit for people with mild to moderate hearing loss (i.e. phonemes cannot be discerned on the basis of word or sentence context) focusing particularly on the /s/ sound.

Results showed that the identification threshold of the /s/ sound clearly improved with NLFC. In addition, subjects reported that listening with NLFC was more pleasant than listening without it.

We therefore conclude that the Adaptive Logatom Test is a reliable method to determine high-frequency hearing loss in people with a mild to moderate hearing loss.

A33

Effects of hearing aid signal processing on cognitive outcome measurements

Elaine H. N. Ng[1,2,3], Mary Rudner[1,2,3], Thomas Lunner[1,2,3,4,5] and Jerker Rönnberg[1,2,3]

[1] *Department of Behavioral Sciences and Learning, Linköping University, Sweden*

[2] *The Swedish Institute for Disability Research, Linköping University, Sweden*

[3] *Linnaeus Centre HEAD, Linköping University, Sweden*

[4] *Oticon A/S, Research Centre Eriksholm, Snekkersten, Denmark*

[5] *Department of Clinical and Experimental Medicine, Linköping University, Sweden*

Sarampralis et al. (2009) showed the Ephrahim-Mallah noise reduction algorithm improved cognitive performance for people with normal hearing. However, better performance on the word-memory task for hearing-impaired listeners has not been reported. This study examines how signal processing in hearing aids affects the cognitive demands of speech recognition and the remaining cognitive capacity in those with hearing impairment. A dual task, which consists of a perceptual speech recognition task and a free recall memory task, was used to measure the cognitive outcomes with the use of binary time-frequency masking noise reduction technique (Wang et al., 2009). Experienced hearing aid users with symmetrical moderate to moderately-severe sensorineural hearing impairment were tested. The task was to report and to remember the final words of lists of eight Swedish Hearing In Noise Test (HINT) sentences. Working memory capacity was measured using a reading span test. In the dual task, individualized compensation of hearing loss using linear amplification was applied. Speech and noise stimuli were presented at individualized signal-to-noise ratios yielding 95% speech intelligibility in noise, so as to optimize equality in listening effort across participants. Memory performance in quiet and in two types of background noises (steady-state noise and four-talker babble) with three different noise reduction conditions (without noise reduction and with realistic and ideal binary masking) was analyzed.

Preliminary results show significant main effects of noise type and noise reduction. The two-way interaction was found to be significant, such that the noise reduction techniques improved the memory performance in the four-talker babble background. Such improvement was not observed in the steady-state noise background. Participants with higher reading span scores performed significantly better in the memory task. In addition, the position of the final words in each of the 8-sentence lists in memory task interacted with both noise type and noise reduction. This three-way interaction suggests the memory performance, particularly for the initial (primacy) and terminal (recency) items, was improved in the four-talker babble background with noise reduction. Reading span performance interacted with noise type when there was no noise reduction, indicating that persons with better working memory are relatively more distracted by competing background speech. The study demonstrates the binary-masking noise reduction technique helps freeing up cognitive resources and hence enhances memory task performance in the four-talker babble background. Memory performance was more disturbed in a competing speech background than in steady-state noise in individuals with better working memory capacity.

A34

The myth of the ideal binary mask in speech enhancement

Nilesh Madhu, Ann Spriet, Sophie Jansen and Jan Wouters

ExpORL, K.U.Leuven

The ideal binary mask has received a lot of interest lately as a tool for target speech enhancement in background noise. Indeed, it has also been touted as a possible goal of computational auditory scene analysis (CASA). Principally it exploits the sparsity and disjointness properties of speech spectra in their short-time—frequency representation, creating a mask that only preserves time-frequency regions where the target is dominant. This all-or-nothing decision is based on either a local or a fixed signal-to-noise-ratio (SNR) threshold. This approach has been demonstrated to yield appreciable intelligibility improvements in low SNR conditions under ideal settings.

However, such a 'hard-decision-first' approach is against the contemporary wisdom of soft-decisions as exemplified by the Wiener filter. This applies a weight to each time-frequency region in proportion to the SNR. Thereby hard decisions are avoided, allowing information to be preserved to a greater extent. Furthermore, due to the lower distortion caused by this soft mask, the extracted signals also sound more natural.

However, despite the demonstrable (and intuitively understandable) benefits of a soft-decision approach, currently there exist no systematic studies which compare these approaches against their binary counterparts.

We report here on a study to evaluate the usefulness of the binary mask as a possible goal of CASA and contrast it with the ideal Wiener filter. Our thesis is that the ideal Wiener filter, which does not make hard decisions like the binary mask, is a better model for the CASA, both in terms of intelligibility and quality. This thesis is quantified by perceptual experiments with about 20 listeners, and at different SNRs. We conclude that focus should shift to algorithms that can approach the ideal Wiener filter in performance, rather than algorithms that try to estimate the ideal binary mask.

A35

Effect of speech material on the benefit of temporal fine structure information in speech for normal-hearing and hearing-impaired subjects

Thomas Lunner, Renskje K. Hietkamp and Martin R. Andersen
Oticon A/S, Research Center Eriksholm, Kongevejen 243, DK-3070 Snekkersten, Denmark.
Kathryn Hopkins and Brian C. J. Moore
Department of Experimental Psychology, University of Cambridge, Downing Street, Cambridge CB2 3EB, United Kingdom

Hopkins et al. [J. Acoust. Soc. Am. 123, 1140-1153, (2008)] measured the speech reception threshold (SRT) for a target talker in a background talker as a function of the frequency range over which temporal fine structure (TFS) information was available. The signal was split into 32 1-$\mathrm{ERB_N}$ wide channels. Above a cut-off channel, *CO*, channels were vocoded, and contained only temporal envelope information. Channels up to and including *CO* were not processed. As *CO* was increased, SRTs decreased more for normal-hearing subjects than for subjects with cochlear hearing loss, suggesting that the latter were less able to use TFS information. The experiments reported here revealed similar results to those of Hopkins et al. when naturally spoken sentence materials with variable structure were used. For Danish HINT sentences, the mean change in SRT when *CO* was changed from 0 (fully vocoded) to 32 (unprocessed) was 6.2 dB for the normal-hearing subjects and 3.4 dB for the hearing-impaired subjects. Thus, for the HINT materials the normal-hearing subjects obtained greater benefit from the addition of TFS information than the hearing-impaired subjects. However, when stylized materials with a highly predictable structure were used (the Danish Dantale 2 materials), the benefit of adding TFS information by increasing *CO* was small and was similar for normal-hearing and hearing-impaired subjects. A speech material analysis supported the idea that that the Dantale 2 material had a more predictable structure than the Danish HINT material; the standard deviation of the mean word length was 81 ms for the Dantale 2 sentences, and 172 ms for the HINT sentences. It is proposed that TFS information is useful for reducing informational masking, by providing cues for the perceptual segregation of signal and background. When the target speech is highly predictable, informational masking may be minimal, rendering TFS cues unnecessary.

A36

Anatomical correlates of temporal resolution in patients with Multiple Sclerosis

M. Samantha Lewis[1,2], William Rooney[3], James Pollaro[3], Michele Hutter[1], Debra Wilmington[1,2], Linda Casiana[1], Stephen Fausti[1,2,5], Robert Folmer[1,2], Garnett McMillan[1], David Lilly[1], Mary Fitzpatrick[4] and Dennis Bourdette[1,4,5]

[1] *National Center for Rehabilitative Auditory Research (NCRAR), Portland VA Medical Center, Portland, OR*
[2] *Department of Otolaryngology/Head and Neck Surgery, Oregon Health and Science University, Portland, OR*
[3] *Advanced Imaging Research Center, Oregon Health and Science University, Portland, OR*
[4] *Neurology Service, Portland VA Medical Center, Portland, OR*

[5] *Department of Neurology, Oregon Health and Science University, Portland, OR*

<u>Rationale/Purpose:</u> Some studies have suggested that individuals with multiple sclerosis (MS) may be susceptible to auditory temporal processing problems. The purpose of the present investigation is to compare the temporal-resolution performance of individuals with MS to that of individuals without MS. Comparisons between groups were accomplished using a clinical measure of temporal resolution, the gaps-in-noise (GIN) test, and using functional magnetic resonance imaging (MRI).

<u>Methods:</u> Thus far, testing has been completed on 18 subjects with MS and 18 control subjects without MS. These subjects met the following inclusion criteria: 1) aged 21–65 years; 2) absence of current major disease other than MS; 3) native speaker of English; 4) less than a mild degree of hearing loss bilaterally; and 4) no left-handedness. The subjects with MS also met the following additional criteria: 1) a diagnosis of relapsing-remitting, primary progressive, or secondary progressive MS; 2) a Kurtzke Expanded Disability Status Score (EDSS) of 0 to 7.0, inclusive; 3) no history of a clinical relapse or change in EDSS for three months preceding entry into the study; and 4) a brain MRI scan that shows at least three white-matter lesions on T2-weighted images consistent with MS.

All subjects completed the following: 1) the GIN test, 2) an anatomical MRI, and 3) a functional MRI conducted during a gap-detection task. Gap-detection paradigms are thought to be sensitive to lesions of the cortex and the corpus callosum.

<u>Results/Discussion:</u> Independent-samples t-tests revealed that there was not a significant difference between experimental groups on the GIN test for either the right ear or for the left ear. Additional independent-samples t-tests, however, revealed that there were significant differences between groups in terms of, fractional cerebral spinal fluid volume ($p=0.004$; control < MS) and fractional callosal volume ($p=0.028$; controls > MS). Our data suggests decreased cortical gray matter (GM) volume fractions in frontal, temporal, occipital, and parietal lobes in MS compared to controls. Functional MRI revealed that different areas of the brain were being recruited in the subjects with MS versus the subjects without MS to perform the gap-detection task. This plasticity in the cortex may provide compensation for reduced cortical GM and callosal cross volume in the MS group which could impair auditory processing efficiency and interhemispheric communication. These findings suggest that there may be differences in how individuals with MS process auditory information. [This work was supported by the VA Rehabilitation Research and Development Service].

Posters for Session B should be put up by 8 AM Friday, August 13, and taken down after 10 PM Friday, August 13 or before 7 AM Saturday, August 14. Presenters should be at their posters from 9:45 AM – 11:10 AM; 4:30 - 5:00 PM.

POSTER SESSION B

Friday 9:45 AM – 11:10 AM

B1

A preliminary investigation into delay and phase preferences with music, open-canal hearing aids, and mild hearing loss

Justin Zakis, Bernadette Fulton and Brenton Steele
Dynamic Hearing Pty Ltd

Most previous studies on delay in hearing aids investigated the interaction between the bone-conducted and delayed own-voice signals in the occluded ear canal. Few studies have investigated the effects of group delay with open-canal aids and/or more broadband stimuli such as music. Open-canal devices can allow the unaided and delayed aided sounds to interact over a wide bandwidth, and are typically fitted to individuals with relatively good low-frequency hearing sensitivity. We are aware of only one previous study that investigated the effect of delay on music with an open-canal fitting. That study showed that with a flat insertion gain of 10 dB and one music stimulus, the mildly hearing-impaired participants found delays of 2, 4, and 10 ms to be 'not at all disturbing', while the normal-hearing participants found the 10-ms delay significantly more disturbing than 4 ms (Groth and Sondergaard, 2004). The current study aimed to investigate whether such findings could be extended to near-ideal conditions for detecting delay effects. These conditions were: 1) Around 0 dB insertion gain to maximise comb-filtering effects; 2) The use of two music stimuli where delay effects were audible to people with normal hearing; and 3) The recruitment of musicians as participants. In the first experiment, 12 mildly impaired musicians performed blind paired comparisons of the following hearing-aid signal-path processing conditions for each music sample: 1) Low Delay (LD) – linear phase response, 3.4 ms nominal group delay; 2) Ultra Low Delay (ULD) – minimum phase response, 1.4 ms nominal group delay; and 3) ULD followed by a 2-ms delay line (ULD+2ms). The third condition aimed to give the nominal delay of the LD processing with the phase response of the ULD processing. For each comparison, the participants nominated their preferred condition and rated the strength of their preference. In the second experiment, each processing condition was blindly compared with a muted hearing-aid output (i.e. no aided signal). Individual preferences could be strong for all compared conditions. At a nominal group delay of 3.4 ms, the minimum phase response (ULD+2ms) was significantly preferred to the linear phase response (LD) for one music sample and vice-versa for the other sample ($p < 0.04$). All other group preferences (including aided versus muted) were non-significant, and the implications of these findings will be discussed.

Groth, J. & Sondergaard, M.B. (2004). Disturbance caused by varying propagation delay in non-occluding hearing aid fittings. International Journal of Audiology, 43, 594-599.

B2

Evaluation of nonlinear frequency compression for school-age children with moderate to moderately-severe hearing loss

Myriel Nyffeler[1], Jace Wolfe[2], Andrew John[3] and Erin Schafer[4]

1. *Phonak AG, Switzerland*
2. *Hearts for Hearing Foundation, Oklahoma, US*
3. *University of Oklahoma Health Sciences Center, Oklahoma, US*
4. *Department of Speech and Hearing Science, University of North Texas, US*

Although children with mild to moderately severe hearing loss typically receive considerable benefit from hearing aid use, they frequently experience difficulty with certain aspects of speech recognition and production. However, the provision of sufficient amplification for high-frequency phonemes, such as /s/ and /z/, is critical for the development of language and speech production. An inability to attain these cues through audition can lead to the development of syntactic and semantic errors in language development.

This study evaluated nonlinear frequency compression (NLFC) as an alternative that may successfully address these difficulties of children with moderate hearing loss. A group of 15 children were fitted with micro-sized, behind-the-ear hearing aids with NLFC. Aided thresholds and speech recognition in quiet were assessed after 6 weeks of use without NLFC and 6 weeks of use with NLFC by word recognition assessments with the University of Western Ontario (UWO) Plural test and recognition of VCV nonsense syllables with the Logatom test. Speech recognition in noise was assessed using the signal-to-noise ratio in dB for 50% correct performance on the BKB-SIN test. Participants were randomly assigned to counterbalanced groups.

Overall, the results of this study conclusively demonstrate that NLFC is a viable means to provide consistent audibility of sounds through 8000 Hz for children with moderate hearing loss. Specifically, aided thresholds for high-frequency stimuli were significantly better when NLFC was enabled, and use of NLFC allowed significantly better speech recognition in quiet for the UWO Plural test and for tokens that included the phonemes /d/ and /s/ on the Logatom test. There was not a statistically significant difference in performance on BKB-SIN test between the NLFC enabled and disabled conditions. These results indicate that NLFC improves audibility for and recognition of high-frequency speech sounds for children with moderate to moderately severe hearing loss.

B3

Perception of sound position and distance in noise with bilateral hearing aids

Martin F. Müller[1], Andrea Kegel[1], Markus Hofbauer[2] and Norbert Dillier[1]

1 *Laboratory for Experimental Audiology University Hospital Zürich, Switzerland*
2 *Phonak AG, Stäfa, Switzerland*

Hearing aid users often complain that the spatial reproduction of sound with their hearing devices is rather poor, while achieving a relatively good increase in speech intelligibility [Noble and Gatehouse, IJA06]. The perception of the auditory space is distorted which makes the correct localization of the various auditory objects a difficult task [Van den Bogaert et. al, JASA05]. In this paper, we investigate how spatial quality is altered by common independent hearing aids and focus on the perception of direction and distance of sound sources in realistic environments.

In the first experiment, we examined the localization of real and virtual sound sources. We simulated realistic and complex acoustical scenes by combining individual head-related transfer functions (HRTFs) with a room acoustics simulator [Schimmel et. al, ICASSP09]. Based on everyday relevance, we implemented four realistic noisy scenarios: a crowded cafeteria, a busy office, a noisy street and a noisy and windy forest. In each of the scenes the test subjects were asked to localize a target sound source (a male speaker, a telephone, an ambulance, a bird) in the scene-dependent noise (babble noise, office machine and babble noise, street noise, wind and river noise). The target signals were selected to cover a wide range of localization cues. We included common behind-the-ear (BTE) hearing aid algorithms: a static beamformer, a monaural noise canceller and the omnidirectional condition. The level of the target sound was set to 60 dB for an SNR of 5 dB. Twelve normal-hearing subjects took part at the experiment.

In the second experiment, we investigated how distance cues are preserved by bilateral BTE hearing aids. For this, we reproduced Akeroyd's distance test [Akeroyd et. al, JASA06] in noise using virtual acoustics. For two reference positions at 2 and 5 meters, a male and a female speaker were presented at separated distances from the listener. We asked the test subjects to indicate which of the talker was the closest to his location and varied the distance between the stimuli. The same conditions as in the first experiment were evaluated.

The results show that the localization ability constantly varies over the scenes, depending on the set of cues available in the target sound. The hearing aid algorithms alter spatial auditory perception differently. The loss of pinna cues, due to the positions of the microphone, increased the perceived diffuseness of the target sound and the rate of front-back confusions. The noise canceller scored worst among the tested algorithms, while the beamformer achieved the best localization score. Overall, our results suggest that virtual acoustics are a good tool for representing artificial sound scenes and can be used for the evaluation of the spatial quality of hearing instruments in realistic conditions.

B4

Effects of dichotic-listening binaural hearing aid processing on speech intelligibility for listeners with sensorineural hearing impairment

Motokuni Itoh[1], Hikaru Takato[2] and Yoshiaki Takagi[1]
1 Panasonic Shikoku Electronics Co., Ltd
2 Tohoku Bunka Gakuen University

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 the present study, we adopted a simple dichotic listening in which speech signals were divided into two bands, and examined the effectiveness of this technique on speech intelligibility.

A listening test was conducted for 27 subjects with mild to moderate high-frequency sensorineural hearing loss. Forty VCV syllables were divided into higher and lower frequency components, taking into account the formant frequency of the preceding vowels. The intelligibilities of consonants and preceding vowels were analyzed for three divided frequencies and both conditions concerning which frequency components were presented to the left and right ears respectively. The results are as follows: 1) Dichotic listening improves consonant intelligibility for all the divided frequencies. However, the degree of improvement varies depending on consonants; 2) There has also been an improvement in intelligibility of the preceding vowels, /u/, /i/ and /e/; 3) The sum of hearing levels in the frequency band where either higher or lower component is presented to either ear is figured out (higher for right and lower for left, or higher for left and lower for right), resulting in two patterns depending on the combination of the ear and components. Selecting the smaller sum makes dichotic listening more effective.

B5

Perceptual consequences of mild hearing loss and benefits of extended bandwidth in hearing aids

Brian C.J. Moore, Christian Füllgrabe, Brian R. Glasberg, Michael A. Stone and Thomas Baer
Department of Experimental Psychology, University of Cambridge, UK

People with mild hearing loss over the frequency range that is most important for speech intelligibility (roughly 0.5 to 4 kHz) rarely use hearing aids, although they do seem to have difficulties in understanding speech in situations where background noise and/or reverberation are present. In the first part of this presentation, I will review evidence suggesting that these difficulties stem partly from a reduced ability to process temporal envelope and/or temporal fine structure information, as revealed in psychoacoustic tests using narrowband low-level stimuli, such that restricted regions in the cochlea are assessed. The perceptual deficits are not compensated by hearing aids, which may partly account for the limited use of hearing aids by people with mild hearing loss.

A second possible reason for the limited use of hearing aids is the limited bandwidth of most hearing aids. People with mild hearing loss may be candidates for extended-bandwidth hearing aids. We have developed a method for fitting such hearing aids called CAMEQ2-HF (Moore *et al.*, 2010). Several laboratory and field studies using this fitting method for subjects with mild hearing loss have shown: (1) An increase in bandwidth from 5 to 7.5 kHz was clearly audible for most subjects and an increase from 7.5 to 10 kHz was usually audible; (2) An increase in bandwidth from 5 to 7.5 kHz improved the ability to detect word-final /s/ and /z/; (3) Under conditions simulating listening via multi-channel (fast or slow) compression hearing aids, there was a small but significant benefit for speech intelligibility of increasing bandwidth from 5 to 7.5 kHz, when the target came from one side of the head and the interferer from the other. There was no significant benefit of increasing bandwidth from 7.5 to 10 kHz.

Reference:
Moore, B. C. J., Glasberg, B. R., and Stone, M. A. (2010). "Development of a new method for deriving

initial fittings for hearing aids with multi-channel compression: CAMEQ2-HF," Int. J. Audiol. 49, 216-227.

B6

Temporal processing and time-compressed speech tests as predictors of speech understanding in noise

Michelle R. Molis, Keri O'Connell Bennett and Marjorie R. Leek
National Center for Rehabilitative Auditory Research

This study investigated whether performance on time-compressed speech recognition tests can predict hearing-impaired listeners' benefit from fast or slow compression release times such as those employed in modern hearing aids. Speech recognition in the presence of three types of background noise (continuous speech-shaped noise, speech-modulated speech-shaped noise, and a single talker) was evaluated using IEEE sentence lists. The sentences were processed for each individual listener to simulate an appropriately-fit hearing aid with either fast or slow compression release times, as well as with linear processing. Preliminary results indicate no effect of compression type on speech understanding performance in noise, although listeners indicated a preference for linear processing over either compression type. Performance on the time-compressed speech test was correlated with speech- in-noise understanding but did not predict differential performance on any of the hearing-aid processed sentence tests. Additionally, measures of working memory and temporal auditory processing were assessed. In contrast to earlier results reported in the literature, results of the current study indicate that performance on time-compressed speech recognition and speech-in-noise understanding are not correlated with tests of working memory. However, speech understanding in all three masking noise conditions was significantly correlated with performance on the Gaps-in-Noise test. This suggests a strong effect of temporal auditory processing on speech-in-noise perception. [Work supported by the VA Rehabilitation Research and Development Service.]

B7

The benefit of binaural beamforming for hearing impaired listeners

Pauli Minnaar, Ulrik Kjems and Lars Bramsløw
Oticon A/S, Smørum, Denmark

In modern hearing aids it is possible to stream audio wirelessly from one ear to the other. This makes it possible to combine the signals at the two ears to form a binaural beamformer, which can suppress sound from one side of the head. Due to the relatively large distance between the microphones, a binaural beamformer is, however, only effective at low frequencies (below about 1.2 kHz) and closed-fit hearing aids are needed in order to get the maximal effect.

In order to test the possibilities of binaural streaming, three signal processing configurations have been implemented in prototype hearing aids running in real time on a PC. In one configuration, referred to as "signal transfer", the microphone signal at the left ear is played in both ears. This effectively suppresses high-frequency sound from the right side due to the head-shadowing effect. In another configuration, referred to as "binaural beamforming", the high-frequency signal is the same as for signal transfer, but a binaural beamformer is used at low frequencies. Thus, in both configurations, the signals are presented diotically. An omni-directional configuration, with no binaural processing, was used as a reference.

Listening experiments were done with 10 normal-hearing (NH) and 12 hearing-impaired (HI) listeners, with large (60-80 dB) low-frequency hearing losses. Close-fit hearing aids were used and hearing losses were compensated using a linear half-gain rule. A speech intelligibility test was done with the target talker from the front and noise to the right side of the listener. The results for HI listeners show that speech reception thresholds (SRT) were improved significantly for binaural beamforming, whereas signal transfer was not significantly different from the omni-directional reference. For NH listeners, neither signal transfer nor binaural beamforming were significantly different from the reference.

A preference test was also done, where listeners could take all aspects of sound into account in their judgements. For this test, talkers in a cafeteria environment and a very reverberant room were simulated through an array of 16 loudspeakers. The results for HI listeners show that both signal transfer and binaural beamforming were preferred over the reference. This was contradicted in interviews, however, where most listeners said that they disliked the in-head localization caused by the diotic configurations. NH listeners didn't prefer any of the binaural streaming configurations. This may be because these configurations do not convey the natural binaural cues, which are preserved in the omni-directional reference.

B8

Feature verification for stops and fricatives

Anjali Menon and Jont B. Allen
Department of Electrical and Computer Engineering, University of Illinois at Urbana Champaign

Hearing aid research has made it possible to deliver advanced hearing aid systems into the market, especially over the last decade. However, a major problem with them is the poor intelligibility of speech for a significant section of hearing aid users. Rectifying this needs a thorough knowledge of the perceptual cues of speech. In our previous study, a new methodology was developed to find the invariant acoustic cues or "features" for 14 Consonants Vowel (CV) pairs namely /pa/, /ta/, /ka/, /ba/, /da/, /ga/, /fa/, /Ta/, /sa/, /Sa/, /va/, /Da/, /Za/ and /za/. This methodology, called the 3D Deep Search (3DDS) method (JASA-Li, Menon and Allen (2010)), uses data from three independent psychoacoustic experiments to find the time –frequency coordinate that defines the feature region for a particular sound. The present study reports the results of an experiment conducted to verify these features regions using perceptual tests. Consonant Vowels, with their feature regions removed using the Short Time Fourier Transform, were used as stimuli. A comparison between unmodified speech sounds and those without the acoustic cues was made. In most of the cases, the scores dropped from 100% to chance at most SNR levels. This clearly emphasizes the importance of invariant features in identifying each CV. A study of the confusions reaffirmed the feature analysis while several new confusion groups were seen. For example, /ba,va,fa/ form a confusion group, deviating from the traditional Miller Nicely confusion groups. Moreover, even in the absence of the acoustic feature, there was a definite pattern in the confusions reported by the subjects. This makes it possible not only to identify the features of different sounds, but also to predict what will be heard when the feature region is absent. The 3DDS method was only used on for CV pairs with the vowel /a/. However, once the feature regions were identified, it was possible to generalize the results to other vowels as well. This study has been conducted in the presence of 2 vowels at 4 different levels of masking by speech weighted noise. CVs spoken by 18 different talkers (male and female) were used as stimuli.

B9

Sharpness and roughness perception in normal-hearing and hearing-impaired listeners

Martin F. McKinney, Kelly Fitz and Tao Zhang, Starkey Laboratories

Sharpness and roughness are psychoacoustic percepts thought to underlie aspects of music perception and general audio quality. *Sharpness*, the perception of the prominence of high-frequency acoustic energy, correlates with a primary dimension of musical timbre perception and has been used in models of sensory pleasantness [von Bismarck, *Acustica* 1974;30:159-72]. *Roughness*, the perception of temporal envelope modulations in the region of 20-170 Hz, is thought to contribute to musical dissonance perception and has also been used as a component in models of sensory pleasantness [Terhardt, *Acustica* 1974;30:201-13]. To understand how hearing loss affects these percepts, we conducted a set of rating experiments to identify differences in sharpness and roughness perception between normal-hearing and hearing-impaired listeners.

Seventeen hearing-impaired and six normal-hearing subjects rated the sharpness of sounds on a scale of 0 to 1. The sounds included a set of 19 band-pass noises and a set of 18 musical instrument tones. All stimuli were presented in both iso-level and iso-loudness (as predicted by a loudness model) conditions. Initial results show that some differences in ratings across listener-type in the iso-level conditions are usually, but not always, reduced in the iso-loudness conditions.

To investigate roughness perception, twenty-one hearing-impaired and six normal-hearing subjects rated the roughness of sounds on a scale of 0 to 1. The sounds included three sets of AM-tones covering a range of modulation indices, and two octave ranges of pure- and complex-tone musical intervals. All stimuli were presented in iso-level conditions. The complex-tone musical intervals were also presented in an iso-loudness condition. Initial results show some differences between hearing-impaired and normal-hearing listeners, but general trends are still unclear.

We will present further analyses for both experiments as well as extensions to current psychoacoustic models in an effort to account for hearing impairment. Our results will contribute to a better understanding of how hearing impairment affects music and audio quality perception.

B10

Comparative study of eleven noise reduction techniques for binaural hearing aids

J. I. Marin-Hurtado, D. V. Anderson
School of Electrical and Computer Engineering
Georgia Institute of Technology, Atlanta, GA 30332

Binaural hearing aids may make use of a wireless communication link between both hearing aids in order to implement more sophisticated signal processing algorithms for noise reduction and spatial filtering. Psycho-acoustical studies conducted on hearing impaired individuals have shown that binaural processing allows the user to understand speech under hostile environments (e.g. diffuse or babble noise) better than independent monaural processing. A wide range of binaural processing techniques has been proposed independently in the literature for background noise cancellation and enhancement of the source signal (which may arrive from a direction other than the front). This work performs a comparative study of eleven different noise reduction algorithms in order to identify the methods more suitable for a binaural hearing aid. The algorithms analyzed correspond to three different categories: scene analysis-based processing, adaptive beamforming, and multichannel Wiener filtering (MWF). Simulations of the performance of these algorithms under six different scenarios and using one and two microphones per hearing aid were conducted. These scenarios include single source under static-amplitude background noise, multitalker, single source under dynamic-amplitude background noise, moving source in a clear environment, moving source in a noisy environment, and moving source in a dynamic-amplitude background noise. The improvement on the signal-to-noise ratio (SNR) and an objective quality assessment were used to compare these algorithms. Results showed that the scene analysis algorithms analyzed provided the best SNR improvement, but the poorest performance in the quality of output sound. On the contrary, beamforming algorithms show poor performance from both SNR improvement and quality viewpoints. In average, the MWF techniques, in particular, the distributive-iterative MWF (DB-MWF) and a MWF technique that preserves binaural cues of noise (N-MWF), provide the better performance. Furthermore, algorithms preserving the binaural cues provide a better SNR improvement. Finally, no perceptual difference or high noise reduction improvement was achieved by transmitting two microphone signals to the contra-lateral hearing aid. Therefore, one microphone per hearing aid and transmitting this single signal can perform well. [This work was supported by National Semiconductor].

B11

Assessing the applicability of instrumental measures for black-box evaluation of feedback control in hearing aids

Nilesh Madhu[1], Ann Spriet[1], Thomas Bisitz[2], Jan Wouters[1] and Volker Hohmann[3]
1 ExpORL, K.U.Leuven
2 Hoertech GmbH
3 Carl von Ossietzky-Universität Oldenburg

There is an increasing trend towards the use of hearing aids with open and vented couplings in hearing impaired subjects. Especially with such couplings, the closed-loop system from the receiver of the hearing aid to the microphone(s) could become unstable when a large forward path gain is introduced, leading to annoying acoustic feedback. This places a limitation on the maximum amplification the hearing aid can provide and, for users with moderate to severe hearing loss, this might not be enough. Adaptive feedback control systems help reduce the effects of unstability, making them a critical part of the signal processing chain in digital hearing aids. Consequently, development of robust procedures and measures to quantify their performance become imperative.

We report on the evaluation of feedback control systems for commercial hearing instruments. The aim of the study is to define and examine the ability of instrumental measures to quantify the performance of the feedback control system in black-box settings and on realistic signals, when more than one element of the signal processing chain may be active (compression, noise suppression, microphone directionality, etc.). The evaluation is carried out on 6 different hearing aids of various manufacturers, for 10 measures, 8 different kinds of stimuli and varying hearing loss patterns. Thereby it is possible to see which measure is best suited to measuring which specific characteristic of the feedback control system, and serves as a beginning for conducting perceptual tests. Furthermore, the use of a wide variety of stimuli also gives an indication of which kind of stimulus presents a more challenging situation for a feedback control system. The study uses static (but variable) feedback paths and is based on signals recorded from the in-ear microphone of an artificial head, on which the hearing instruments are mounted.

We stress that it is not the aim of this study to present a comparison of the feedback control systems of different manufacturers. Rather, intentionally selecting a wide range of settings, hearing aids and stimuli allows us to identify challenging scenarios for the feedback control systems and to assess the general applicability of instrumental measures for feedback control system evaluation.

B12

Acoustical and perceptual effects of instantaneous and fast-acting compression

Alexandru Madalin Simion[1], Henrik Lodberg Olsen[2], Jörg M. Buchholz[1], Lars Bramsløw[2] and Torsten Dau[1]
1 *Centre for Applied Hearing Research*
2 *Oticon A/S, Smoerum, Denmark*

This project has implemented and tested an alternative dynamic range compression scheme for hearing aids, which is based on the concepts described by Herzke and Hohmann (2005). The proposed instantaneous compression system acts directly on the instantaneous Hilbert envelope in a multiband gamma tone filter bank. The idea is that the introduced nonlinearity replaces the cochlear compression lossdue to hearing loss. The implementation of such a compression system is inspired from models on cochlear compression in normal and impaired hearing.

However, the nonlinearity may have other drawbacks such as increased distortion. The trade-off between degree of compression and the perception of distortion is an important issue that was investigated in this study.

The speed and the realization of such compression schemes are expected to have an effect on the interaction between perceptual benefit and distortion. For this reason a more traditional compression scheme was also included together with a linear reference. The traditional compression was based on the same filterbank but used a band-specific RMS detector and a time constant.

To evaluate the different compression schemes both acoustical metrics as well as perceptual measures were obtained. The acoustical metrics used were EDI (Envelope difference Index), effective compression ratio based on short-term level distribution, and distortion produced with Hilbert-pair related signals as described by Olofsson and Hansen (2006). Perceptually, sound quality rating and speech intelligibility in noise were measured both in hearing- impaired and normal-hearing listeners. Preliminary results of the distortion measure shows, as expected, that the instantaneous compression contains the most distortion and the linear reference the least. The speed of the fast-acting compression was controlled by low-pass filtering the control signal in each channel which also controls the amount of distortion produced by this system. Perceptual results will be presented in the poster.

References:
Herzke & Hohmann, 2005: Effects of instantaneous multiband dynamic compression on speech intelligibility. EURASIP journal on applied signal processing, 2005, no.18, 3034-3043

Olofsson, & Hansen, 2006:Objectively measured and subjectively perceived distortion in nonlinear systems. Journal of the Acoustical Society of America; 120 no.6 - p. 3759-3769, 2006

B13

Delivery of sound to the cochlea via the cerebrospinal fluid

Professor Alan J Lupin, F.R.C.S. (C.), F.A.C.S.
Victoria, B.C., Canada

The development of implantable hearing aids has been attempted for more than three decades. To date, there has been no long-term device successful enough to compare with the advances in cochlear implant technology. This may well be related to the fact that most of the attempts at producing an implantable hearing aid have involved placing mechanical devices in the middle ear and usually making some connection to the ossicles. This may disturb the functioning of the ossicles, or interfere with their blood supply which is largely on their surface. These considerations led to the writer's investigation of a device which applies the sound to the dura and hence to the cerebrospinal fluid and brain. There is evidence in the literature that such an approach could be suitable and indeed we now have proof of the efficacy of this concept.

Since 2003, we have tested the application of sound to the dura in humans under general anaesthesia during cochlea implant surgery*. The surgical dissection allowed of examination of the skull, an island of bone on the dura, and other structures, including the round window membrane. A laser Doppler vibrometer was used to measure movements of the individual structures while various elements were vibrated with a transducer from the BAHA device. The efficacy of stimulating via the dura and cerebrospinal fluid was shown in 2005 and subsequently. This indicates that a new method has been found for the delivery of sound to the cochlea in an implantable device. Recent tests, again in humans under general anaesthesia, confirmed these findings over the whole range of frequencies used by the human ear and the findings were reproducible. The concept of a "third window" for the cochlea is advanced.

This use of the information concerning the pathways of sound in the human head, allows for a safe, short surgical procedure, preservation of the ossicles and middle ear, avoidance of surgery near the facial nerve, and yet with total implantation and with adequate enhancement of hearing. It has the added advantage of transmitting bilaterally to the cochleae from a unilateral implant. It is also ideal for use with a cochlea implant in patients with some residual hearing.

This poster presents the latest findings of our investigations into this novel understanding of the pathway taken by "bone conduction" sound waves in the human head and the history of the development of this concept. A live demonstration of this effect will be demonstrated at the poster.

<u>Acknowledgement:</u> I wish to acknowledge Professor Thomas Lenarz of Hannover Germany and his team for performing the tests.

B14

Sensitivity to low-frequency temporal fine structure is correlated with aided spatial release from masking

Thomas Lunner, Tobias Neher
Oticon A/S, Research Center Eriksholm, Kongevejen 243, DK-3070 Snekkersten, Denmark.
Kathryn Hopkins and Brian C. J. Moore
Department of Experimental Psychology, University of Cambridge, Downing Street, Cambridge CB2 3EB, United Kingdom.

In order to study the spatial hearing abilities of bilateral hearing aid users in multi-talker situations, Neher et al. (2009) fitted subjects with hearing aids configured to preserve acoustic cues salient for spatial hearing. Following acclimatization, speech reception thresholds (SRTs) were measured using three competing talkers that were either co-located or spatially separated along the left-right dimension. Both SRTs when the target and background were spatially separated and spatial release from masking (SRT for co-located condition minus SRT for spatially separated condition) varied over a range of more than 14 dB across subjects.

To investigate the importance of low-frequency temporal fine structure (TFS) cues for spatial release from masking, 17 of the 20 hearing-impaired subjects from the study of Neher et al. (2009) were tested using the TFS-LF test developed by Hopkins and Moore (2010). The TFS-LF test is based on measuring thresholds for detecting an interaural difference in the TFS of pure-tone stimuli. An adaptive two-alternative forced-choice task was used. Each interval contained four tones with the same frequency; in one interval all tones were diotic (AAAA), whereas in the other interval tones one and three were diotic while tones two and four had an interaural phase shift ($\Delta\Phi$, ABAB). The task was to identify the interval containing the B tones, in which the tones appeared to move from left to right. The outcome measure was expressed as the phase shift at threshold (71% correct). Two frequencies were tested, 250 and 750 Hz.

TFS-LF thresholds were significantly correlated (r = -0.58, p = 0.015) with spatial release from masking; lower TFS-LF thresholds were associated with larger spatial release from masking. This indicates that access to low-frequency TFS information is important for spatial hearing abilities.

References:

Hopkins, K., and Moore, B. C. J. (2010). "Development of a fast method for measuring sensitivity to temporal fine structure information at low frequencies," Int. J. Audiol. (submitted).

Neher, T., Behrens, T., Carlile, S., Jin, C., Kragelund, L., Petersen, A. S., Schaik, A. (2009). "Benefit from spatial separation of multiple talkers in bilateral hearing-aid users: Effects of hearing loss, age, and cognition," Int. J. Audiol. 48, 758-774.

B15

Anatomical correlates of dichotic listening in patients with Multiple Sclerosis

M. Samantha Lewis[1, 2], William Rooney[3], James Pollaro[3], Michele Hutter[1], Debra Wilmington[1, 2], Linda Casiana[1], Stephen Fausti[1, 2, 5], Robert Folmer[1, 2], Garnett McMillan[1], David Lilly[1], Mary Fitzpatrick[4] and Dennis Bourdette[1, 4, 5]

1 *National Center for Rehabilitative Auditory Research (NCRAR), Portland VA Medical Center, Portland, OR*
2 *Department of Otolaryngology/Head and Neck Surgery, Oregon Health and Science University, Portland, OR*
3 *Advanced Imaging Research Center, Oregon Health and Science University, Portland, OR*
4 *Neurology Service, Portland VA Medical Center, Portland, OR*
5 *Department of Neurology, Oregon Health and Science University, Portland, OR*

<u>Rationale/Purpose:</u> Some studies have suggested that individuals with multiple sclerosis (MS) may be susceptible to dichotic-listening deficits. The purpose of the present investigation is to compare the dichotic-listening performance of individuals with MS to that of individuals without MS. Comparisons between groups were accomplished using both a clinical measure of dichotic listening, the Dichotic Digits Test (DDT), and using functional magnetic resonance imaging (MRI).

<u>Methods:</u> Thus far, testing has been completed on 18 subjects with MS and 18 control subjects without MS. These subjects met the following inclusion criteria: 1) aged 21–65 years; 2) absence of current major disease other than MS; 3) native speaker of English; 4) less than a mild degree of hearing loss bilaterally; and 4) no left-handedness. The subjects with MS also met the following additional criteria: 1) a diagnosis of relapsing-remitting, primary progressive, or secondary progressive MS; 2) a Kurtzke Expanded Disability Status Score (EDSS) of 0 to 7.0, inclusive; 3) no history of a clinical relapse or change in EDSS for three months preceding entry into the study; and 4) a brain MRI scan that shows at least three white-matter lesions on T2-weighted images consistent with MS.

All subjects completed the following: 1) the DDT, 2) an anatomical MRI, and 3) a functional MRI conducted during both a diotic task and a dichotic task. Dichotic paradigms are thought to be sensitive to lesions of the cortex and the corpus callosum.

<u>Results/Discussion:</u> Independent-samples t-tests revealed that there was not a significant difference between experimental groups on the DDT for either ear. Additional independent-samples t-tests revealed that there were, however, significant differences between groups in terms of, fractional cerebral spinal fluid volume (p=0.004; control < MS) and fractional callosal volume (p=0.028; controls > MS). Our data suggests decreased cortical gray matter (GM) volume fractions in frontal, temporal, occipital, and parietal lobes in MS compared to controls. Functional

MRI revealed that different areas of the brain were being recruited in the subjects with MS versus the subjects without MS to perform the gap-detection task. This plasticity in the cortex may provide compensation for reduced cortical GM and callosal cross volume in the MS group which could impair auditory processing efficiency and interhemispheric communication. These findings suggest that there may be differences in how individuals with MS process auditory information. [This work was supported by the VA Rehabilitation Research and Development Service].

B16

Bimodal listeners show no sensitivity to interaural time differences in unmodulated low-frequency stimuli

Anneke Lenssen[1], Tom Francart[1], Jan Brokx[2] and Jan Wouters[1]

1 *ExpORL, Department of Neurosciences, Katholieke Universiteit Leuven*

2 *Department of Otorhinolaryngology and Head & Heck surgery, University Hospital Maastricht*

Users of a cochlear implant and a contralateral hearing aid (bimodal listeners) have shown sensitivity to ITD with modulated stimuli with frequency content between 600 Hz and 3600 Hz. Here, sensitivity to ITD for unmodulated stimuli was assessed in bimodal listeners who were good performers in previous ITD experiments.

Two types of stimuli were used: (1) an acoustic sinusoid combined with an electric transposed signal and (2) an acoustic sinusoid combined with an electric pulse train. The subjects were highly trained on the task and were tested in multiple test sessions.

In contrast to what was expected from normal hearing listeners, bimodal listeners showed no sensitivity to ITD with the stimuli used in the current study.

The results indicate that users of a cochlear implant and a hearing aid on the contralateral side would benefit more from a speech processing strategy providing salient envelope ITD cues than from a strategy that preserves fine-timing ITD cues present in acoustic signals.

B17

Pinna cue-preserving hearing-aid fittings: Audibility considerations

Søren Laugesen[1], Tobias Neher[1], René Burmand Johannesson[1], Niels Søgaard Jensen[1] and Thomas Behrens[2]

1 *Eriksholm, Oticon Research Centre*

2 *Oticon A/S*

When hearing-impaired test subjects take part in a listening test, making the acoustic stimulus audible across the entire frequency range of interest is typically essential. The importance – and many pitfalls – of this were elucidated by Humes (2007). In addition, work presented by Moore et al. (2008) points out that because of the dynamic range of the speech signal, making speech audible may involve more than just ensuring that the long-term RMS value is above hearing threshold. Similar considerations are just as relevant for other types of complex stimuli.

The present work forms part of a larger study of the potential benefit of making acoustic pinna cues available to hearing-aid users. Since pinna cues belong in the upper part of the audio bandwidth of hearing aids, where Hearing Thresholds Levels typically are greatest and the dynamic range of hearing typically is narrow, audibility considerations are of particular importance.

Regarding the amplification prescribed in hearing aids, further considerations come into play:

- Many natural signals (including speech) have little power at high frequencies.
- Furthermore, salient high-frequency information in speech is relatively sparse in time.
- Inter-individual variation in Real Ear to Coupler Differences is very large at high frequencies.

This poster will present examples of the audibility considerations that went into different parts of the pinna-cue study, including a headphone listening test of the listener's ability to detect changes in monaural spectral ripple patterns, and a free-field listening test (with open ears) of spatial release from speech-on-speech masking. Results obtained with the resulting amplification schemes will be presented in a companion poster (Jensen et al.). In addition, the gain prescription rule and an audiometric procedure for determining aided thresholds with the experimental hearing aids used in the pinna-cue field test (to be presented by Neher et al.) will be discussed.

References:

Humes L (2007), "The contributions of audibility and cognitive factors to the benefit provided by amplified speech to older adults," J Am Acad Audiol, 18, 590-603.

Moore BCJ, Stone MA, Füllgrabe C, Glasberg BR, Puria S (2008), "Spectro-temporal characteristics of speech at high frequencies, and the potential for restoration of audibility to people with mild-to-moderate hearing loss," Ear Hear, 29, 907-922.

B18

Objective and subjective analysis of different hearing instruments in First-Fit-Setting

Matthias Latzel[1], Inga Holube[2], Matthias Vormann[3] and Michael Schulte[3]

1 *Siemens Audiological Engineering Group Erlangen, Germany*
2 *Jade University of Applied Sciences, Institute of Hearing Technology and Audiology, Oldenburg, Germany*
3 *Hörzentrum Oldenburg, Germany*

Introduction: The initial setting of a hearing instrument based on pure tone audiometric data, or the so called *First Fit Setting*, plays an essential role in the spontaneous acceptance of a hearing instrument and the buying decision of the customer. As a result, studies are regularly conducted to investigate the spontaneous acceptance of new hearing aids and to identify their market opportunities.

The submitted poster describes two comparative studies on BTE-devices with 15 experienced hearing aid wearers each.

Method: One study compares four BTE-devices with external receivers while the other study compares five BTE-devices with conventional ear molds. In order to suppress the effect of confounding factors like size, form, color, wearing comfort etc., and to allow for conducting direct paired comparisons, a special method called "virtual hearing aid" is used in this study. Data for both subjective and objective measures are collected. Subjective measures include the subjects' evaluation of naturalness, sharpness, loudness, speech intelligibility, and overall satisfaction. Objective measures include percentile analysis of all real-ear recordings used for paired comparison, in-situ measurements, and coupler measurements with noise and the International Speech Test Signal (ISTS).

Results: The results show that "the superior hearing aid" does not exist, but that acceptance of hearing aids highly depends on the acoustical scenario. Such results somewhat contradict the experiences in the market and therefore demonstrate that the acceptance of a hearing aid and its *First-Fit-Setting* does not only depend on acoustical performance, but also on other non-technical factors e.g. quality, marketing, service etc.

Further results of the same study are shown by Holube et al. and Steinbuss et al. at this conference.

Conclusion: The analysis of the data, especially the correlation between the objective and subjective data, provides several meaningful indications on how to design the initial setting of a hearing aid in order to maximize the spontaneous acceptance in diverse acoustical scenarios.

B19

Adults with acquired hearing impairment seeking help for the first time: Predictors of the decision to obtain hearing aids, of hearing aid uptake, and of successful hearing aid outcomes

Ariane Laplante-Lévesque, Louise Hickson, and Linda Worrall, University of Queensland

Rehabilitation interventions such as hearing aids and communication programs are effective for adults with acquired hearing impairment, but their uptake, adherence, and outcomes are suboptimal. This study investigated the predictors of the decision to obtain hearing aids, of hearing aid uptake, and of successful hearing aid outcomes in a sample of 153 adults with acquired hearing impairment seeking help for the first time. Each participant met with the research audiologist who discussed intervention options: hearing aids, communication programs, and no intervention. The intervention options were presented to the participants with a decision aid. Participants considered the intervention options for at least one week before making their decision.

Of the total sample, 53% decided to obtain hearing aids, 27% decided to complete communication programs, and 20% decided not to complete any intervention. Six months later, over one-fifth (21%) of the participants who had decided to obtain hearing aids had not taken them up. For those who obtained hearing aids, self-reported outcomes (COSI, IOI-HA, and reduction in hearing disability) were recorded three months after hearing aid uptake.

Logistic and linear regression identified significant predictors of the decision to obtain hearing aids, of hearing aid uptake, and of successful hearing aid outcomes when other variables were held constant. Significant predictors were degree of hearing impairment, hearing disability, socio-economic status, application for subsidized hearing services, communication self-efficacy, stages of change, and loci of control. Clinicians should explicitly elicit the predictors identified here in order to successfully help adults with acquired hearing impairment make optimal intervention decisions that result in hearing aid uptake, adherence, and successful outcomes.

B20

Modification of audibility indices for use with frequency lowering hearing aids

Julie Ladiges, Vijay Parsa, Danielle Glista and Susan Scollie
The University of Western Ontario

There has been a renewed interest in frequency lowering technologies in digital hearing aids as an alternative amplification strategy for severe to profound high frequency hearing loss profiles. Frequency lowering technologies transfer high frequency information to lower frequency spectral regions with better residual hearing via frequency transposition or frequency compression signal processing algorithms. Speech-based electroacoustic indices (SII, AI, STI) are commonly used in hearing aid research to quantify the amount of speech that is audible to the hearing aid user, and sometimes to predict listeners' speech recognition scores. However, these indices do not account for the frequency-lowering effect in some modern hearing aids. Therefore, it is not known whether audibility indices can be applied to this category of technology. The purposes of this study are (a) to modify several published audibility indices for use with frequency lowering signal processing, and (b) to compare their relative accuracy in predicting speech recognition and/or loudness rating data. The indices compared include: (1) the Speech Intelligibility Index; (2) the Audibility Index (similar to that in the Situational Hearing Aid Response Profile); and (3) a loudness model based speech intelligibility metric for frequency lowering device users. This metric, called the articulation index, is based on the excitation pattern of speech and is derived from Moore and Glasberg's loudness model for hearing impaired listeners. Each index was computed from measures of patients' hearing aids using custom-developed codes in Matlab. Patients were enrolled in a multiple-measure study of outcomes with nonlinear frequency compression and indices will be compared to the patients' speech recognition scores. Preliminary results indicate that the loudness model approach may be more predictive than the other indices.

B21

Predicting speech quality using a computational auditory model

Abigail Kressner, Christopher Rozell and David V. Anderson, Georgia Tech

Presently, the most accepted method for evaluating speech quality and intelligibility is through listener tests. However, these tests are often time-consuming and costly. Therefore, it is often advantageous to use objective measures of speech quality and intelligibility prior to (or in place of) conducting listener tests.

By using a well-developed model of the auditory periphery [4], we are developing an objective measure of speech quality based on neural coding in the auditory nerve. Previously, Hu and Loizou evaluated the performance of a comprehensive selection of standard objective measures for predicting the quality of noisy speech enhanced by noise suppression algorithms [3]. By using the same speech samples and subjective evaluations, as presented in [2] (obtained with permission), we evaluate the performance of this metric while concurrently enabling direct comparison to standard measures. Preliminary results suggest that distance measures on the output of the computational model predict speech quality reasonably well (Pearson's correlation between -0.85 and -0.5, where higher scores indicate better performance for subjective evaluations and lower scores indicate better performance for the metric).

Conversely, an intelligibility metric based on auditory neural coding using an older version of the same computational model has been shown to correlate well with intelligibility (Neural Articulation Index) [1]. Given that the preliminary results suggest the auditory model can be used to predict speech quality well and that others have shown this model can be used to predict intelligibility well, this computational model can be used as a valuable tool for evaluating signal processing strategies in hearing aid design.

References

1 *J. Bondy, I.C. Bruce, S. Becker, and S. Haykin. Predicting speech intelligibility from a population of neurons. In NIPS 2003 Conference Proceedings: Advances in Neural Information Processing Systems, Volume 16, pages 1409–1416, 2004.*

2 *Y. Hu and P. C. Loizou. Subjective evaluation and comparison of speech enhancement algorithms. Speech Communication, 49:588–601, 2007.*

3 *Y. Hu and P. C. Loizou. Evaluation of objective quality measures for speech enhancement. IEEE Transactions on Acoustics, Speech and Signal Processing, 16(1):229, 2008.*

4 *M. S. A. Zilany and I. C. Bruce. Representation of the vowel /epsilon/ in normal and impaired auditory nerve fibers: model predictions of responses in cats. The Journal of the Acoustical Society of America, 122(1):402–17, July 2007.*

B22

The effect of lip-reading on identification of linear frequency transposed speech in hearing impaired adults

Petri Korhonen, Widex Office of Research in Clinical Amplification (ORCA), Lisle, IL

Several studies on linear frequency transposition have demonstrated its efficacy in speech recognition in quiet and in noise (Kuk et al. 2009). Few studies have investigated effects of audio-visual cues on phoneme identification in frequency lowering. Inherent to all frequency lowering techniques is that the new lowered acoustic cues may overlap perceptually with natural speech cues causing confusions between speech sounds that have similar acoustic representation after processing. However, both normal and hearing-impaired people use lip-reading consciously or unconsciously to aid comprehension. Some phoneme confusions that have very similar acoustic representations after processing (for example /s/ and /ʃ/) may be distinguished when using lip-reading. The current study examined the phoneme identification performance with commercially available linear frequency transposition algorithm when using lip-reading cues.

Ten test subjects with sloping hearing loss were fitted binaurally with Widex Mind440-9 BTE hearing aid using open ear tips. Their speech identification performance was tested using ORCA-NST nonsense bisyllable speech test with and without frequency transposition in quiet (50 dB SPL) and in noise (68 dB SPL, SNR = +5 dB). In addition to auditory only condition we also included a visual only and audiovisual conditions to study the phoneme confusion patterns when using lip-reading cues.

The phoneme identification scores for all phonemes in quiet were 4.5% better with frequency transposition than with conventional amplification in audio only mode ($p<0.005$). The use of nonsense bisyllable test allowed us to further analyze the results to identify the type of phoneme confusions that were present. Linear frequency transposition created confusions between some phonemes whose acoustic representation were similar after processing. There were eight phoneme pairs that were confused consistently (more than 30% of the time). Most commonly confused phoneme pairs were /ŋ,m/, /dʒ,d/, and /g,d/. The use of lip-reading cues improved the phoneme identification of transposed phonemes by 22.2%. Lip-reading cues improved seven of the eight most common phoneme confusions. In only one of the most commonly confused phoneme pairs (/g, d/) the phonemes shared the same viseme.

The results of the current study demonstrate that the potential phoneme confusions created by linear frequency transposition processing are typically between phonemes that do not share the same visemes, and therefore can be correctly identified in the presence of lip-reading cues.

Reference:

Kuk F, Keenan D, Korhonen P, Lau C (2009). Efficacy of linear frequency transposition on consonant identification in quiet and in noise. JAAA, 20(8):465-479.

B23

Closed-set sentence intelligibility tests with across-language compatibility

Birger Kollmeier, Sabine Hochmuth, Ania Warzybok, Melanie Zokoll-van der Laan, Thomas Brand*
*Medizinische Physik, Universität Oldenburg *and HörTech gGmbH, D-26111 Oldenburg, Germany*

In order to evaluate hearing aid performance in an internationally compatible and standardized way, appropriate speech intelligibility tests are necessary that are valid, efficient, and comparable across languages and laboratories. To achieve this goal, the development and validation of appropriate speech intelligibility tests for six languages (German, Dutch, English, French, Swedish, Polish) was performed within the Hearcom project (finished in 2009 [1]). This contribution focuses on the so-called "Matrix" test (motivated by [2] and [3]) and its extention to Spanish, Turkish, American English and Russian as well as the comparison of the test outcomes obtained with the respective other languages. The basic speech material consists of 10 names, 10 verbs, 10 numerals, 10 adjectives and 10 nouns. From this material a large number of syntactically equal, but semantically unpredictable sentences can be composed that are phonetically balanced. The test permits repeated measurements which are required, e.g., for hearing instrument fitting or research purposes.

The closed-set format and the comparability of average test results across languages enable the tests to be used with subjects in their native language, even if the test instructor can not comprehend the subjects language. To achieve a high efficiency of the tests for speech reception threshold (SRT) estimation, a steep slope of the discrimination function was obtained by maximizing the homogeneity in intelligibility across the recorded word materials. In addition, the recording conditions and the selection of test conditions were almost identical across languages.

The recordings of the speech material for all tests took place in Oldenburg using a method that accounts for the effect of coarticulation during resynthesis of the complete test sequences [3]. The results of the subsequent optimization procedure will be described where the word-specific speech discrimination functions were measured with native subjects. This led to high similarity in intelligibility across words, sentences and test lists. To demonstrate the compatibility with the languages covered so far with appropriate closed-set test methods, the resulting overall discrimination functions and between-list variability will be compared to the functions obtained for the respective other languages. [Supported by EFRE (European infrastructure fond, project HurDig)]

References:

1 *www.hearcom.org Outcomes of the European Integrated Project on Hearing in the communication society*

2 *Hagerman B., 1982, Scand Audiol, 11, 79-87*

3 *Wagener et al., 2003, Int J Audiol, 42, 10-17*

B24

Realistic sound environments: Can they improve the hearing instrument fitting process?

Martin Kinkel, KIND Hoergeraete, Kokenhorststr. 3-5, D-30938 Burgwedel

Realistic sound environments are used more and more in hearing aid fittings. The use of more advanced sound systems allows for significant improvements in several steps of the fitting process: E.g., it allows for a comparison of multiple hearing instruments under identical acoustical conditions, or to "test drive" hearing aid features during the counselling. For the fine tuning of hearing instruments, simulations of typical every-day listening situations can make the fitting process significantly more realistic and effective. Especially, for mild to moderate hearing impaired subjects and fittings using "open" hearing instruments, the fitting process could follow a time-condensed protocol integrating the single steps of a professional hearing instrument fitting in as few appointments as possible enabling for an overall less time-consuming procedure. Experiences with a commercially available procedure and results from studies will be discussed.

B25

Noise reduction algorithm evaluation using hearing loss simulator

Jin Ryoul Kim[1], Jonghee Han[1], Yoon Su Hwang[1], Sangwan Seo[1], Sung Hwa Hong[2], Sang Min Lee[3], Dong Wook Kim[4], Sun I. Kim[1] and In Young Kim[1]

1 *Department of Biomedical Engineering, Hanyang University*
2 *Department of ORL-HNS, Sungkyunkwan University*
3 *Department of Electronic Engineering, Inha University*
4 *Bio & Health Lab, Samsung Advanced Institute of Technology*

In digital hearing aids, noise reduction and hearing loss compensation is executed sequentially. In hearing aid algorithm evaluation, generally noise reduction algorithms have been evaluated by speech quality measures for only themselves. However, actual signal which are transferred to hearing impaired persons went through a hearing loss compensation algorithm. Hence after noise reduction algorithm is applied on an input noisy signal, hearing loss compensation algorithm compresses a dynamic range of the noise reduced output according to hearing impairment. Actual signal which hearing impaired persons finally listened is the compressed output. Therefore, it is necessary for better evaluation of noise reduction to evaluate the compressed output through hearing loss compensation algorithm.

This paper proposed new noise reduction algorithm evaluation protocol in order to consider hearing impairment. It applied a hearing loss simulator to the output of hearing loss compensation algorithm which was processed by noise reduction. Hearing loss simulator was set to consider hearing loss characteristics of patient (generic moderate). The proposed method used WDRC as hearing loss compensation algorithm, and HLsim (NIOSH) as hearing simulator, and TIMIT corpus as noisy data (noisy SNR: 5dB). It compared noise reduction algorithms (wiener filter, MBSS, logMMSE and om-lsa) through objective measurements (SNR, segSNR, fwsegSNR, LLR, composite, PESQ).

Without simulator, Best result of noise reduction was omlsa by objective measurements. (Improved measurement: SNR[2.53], segSNR[2.14], fwsegSNR[0.88], LLR[0.04], PESQ[0.08], composite-sig[-0.21], composite-bak[0.01], composite-ovl[0.09]). But with simulator, best result of noise reduction was MBSS by objective measurements. (Improved measurement: SNR[2], segSNR[2.84], fwsegSNR[-0.69], LLR[0.04], PESQ[0.09], composite-sig[-0.03], composite-bak[0.2], composite-ovl[0.02]).

The best algorithm along objective measurements was different between with simulator and without simulator. In the other words, it was different that hearing loss characteristics of patient consider or not. In order to consider hearing loss characteristics of patient, noise reduction evaluation should use a hearing loss simulator.

In conclusion, best algorithm for hearing impaired persons was MBSS among the mentioned 4 methods. [This work was supported by grand No. 10031731 from the Strategic Technology Development Program of Ministry of Knowledge Economy.]

B26

Wide dynamic range compression using Optimized Least-square non-uniform FFT (OLS-NUFFT)

Heepyung Kim[1], Jonghee Han[1], Kyeongwon Cho [1], Yun Su Hwang[1], Sung Hwa Hong[2], Sang Min Lee[3], Dong Wook Kim[4], Sun I. Kim[1] and In Young Kim[1]

1 *Department of Biomedical Engineering, Hanyang University*
2 *Department of ORL-HNS, Sungkyunkwan University*
3 *Department of Electronic Engineering, Inha University*
4 *Bio & Health Lab, Samsung Advanced Institute of Technology*

Frequency resolution is one of issues in the development of dynamic range compression. In the earlier studies about the dynamic range compression (DRC), speech intelligibility and quality were decreased by mismatch between the frequency resolution of FFT based DHA and the frequency resolution of Human auditory system. For this reason, several researches have been performed for development of dynamic range compression that matches the frequency resolution of DHA to the frequency resolution of bark scale based on Human auditory system to enhance the speech intelligibility and quality.

In this study, we suggest a wide dynamic range compression (WDRC) using optimized least-square non-uniform FFT (OLS-NUFFT) for matching the frequency resolution of DHA to the frequency resolution of bark scale. WDRC using OLS-NUFFT was based on the side-branch compression structure. This structure consists of 6 blocks: Input buffer and windowing, N point OLS-NUFFT, Frequency bin Power computation, compression gain computation, filter coefficient computation and FIR filter. OLS-NUFFT is used to supply non-uniform spacing in frequency domain and a memory efficient approximation. We used model based design using Simulink for easy modification and integration with other algorithms (Noise Reduction block, Beamfomming block and etc...). Also, we evaluated the WDRC using NUFFT system by mean opinion score (MOS) and speech reception threshold(SRT) test at 5 normal hearing persons that simulated by hearing loss simulator [HeLPS].

This study aim is enhancement of speech intelligibility and quality by WDRC using OLS-NUFFT. So, when processed signal WDRC using FFT and WDRC using OLS-NUFFT were simulated though HeLPS to 5 normal hearing persons, the later one showed higher MOS and SRT. [This work was supported by grand No. 10031764 from the strategic Technology Development Program of Ministry of Knowledge Economy.]

B27

Verification of wireless transceiver for In The Canal hearing aids

Dong-Wook Kim[1], Ju-Man park[1], Zia Mohy-Ud-Din[2], Sang-Hyo Woo[1], Ki-Woong Seong[3], Jyung-Hyun Lee[4], Myoung-Nam Kim[2] and Jin-Ho Cho[1]

1 *School of Electronics Engineering, College of IT Engineering, Kyungpook National University*
2 *Graduate School of Medical & Biological Engineering, Kyungpook National University*
3 *Advanced Research Center for Recovery of Human Sensibility, Kyungpook National University*
4 *Department of Biomedical Engineering, Kyungpook National University Hospital.*

Nowadays, the binaural hearing aids are being developed with advances in related technology such as digital signal processing (DSP), wireless communications, and integrated circuit design. The binaural hearing aid requires two hearing aids for both side ear and wireless communication between each other hearing aids for the improvement of the speech perception. The popular type of binaural hearing aids is behind the ear (BTE) type hearing aids because the in the ear (ITC) or smaller size hearing aids has problem of a battery capacity and limitation of wireless transceiver size. However, the BTE type hearing aids have a cosmetic problem and ITC type hearing aids can solve this problem. Therefore, low power consumption and small size wireless transceiver is a fundamental hardware for implementation of ITC type binaural hearing aids.

In this paper, we design and implement the small size and low power consumption wireless transceiver for ITC type hearing aids. In order to implement a wireless transceiver, we used a RF chip from Nordic Semiconductor, simple 8051 controller from Silicon Labs and the 8.5 X 2.2 mm small size antenna from Antenna Factor. The size of implemented transceiver was only 7 X 7 mm that excepts the antenna part. The size of implemented transceiver with antenna could put in the ITC type ear shell. The ear shell was custom designed from a hearing aid company. The implemented transceiver used 2.4 GHz RF band that is industrial, scientific and medical (ISM) band. Also, the finite difference time domain (FDTD) method and Maxwell’s curl equations were used for RF radiation pattern simulation of a designed wireless transceiver.

In order to measure the RF radiation pattern, kemar manikin as RF phantom which is filled with the phantom liquid was used and ingredients of liquid are combined with water, diethylene glycol mono-buthyl ether (DGBE), trition and NaCl. The transceiver can send 32 Kbits/second with average power consumption of 0.966 mW. When the reference antenna was placed at a distance of 2 m from the transmitter, the RF radiation pattern is measured with and without the phantom and the maximum attenuation from the phantom was observed to be 23dB.

As a result, the implemented system showed the feasibility to communicate with opposite hearing aids and it can minimize cosmetic and power consumption problem. [This work was supported by grand No. 10031779 from the Strategic Tech-

nology Development Program of Ministry of Knowledge Economy and this work was supported by the Brain Korea 21 Project.]

B28

Derivation of the NAL-NL2 prescription procedure

Gitte Keidser, Harvey Dillon, Teresa Ching, Matthew Flax and Scott Brewer
National Acoustic Laboratories and the Hearing CRC

The first procedure from the National Acoustic Laboratories (NAL) for prescribing nonlinear gain (NAL-NL1) was introduced in 1999. Consistent with procedures developed at NAL for prescribing linear amplification, the aim of NAL-NL1was to maximize speech intelligibility of speech while ensuring that overall loudness of speech did not exceed normal overall loudness. The formula was derived from optimum gain calculated for more than 200 different audiogram configurations and 11 levels of speech, using a modified version of the speech intelligibility index (SII) formula and a loudness model by Moore and Glasberg (1997). Based on empirical data collected during the past decade, a revised version of NAL-NL1, known as NAL-NL2, has just been released.

NAL-NL2 is derived in a similar manner to NAL-NL1 using new models for both predicting speech intelligibility and estimating loudness. The new modified SII formula was based on filtered speech data measured in quiet and in noise on 75 adults with varied degrees of hearing loss. Due to the new effective audibility data arising from this study, NAL-NL2 prescribes a gain-frequency response with relatively more gain across low and high frequencies and less gain across mid frequencies than NAL-NL1. An updated model from Moore and Glasberg (2004) was used to calculate loudness. In addition, based on empirical data, some constraints were applied to the selected gain during the optimisation procedure and adjustments were made to the formula extracted from this process. As a result, NAL-NL2 prescribes gain depending on age (child vs adult), gender (female vs male), hearing aid experience (new vs experienced), aid configuration (unilateral vs bilateral), and language (tonal vs non-tonal). Different compression parameters are further prescribed for hearing aid users with severe or profound hearing loss depending on the speed of the chosen compressor; lower compression ratios are prescribed for faster compressors. This paper details the derivation of NAL-NL2 formula and presents the data behind the adjustments made to the prescription formula resulting from the optimization process.

References:
Moore, B. C. J., and Glasberg, B. R. 1997. A model of loudness perception applied to cochlear hearing loss. Auditory Neuroscience 3:289-311.
Moore, B. C., and Glasberg, B. R. 2004. A revised model of loudness perception applied to cochlear hearing loss. Hearing Research 188: 70-88.

B29

Design of multi-channel WDRC and FBC for improvement in performance of hearing compensation in digital hearing aids

Hyundeok Kang, Yuyong Jeon, Youngrok Song, Sangmin Lee, Department of Electronic Engineering, Inha University, Korea

Digital hearing aids can perform many functions because of digital technology. Among them, WDRC (wide dynamic range compression) and FBC (feedback cancellation) are functions that compensate effectively for the loss of a person's ability to hear.

In this study, we proposed a method of design for multi-channel WDRC and FBC using Simulink. We took into account the efficiency of the multi-channel WDRC's capability to amplify at each channel and the spectral contrast when we designed a 4 channel WDRC that uses filterbanks. The 4 channel WDRC consists of channel separation, level measurement, gain application, and transformation between the time and frequency domain using the WOLA filterbank. Channels 1~4 have different frequency bands of 0~2kHz, 2~4kHz, 4~6kHz, and 6~8kHz, respectively.

We used two tests to verify the 4 channel WDRC. First, we tested to see the 4 channels separated at each of the frequency bands. We measured the output signal at the separation block when we inputted the signal of mixed pure tones 1, 3, 5, and 7kHz. The results show that the output signals at channel 1~4 are 1, 3, 5, and 7kHz, respectively. The second was to verify the gain application at each channel. We set the gain differently at every channel and measured the input and output signal. The results show that the output signal was different according to the amount of gain applied to each channel.

FBC consists of a feedback path and adaptive feedback cancellation. Adaptive Feedback cancellation uses the NLMS algorithm. We evaluated the performance through experiments of channel separation and gain control at each channel. The results of those experiments show that the channel separation and gain control at each channel were successful. We tested the FBC system by analyzing the waveforms, frequencies, and spectrogram before and after the feedback cancellation and then confirmed that the feedback was cancelled. [This work was sponsored by ETRI System Semiconductor Industry Promotion Center, Human Resource Development Project for SoC Convergence.]

B30

Study of an acoustic pipe for a wideband implantable microphone

Eui-Sung Jung, Ki-Woong Seong, Jang-Woo Lee, Jyung-Hyun Lee, and Jin-Ho Cho, Kyungpook National University in Republic of Korea

An implantable microphone used in a fully-implantable hearing device is the assembly to be composed of the diaphragm, the housing case and the acoustic transducer. A implantable microphone assembly requires that its' case and diaphragm be the biocompatibility and the hermetic seal. However one of the important problems encountered in an implantable microphone assembly, is the design of a wideband microphone. Since the microphone implanted to the human skin and tissue, the amplitude of the sound wave is attenuated by absorption and scattering. In the frequency domain, it shows poorer frequency response in the acoustic high-frequency band than it in the low- to mid- frequency band.

The objective of this study is to compensate the reduced high-frequency sensitivity of the microphone assembly after the implantation under the skin and tissue. This problem can be overcome by applying a resonance effect of acoustic pipe between the diaphragm mounted on the case and the acoustic transducer in the case. The simple model of the acoustic cannel is similar to it of the closed pipe at one-side end. So the acoustic pipe produces resonant wave at $\lambda/4$, $3\lambda/4$ and so on, and act to amplify the high frequency of speech. For the evaluation of the designed microphone assembly, we measured the sensitivity of several microphones in the water and on air using the stimulus of pure tones in 0.1 - 10 kHz. The sensitivity of implemented microphone assembles without an acoustic pipe is about -30 dB (re 0 dB = 1 V/Pa) on air at low- to mid-frequency and has the resonance peak at the vicinity of 3 kHz. And the resonance frequency of microphones in the 5 mm-depth water moved to approximately 1-1.5 kHz and frequency response showed the characteristic of 2nd order low pass filter. And, the implemented microphone assembly with acoustic pipe has the resonance peak at about 6-8 kHz. Through several experiments in the water, it has been verified that the transfer function in the high frequency band of the implantable microphone assembly was improved by the acoustic pipe. [This work was supported by the Ministry of Health & Welfare (A092106), and by MEST (No. 2010-0010570)] in KOREA.

B31

A comparison and contrast of the NAL-NL1 and NAL-NL2 prescriptive methods

Earl Johnson
US Department of Veterans Affairs

The Australian National Acoustic Laboratories (NAL) has recently released the NAL-NL2 prescriptive method for hearing aid fittings following a decade of empirical and clinical research observations with the previous NAL-NL1 method. While there are several differences between the NAL-NL1 and NAL-NL2 methods, this study focuses on prescribed frequency response for soft and medium input levels for four audiograms of varying hearing loss severities above 2 kHz (i.e., mild, moderate, severe, and profound). Thus, the purpose is to demonstrate differences in prescribed frequency response between the two methods and relate these differences, when present, to calculated speech intelligibility and loudness.

Given the presence of hearing loss above 2 kHz only in these study examples, it is of first importance to point out a major difference in the low-pass bandwidth characteristics of the two prescriptive methods. The NAL-NL1 method generally prescribes amplification through 6 kHz whereas NAL-NL2 prescribes amplification through 10 kHz. Within these respective low-pass frequencies, both methods place the RMS levels of the long-term average speech spectrum near audible levels above 2 kHz for hearing loss severity of a moderate degree. In general, for less high-frequency hearing loss more audibility is prescribed; while worse hearing loss levels are prescribed less or no audibility. The averaged difference in prescribed frequency response across the four audiograms indicated significantly less gain (5 dB) in the mid frequencies (.630-2 kHz) and significantly more gain (13 dB) in the high frequencies (5-8 kHz) for NAL-NL2 relative to NAL-NL1 ($p < 0.05$).

Using one-third octave levels of the prescribed frequency responses, both the speech intelligibility index (SII) and loudness were calculated. A NAL-modified SII, which uses a greater desensitization factor than ANSI S3.5 (1997), was used. To determine loudness, the Moore and Glasberg (2004) model was utilized. Results indicated that for hearing loss magnitudes of up to a moderate degree, differences in prescribed frequency response yielded approximately equivalent loudness and higher SII values for NAL-NL2 re: NAL-NL1. For severe hearing loss or worse, differences in prescribed frequency response yielded an approximately equivalent SII and less loudness for the NAL-NL2 method in comparison to the NAL-NL1 method. These findings are the result of NAL-NL2 prescriptive design which extends intelligible bandwidth for hearing loss levels up to a moderate degree and lessens prescribed loudness for hearing loss of a severe degree or worse.

B32

Predicting effects of impaired cochlear processing on consonant discrimination in stationary noise

Morten L. Jepsen[1], Torsten Dau[1] and Oded Ghitza[2]
1 Centre for Applied Hearing Research
2 Boston University

Cochlear hearing loss is typically associated with reduced sensitivity due to inner hair-cell (IHC) and outer hair-cell (OHC) dysfunction. OHC dysfunction also leads to supra-threshold deficits, such as reduced basilar-membrane (BM) compression as well as reduced frequency selectivity and temporal resolution. Listeners with a cochlear damage typically have difficulties with speech understanding in the presence of background noise. In this study, the goal was to investigate the relation between individual consonant confusions in stationary noise and deficits in cochlear signal-processing as characterized by the audiogram and estimates of the BM input-output characteristics. Cochlear processing in individual listeners was simulated using a computational model of auditory signal processing and perception (CASP) and was used as a front end in a consonant discrimination system. Individual error patterns from a Diagnostic Rhyme Test (DRT) were measured and analyzed in terms of acoustic-phonetic features. This was done for three listeners with cochlear hearing loss and at two signal-to-noise ratios. It is shown that the predicted errors patterns matched the measured patterns in most conditions. Thus, an incomplete representation of the speech sounds due to deficits in cochlear processing could be related to the performance in the speech perception task. In addition, it was studied to what extent the data could be accounted for based on reduced sensitivity only – assuming that BM compression, frequency selectivity and temporal resolution are the same as in normal-hearing listeners. For two out of the three listeners, the supra-threshold deficits needed to be included in order to account for the data, while for the third listener, the predicted error rates were similar for the two model versions. Overall, the results suggest a clear relation between deficits in cochlear signal processing and consonant identification error patterns, and indicate that the supra-threshold deficits associated with a cochlear damage need to be taken into account. The findings might be interesting for applications, such as the evaluation of hearing-instrument signal processing, where the effects of specific processing strategies can be simulated for individual hearing losses.

B33

A speech enhancement method using on MARA and speech reinforcement algorithm based on partial masking effect

Yuyong Jeon, Young Rok Song and Sangmin Lee, Department of Electronic Engineering, Inha University, Korea

To enhance the quality of the perceived speech on the present of the background noise, acoustic masking that is one of the most well-known property of human auditory have been used in many studies. In this study, we proposed an algorithm to enhance the perceived quality of speech signal masked partially by background noise. Proposed algorithm is consisted by two algorithms; MCRA (minima controlled recursive averaging) algorithm, speech reinforcement algorithm.

MCRA algorithm calculates how much noise is contained in the input signal based on the ratio between the input spectrum and the local minimum value of the input spectrum. If the ratio is larger than threshold, speech presence indicator is 1. This parameter also will be used in the speech reinforcement part as a criterion for speech presence. And then estimated noise and speech power by MCRA algorithm is used for speech reinforcement. In the speech reinforcement algorithm, absolute hearing threshold from standard equal loudness contour (ISO226) and loudness perception model proposed by Moore et al was used for modeling the human hearing properties. Excitation level of noise and speech was calculated in each band, and if the speech level was larger than threshold, specific loudness and partial specific loudness of the speech was calculated from excitation level of speech and noise respectively. Then, comparing the specific loudness and partial specific loudness, gain for reducing partial masking effect was computed. However if the speech level was smaller than threshold, gain was 1. This gain can make the partial specific loudness to specific loudness.

To confirm the effectiveness of proposed algorithm, we compared the noisy speech (NS), noise reduced speech by spectral subtraction based on MCRA (NR), and speech processed by proposed algorithm (PM). 3 types of noise (white, babble, car interior) were mixed with speech by -5dB, 0dB or 5dB SNR. In view of the results, although increment of segSNR and PESQ score in condition of PM than other conditions is smaller in babble noise, all of segSNR and PESQ score in the condition of PM were increased than that in the condition of SS or NS. [This work was supported by the grant from the Korea Health 21 R&D Project, Ministry of Health & Welfare, Korea (A091039)]

B34

Evaluation of three types of speech-in-noise tests in hearing-impaired listeners from Belgium and France

[1]Jansen S., [1]Luts H., [1]van Wieringen A., [2]Del Rio M., [2]Dauman, R., [3]Lamy A., [3]James C., [3]Fraysse B., [4]Vormès E., [4]Frachet B., [1]Wouters J.

1. *ExpORL, Dept. Neurosciences, K.U.Leuven, Belgium*
2. *Audiology Department, Hôpital Pellegrin, CHU, Bordeaux, France*
3. *Service d'ORL, Hôpital Purpan, Toulouse, France*
4. *Hôpital Avicenne, Assistance Publique Hôpitaux de Paris, Université Paris XIII, France*

Recently, we developed three types of speech-in-noise tests for Francophone listeners: an open-set (FIST) and closed-set (FrMatrix) sentence test, and a digit triplet screening test (FrDigit3). Compared to the 'gold standard' FIST, the automatic scoring procedure of FrMatrix is very advantageous. Moreover, FrMatrix is not liable to memorization. FrDigit3 is a fully automatic test as well, is very short, and requires less cognitive capability, making it perfectly suitable for screening of large populations. Thorough evaluation in normal-hearing subjects showed steep slopes of the intelligibility function and the adaptive procedures proved to reliably measure the speech reception threshold (SRT). The aim of the current multi-center study is (1) to validate and mutually compare these tests in mildly-to-moderately hearing-impaired persons, and (2) to compare SRTs of Belgian (Brussels), North-French (Paris), and South-French (Bordeaux-Toulouse) subjects. In the end, we want to be able to use these tests to reliably screen for hearing loss and to precisely assess benefits from speech enhancement strategies in hearing instruments.

Twenty normal-hearing and 80 hearing-impaired subjects were tested under headphones (monaurally), in test and retest. The speech signals were presented at a fixed (individually determined) level, and both stationary speech-weighted noises and the fluctuating ICRA-4-250 noise were used as interferers.

The normal-hearing results demonstrate that reliability deteriorates when a more realistic fluctuating noise is used compared to a stationary. Nevertheless, the root-mean-square of the within-subject standard deviations was still only 1.4 dB for FrMatrix in ICRA-4-250. This will still be sufficient to reliably determine hearing aid benefit in one condition over another.

Based on the correlation between SRT's for FrMatrix and FIST, we will investigate whether a matrix test can be an alternative to realistically evaluate intelligibility performance in a specific listening situation. The automatic scoring procedure (which can easily be implemented on an Internet test platform) and the possibility to reuse test lists within the same subject, would then provide a major advantage in hearing aid research.

B35

A generic noise environment for evaluating adaptive noise reduction algorithms

Adam Hersbach, Cochlear Ltd.

The evaluation of noise reduction algorithms through the use of objective speech intelligibility tests is highly dependant on the acoustic situation created in the controlled environment of a sound booth. The noise location and/or noise type is often chosen to demonstrate the benefits of a particular algorithm, but can lead to tests which have limited real-world relevance, or that do not reflect the benefit reported by users in the field.

The proposed test uses eight fixed loudspeakers in a generic sound room. The reverberation and room acoustics are determined by the room itself rather then introduced room impulse responses. The noise is generated such that individual noise makers are both separated in space and change location during the course of the test. For example, four talker babble noise is generated by separating the four individual maskers so each can be presented on any one of seven loudspeakers arranged in the rear hemisphere. During the test, at intervals of three seconds, the location of one of the maskers is changed and presented from a new loudspeaker position chosen at random. The target speech is always presented from zero degrees.

The noise has been designed to improve on traditional tests where the locations of noise maskers are fixed, or are presented from a single location. The proposed noise represents a range of real life environments where interfering talkers are neither stationary nor co-located. By distributing the interfering maskers at different locations that change during the test, the noise reduction algorithm under evaluation is forced to adapt to the varying environment. The test is suitable for the evaluation of a wide variety of noise reduction algorithms, including multi-microphone and binaural algorithms.

To evaluate test re-test reliability, the noise was used in an adaptive speech reception threshold measurement with a group of 14 cochlear implant recipients. Three different roving location spatially separated noise types were evaluated using speech weighted noise, 4 masker babble, and 20 masker babble. The test re-test variability was shown to be less than 1.2dB typically required of an SRT test of this type. Three different speech processing conditions were also tested, demonstrating a progressive benefit of a standard microphone, a fixed directional microphone and an adaptive beamformer. The observed improvements were further corroborated by a questionnaire that was administered after a period of take home experience with each algorithm, supporting the real-world relevance of the test.

B36

A psychoacoustic model for predicting sound quality (MCHI-S)

Jörg Haubold, Lorenz Goenner
ciAD – competence in AudioDesign, D-44287 Dortmund, Witthausstr. 42, Germany

An individually successful hearing aid fitting is characterized by a comfortable sound impression of the hearing instrument beside improved speech intelligibility in everyday life situations. Every hearing aid manufacturer endeavors to achieve this objective. In doing so, many effort and costs have to be spent for field studies with subjects. A prediction of important auditory dimensions affected by hearing instrument signal processing could support both, reducing the effort and speeding up the development process.

A prediction of the acoustic quality of hearing aids and hearing aid algorithms must comprise important quality aspects for hearing impaired persons, such as loudness, timbre, auditory comfort, listening effort and speech intelligibility. For normal hearing people, such models are already available. We will show that the MCHI-S model enhances the sound quality prediction for hearing impaired persons. It allows to predict scaled ratings of the above mentioned quality aspects for hearing impaired persons with and without hearing aids and for natural sounds. On the basis of the processed signal, the original sound and the audiological data of a patient, the quality aspects of loudness, timbre, auditory comfort, listening effort and speech intelligibility can be predicted without a direct measure with the subject.

The presentation will demonstrate the fundamentals of the MCHI-S model as well as results of validation studies.

B37

The neuro-compensator: A hearing compensation algorithm based on cochlear modeling and machine learning

Becker S[1], Pango P[2], De Laat JAPM[3]
1 *McMaster University of Toronto, Canada*
2 *VitaSound Audio, Ontario, Canada*
3 *Leiden University Medical Center, Netherlands*

In this presentation, we will describe the design and present testing results of a novel hearing aid based on the Neuro-Compensator algorithm. The Neuro-Compensator employs neurocomputational models of the pathophysiology of hearing loss to optimize the gain calculation in a hearing aid so as to generate a more normal pattern of neuronal firing in the auditory nerve of the hearing impaired listener.

The Neuro-Compensator employs sophisticated computer models of the cochlea to determine the optimal gain calculation for a given hearing impaired listener. The auditory models are used to simulate how the auditory nerve of a hearing impaired person would respond to a given level of amplification across all frequency channels. The resultant auditory neuronal firing pattern for the impaired ear model is compared to that of a healthy ear model. Machine learning methods are used to iteratively adjust the gain calculation performed by the Neuro-Compensator, until a normal pattern of neuronal activity is restored. Once this training procedure is completed, the Neuro-Compensator no longer requires the complex auditory model, and the final gain calculation can then be incorporated into a conventional hearing aid microprocessor. In the resulting hearing aid, rather than calculating the gain separately within each frequency channel, the gain of each channel is dynamically calculated as a function of the entire spectral content of the signal. The Neuro-Compensator thereby has the ability to restore the non-linearities and cross-channel modulatory functions normally achieved by the outer hair cells. Computer simulations indicate that compared to the widely used WDRC gain calculation, the Neuro-Compensator is better able to restore intelligibility of higher frequency components of speech signals. Preliminary subjective reports from hearing-impaired individuals indicate that compared to conventional hearing aids the Neuro-Compensator-based hearing aid restores a much more natural sound for both speech and music, while sound perception in noise and sound localization are greatly enhanced.

B38

The role of temporal fine structure in sound quality perception

Melinda C. Anderson[1], Kathryn H. Arehart[1] and James M. Kates[1,2]
1 *University of Colorado, Boulder*
2 *GN ReSound*

Hearing loss can affect the ability of listeners to process important components of sound. Some of these components relate to the temporal structure of sound, and include slowly varying changes (temporal envelope, TE) and much faster-changing components (temporal fine structure, TFS). This study focuses on one dimension (temporal structure) that underlies sound quality judgments for both listeners with hearing loss and younger and older adults with normal hearing.

We have two research aims. The first aim is to quantify the role of TFS in sound quality perception. The second aim is to determine if the role of TFS in sound quality perception differs for listeners with normal hearing and listeners with hearing loss. As in studies examining the role of TFS in speech intelligibility (e.g. Hopkins et al., *J. Acoust. Soc. Am*, 123, 1140-1153), this study parametrically varied the amount of TFS in different frequency regions.

The results indicate that removal of TFS from regions above 4100 Hz for speech in quiet has no effect of sound quality perception for both listeners with normal hearing and listeners with hearing loss. Removal of TFS from speech in quiet below 4100 Hz has a negative impact on sound quality perception, with listeners with normal hearing showing greater reductions in sound quality ratings compared to listeners with hearing loss. When speech is in the presence of background noise (18 dB SNR and 12 dB SNR), removal of TFS results in smaller, but still measurable, reductions in sound quality ratings for both groups of listeners. These reductions are similar in scale between groups, in contrast to the scale of the reductions in quality ratings in quiet.

The results have implications for a) the perceptual tolerance for hearing aid signal processing strategies that affects TFS and b) for modeling hearing aid sound quality in terms of the relative importance of TE and TFS cues in listeners with normal hearing and with hearing loss. [Work supported by the Institute of Cognitive Science at the University of Colorado, Boulder and a research grant to the University of Colorado from GN ReSound]

Posters for Session C should be put up by 8 A.M. Saturday, August 14, and taken down after 10 PM Saturday, August 14 or before 7 AM Sunday, August 15. Presenters should be at their posters from 9:45 AM – 11:10 AM; 4:30 - 5:00 PM

POSTER SESSION C

Saturday 9:45 AM to 11:10 AM

C1

Development of advanced companding strategy by using classification system for digital hearing aid users

Yoon Su Hwang[1], Jonghee Han[1], Jin Ryoul Kim[1], Kyeongwon Cho[1], Sung Hwa Hong[2], Sang Min Lee[3], Dong Wook Kim[4], Sun I. Kim[1] and In Young Kim[1]

1 *Department of Biomedical Engineering, Hanyang University*
2 *Department of ORL-HNS, Sungkyunkwan University*
3 *Department of Electronic Engineering, Inha University*
4 *Bio & Health Lab, Samsung Advanced Institute of Technology*

Persons with sensorineural hearing losses are unable to fully utilize the temporal and spectral cues because they have lower temporal and spectral resolution than normal hearing listeners. Therefore, spectral enhancement algorithms have been studied for digital hearing aid users. One of spectral enhancement algorithms is companding, which can control spectral contrast by several parameters. However, as the contrast becomes stronger, the speech signal gets more distorted. In order to achieve spectral enhancement with less distortion, we applied different companding strategies for frequency regions corresponding to formants and the others.

In this study we suggests advanced companding algorithm combining classification system. Proposed algorithm consists of classification system, formant detection, and companding. Input speech signal was divided into frames by a length of 960 samples and windowed by the Hamming window. By analyzing the spectrum of each frame, it was classified into silence, voiced or unvoiced speech. For the voiced frame, formant detection was performed using 18 orders linear predictive coding (LPC) and formant frequencies (F1, F2 and F3) were estimated. A companding algorithm using 50 channels was used and parameters used in the algorithm were selected as n1=0.3, n2=1, q1=2.8. q2=4.5, 6, 9. For the frequency band including formants, q2=6, 9, 12 was used while in the other bands q2=4.5, 6, 9 was used respectively. When silence was detected, companding was not used. We simulated the algorithm with 5 test speech material IEEE sentences and compared it to the conventional companding algorithm.

We observed the spectrum of the conventional companding and proposed algorithm. Consequently, we found out formant peaks are more sharpen and less distorted in the formant frequencies using proposed algorithm. [This work was supported by grand No. 10031741 from the Strategic Technology Development Program of Ministry of Knowledge Economy].

C2

The Psychoacoustic Effects of Release Time on Automatic Gain Control

Se Mi Hwang, Yu Yong Jeon, Young Rok Song, Heon Jin Park, Sang Min Lee
Department of Electronic Engineering, Inha University, Korea

Automatic gain control (AGC) is used in hearing aids to compensate for the hearing level due to reducing dynamic range. Release time, one of components of automatic gain control, can have an effect on speech intelligibility. In this study, we focused on individual optimum release time and performed speech discrimination score test with normal hearing listeners and hearing impaired listeners. Two groups consisted of twelve normal hearing listeners and twelve hearing impaired listeners participated in the test. To compare recognition score in various speech situations, we conducted the test under quiet and noise conditions according to variable release time.

One syllable speeches and sentences from Korean hearing in noise test (KHINT) were recorded by same energy. And then, noise is mixed to sentences by 12dB, 6dB, 0dB signals to noise ratio.

Attack time was fixed at 4ms and the release time was 12, 64, 128, 512, 2094, or 4096ms to look into individual psychoacoustic effects. And then, we conducted behavior test through speech recognition tests. In speech discrimination score test, each speech was played at 65dB SPL for normal hearing listeners and most comfortable level for hearing impaired listeners. All tests were performed in twice per person.

All experiments were analyzed using a statistical analysis system (SAS) software. Analysis of variance (ANOVA) was used to compare with scores in normal hearing and hearing impaired group. In addition, frequency analysis used to know whether each of the people in twice experiments had different score or not. As a result, each group had optimum recognition scores at 4096, 2094, 12ms release time in KHINT in 12, 6, 0dB SNR. In case of one syllable test, however, normal hearing group had optimum recognition scores at 4096ms release time while hearing impaired group recorded at 4096ms. Furthermore, the second optimum recognition score of the listeners about 92% ($p < 0.05$) accord closely with the first optimum recognition score. In adjusting release parameter of the hearing aid, in conclusion, each of the individuals will recommend optimum release time setting. Thus, most people can improve to recognize and discriminate speech words. [This work was supported by grant No. 10031764 from the Strategic Technology Development Program of Ministry of Knowledge Economy].

C3

Individual Differences in Auditory Temporal Processing among Middle-Aged and Older Adults

Larry Humes, Diane Kewley-Port, Thomas Busey and James Craig
Indiana University

This paper presents data on individual differences in auditory temporal processing among 186 middle-aged and older adults. The study sample included 40 adults 40-55 years of age (M=48 yrs., 27 females, 13 males), 66 adults 60-69 years of age (M=65 yrs., 34 females, 32 males), 60 adults 70-79 years of age (M=74 yrs., 37 females, 23 males), and 20 adults 80-89 years of age (M=82 yrs., 10 females, 10 males). Measures of auditory temporal processing included conventional measures of gap detection threshold at 1000 and 3000 Hz, as well as twelve measures of auditory temporal-order identification for vowel sequences and auditory temporal masking for vowel identification. All measurements were obtained using rigorous psychophysical methods and multiple blocks of trials in order to get reliable measures of auditory temporal-processing performance. Vowel stimuli were low-pass filtered at 1800 Hz and presented at 85 dB SPL to minimize the influence of high-frequency hearing loss on performance. In addition to these measurements in hearing, parallel measures were obtained from all participants and for all tasks in the visual and tactile modalities. Among other things, this allowed for assessment of the modality specificity of auditory temporal processing in middle-aged and older adults.

Since the focus here is on individual differences in performance, the data for the 186 participants were pooled into one large dataset and initial correlational analyses were performed. Several patterns of correlations were observed and the data were subsequently analyzed using principal-components factor analysis. Six oblique (correlated) factors emerged from this analysis, accounting for 70% of the variance. For the temporal –order and temporal-masking tasks, very little overlap was observed across modalities in middle-aged and older adults. That is, auditory temporal-order processing was largely independent of similar processing in the tactile or visual senses. An exception was observed for temporal-order tasks in which only the location of the stimulus pair (e.g., right-left vs. left-right) was required rather than the identification of the actual stimuli comprising the pair (e.g., ah-eh). Performance on this task was amodal; that is, there were strong correlations for the performance across the auditory, tactile and visual versions of this task. Cross-modal correlations were also observed for the auditory, tactile and visual gap-detection measures. The role of chronological age and cognitive function on individual differences in temporal-processing performance among middle-aged and older adults will also be discussed. [Work supported, in part, by NIA R01 AG022334.]

C4

The effects of age and cochlear hearing loss on temporal fine structure sensitivity, frequency selectivity and speech reception in noise

Kathryn Hopkins, University of Cambridge and University of Manchester, and Brian C. J. Moore, University of Cambridge

Listeners with cochlear hearing loss are poorer than normal-hearing listeners at tasks thought to rely on sensitivity to temporal fine structure (TFS) information, such as pure-tone interaural phase discrimination, low-rate frequency modulation detection, and the discrimination of band-pass filtered harmonic and frequency-shifted tones. However, the mechanism underlying the reduced sensitivity to TFS among hearing-impaired subjects remains unclear, and it is not known whether age affects TFS sensitivity in the absence of an elevation in audiometric thresholds.

Sensitivity to TFS, frequency selectivity, and speech reception in noise were measured for young normal-hearing, old normal-hearing and hearing-impaired subjects. Two measures of TFS sensitivity were used: the 'TFS-LF test' (interaural phase discrimination) and the 'TFS2 test' (discrimination of harmonic and frequency-shifted tones). Performance on the TFS2 test was very poor for the hearing-impaired subjects, even for subjects with a mild hearing loss whose audiometric thresholds were within the normal range at the test frequencies. Measures of TFS sensitivity and frequency selectivity were not significantly correlated (controlling for the effect of audiometric threshold), suggesting that insensitivity to TFS does not directly result from a broadening of auditory filters.

The results of the two tests of TFS sensitivity were significantly but only modestly correlated, suggesting that performance in the two tests may not solely be limited by the same factor. For example, performance for TFS-LF test might be limited by central deficits in binaural integration, whereas performance for the TFS2 test might be limited by deficits in the coding of TFS information in the peripheral auditory system.

The old normal-hearing subjects performed more poorly than the young normal-hearing subjects for both measures of TFS sensitivity, but not frequency selectivity, suggesting that TFS sensitivity declines with age in the absence of elevated audiometric thresholds or broadened auditory filters. Peripheral or central mechanisms could account for this reduction in TFS sensitivity with age.

When the effect of mean audiometric threshold was partialed out, speech reception thresholds in modulated noise were correlated with TFS2 test scores, but not frequency selectivity measurements. This finding is consistent with the idea that TFS information is particularly important when listening in the dips of an amplitude-modulated background. [This work was supported by the Oticon Foundation and the Medical Research Council, UK].

C5

Percentile Analysis of different hearing instruments in First-Fit-Setting

Inga Holube[1], Alexandra Winkler[1] and Matthias Latzel[2]

[1] *Jade University of Applied Sciences, Institute of Hearing Technology and Audiology, Oldenburg, Germany,*

[2] *Siemens Audiological Engineering Group, Erlangen, Germany*

A draft for a new standard for measuring hearing instruments (IEC 60118-15) is internationally being approved and voted. This standard includes an International Speech Test Signal (ISTS), an analysis procedure and representative audiograms. The analysis procedure is based on the percentile analysis of the short-time level distribution of the input and the output signal of the hearing instrument. From the comparison of the output to the input signal, the gain of the hearing instrument for speech can be derived. The new standard was applied to five hearing instruments of different manufacturers. Instead of using the representative audiograms, the hearing instruments were fitted to 15 different subjects with a moderate hearing loss. The fittings were based to the recommendations of the manufacturers ("First Fit"), i.e. fine tuning was not applied. The hearing instruments were exposed to the ISTS, a German conversation in quiet, a stationary speech-shaped noise, and white noise in a measurement box using an ear simulator according to

IEC 711. In addition to the gain, the percentile distribution of the output of the hearing instruments was compared to the individual pure tone audiogram and the effective compression for speech was analyzed. The results demonstrate the different fitting approaches of the manufacturers in terms of output level in relation to hearing threshold. On the other hand, the manufacturers seem to agree on low compression for the speech signal itself when using a constant input level. This can be achieved e.g. by low compression ratios or long time constants or both. Further results of the same study including subjective ratings of the hearing instruments by the subjects are shown by Latzel et al. at this conference.

C6

Creating and Validating a Danish Acceptable Noise Level Test

Marcus Holmberg[1], Jens Henrik Andersen[2], Christian Stender Simonsen[1] and Thomas Lunner[3]

1 *Oticon A/S*
2 *University of Copenhagen, INSS*
3 *Oticon A/S Research Centre Eriksholm*

The Acceptable Noise Level (ANL) test was suggested by Nabelek et al. (JSHR 1991;34:679-685), and has been shown to predict hearing aid usage. We have created a Danish version, with minor modifications to the procedure to ensure comparison with the original ANL test, and validated this new test on normal hearing and hearing impaired listeners.

The speech material of the Danish ANL is based on an existing high quality audio book with a female speaker. A total of about 35 minutes of speech was selected according to both content and technical analysis. Random segments of one minute length are presented to avoid that the test person gets too involved in the story. The background noise is a 12-talker babble noise made from the speech material of the test, to ensure that the long-term spectrum of the speech signal and the babble background is the same.

The ANL procedure is implemented in a Matlab program, controlled by the test person via a touch screen. Procedure is as follows: The most comfortable level (MCL) of the speech signal is measured as in the original ANL procedure, but carried out three times in succession. The median of the three MCL measurements is used. In the same manner, the background noise level (BNL) is measured as in the original ANL procedure, but again three times. The ANL is calculated as the median of the three MCL measurements minus the median of the three BNL measurements.

Validation of the test was carried out with 14 normal hearing and 20 hearing impaired test persons. The hearing impaired test persons wore linear amplification hearing aids for this study. We investigated within-visit and between-visit effects. There were two visits, with either 2 or 4 weeks between them. Each visit included one training run with oral instructions and two test runs with instructions on the touch screen. The results show that the Danish ANL test gives about the same scores as the American ANL test for young normal hearing listeners and hearing impaired listeners that are full-time users of hearing aids. The within-visit test-retest standard deviation for the hearing impaired group was 1.55dB. We can also conclude that the training round does not differ from following test runs, and thus can be used as a valid measurement.

C7

Reproduction of the Performance/Intensity Function using image processing and a computational model

A. Hines and N. Harte, Department of Electronic and Electrical Engineering, Sigmedia Group, Trinity College Dublin, Ireland

It has been shown that auditory-nerve discharge patterns in response to sounds as complex as speech can be accurately modelled. It is predicted that this knowledge could be used to test new strategies for hearing-aid signal processing (Sachs et al, 2002). Auditory-nerve representations of phonemes in normal and noise-damaged ears can be assessed via subjective visual inspection to judge how the impaired representations differ from the normal. This work seeks to automate this inspection process and rank hearing losses based on auditory-nerve discharge patterns. The aim of this study is to establish the optimal weighted components of a mean structural similarity index measure (MSSIM) for comparing human listener performance tests with simulated auditory nerve (AN) model tests. The

AN model produces simulated auditory nerve neural spike train outputs at specific characteristic frequencies (CF). These can be assessed by examination of the spectro-temporal output visualised as neurograms.

MSSIM is an objective measure originally developed to assess perceptual image quality (Wang et al, 2004). The measure is adapted here for use in measuring the phoneme recognition in neurograms derived from simulated AN outputs. The MSSIM metric is composed of three weighted components: luminance, contrast and structure. The optimal weightings of these components is assessed with neurograms produced using experimental data and reported results from clinical human listener tests.

The performance versus intensity (PI) function test describes recognition probability as a function of average speech amplitude, showing the cumulative distribution of useful speech information across the amplitude domain as speech rises from inaudibility to full audibility (Boothroyd, 2008). The test set contains word groups of 10 phonemically balanced CVC words. Listener performance in these tests was modelled using an AN model (Zilany, 2009) and the resultant neurogram outputs were analysed with MSSIM following the methodology outlined in prior work (Hines and Harte, 2010).

Ten groups were used for data fitting to establish the optimal SSIM weighting values. A further ten groups were used to assess the test-retest consistency of the weightings. Each word was tested at 10 hearing intensities from 5 to 50 dB SPL. The recognition was tested by evaluating each phoneme and comparing neurograms from each intensity to a baseline 65 dB SPL neurogram.
The simulated results follow closely to those for human tests. For both neurogram types, the MSSIM weightings biased the luminance and structure components over contrast.

References:

Boothroyd A. 2008. The Performance/Intensity Function: An Underused Resource. Ear and Hearing, 29, 479-491.

Hines A. & Harte N. Speech Intelligibility from Image Processing. Speech Communication, In Press, Accepted Manuscript, doi:10.1016/j.specom.2010.04.006.

Rosen S. 1992. Temporal Information in Speech: Acoustic, Auditory and Linguistic Aspects. Philosophical Transactions: Biological Sciences, 336, 367-373.

Sachs M.B., Bruce I.C., Miller R.L. & Young E.D. 2002. Biological Basis of Hearing-Aid Design. Annals of Biomedical Engineering, 30, 157–168.

Wang Z., Bovik A.C., Sheikh H.R. & Simoncelli E.P. 2004. Image quality assessment: from error visibility to structural similarity. Image Processing, IEEE Transactions on, 13, 600-612.

Zilany M.S.A., Bruce I.C., Nelson P.C. & Carney L.H. 2009. A phenomenological model of the synapse between the inner hair cell and auditory nerve: Long-term adaptation with power-law dynamics. The Journal of the Acoustical Society of America, 126, 2390-2412.

C8

Noise reduction delivers speech intelligibility benefit for cochlear implant users

Adam Hersbach[1], Stefan Mauger[1], Komal Arora[1,2], Pam Dawson[1,2]
1 Cochlear Ltd
2 CRC Hear

AIM: To test if an experimental noise reduction algorithm for cochlear implant speech processing provides speech intelligibility benefit upon state of the art directional microphone processing.

An experimental single channel noise reduction algorithm has been developed for cochlear implant (CI) speech processing. It was evaluated as a pre-processing strategy on its own, in conjunction with a fixed directional microphone and in conjunction with an adaptive beamformer. Speech intelligibility improvement was measured in a sound booth using a speech in noise test where the locations of the noise maskers were both separated in space and also changed position during the course of the test. After a period of take home use, experience using each of the algorithms was further assessed via a questionnaire.

The experimental noise reduction algorithm operated by removing frequency channels with poor signal-to-noise ratio (SNR). The real-time algorithm estimated the noise in each frequency channel using a minimum statistics recursive smoothing algorithm followed by an SNR estimation stage. The signal level in each frequency channel was then attenuated using an SNR dependant gain curve. Those channels with poor SNR were attenuated whilst those with good SNR were not.

Speech reception thresholds (SRTs) were measured for three different roving location spatially separated noise types, using speech weighted noise, 4 masker babble and 20 masker babble. The experimental noise reduction algorithm was tested as a pre-processing algorithm using a standard microphone, and then in combination with a fixed directional microphone and an adaptive beamformer. SRT results across the group of 14 experienced CI recipients demonstrated a statistically significant progressive improvement from standard microphone, to fixed directional, to adaptive beamformer. In each case the inclusion of the noise reduction algorithm further improved the SRT, providing additional benefit over and above the directional algorithm alone.

These results are particularly encouraging given that single channel noise reduction algorithms of this class, when applied to systems with an acoustic output such as hearing aids, usually fail to demonstrate an improvement in speech intelligibility as has been shown here for a group of cochlear implant recipients.

The questionnaire data showed positive acceptance of the experimental noise reduction algorithm in a wide range of listening conditions, including noisy and quiet situations, supporting the speech intelligibility improvements that were observed in the study.

C9

Effect of NAL-R amplification on consonant recognition in hearing-impaired listeners

Woojae Han (Dept. of Speech and Hearing Science, Univ. of Illinois at Urbana-Champaign, 901 S. Sixth St., Champaign, IL 61820, whan5@illinois.edu), Riya Singh (Dept. of Electrical and Computer Engineering, Univ. of Illinois at Urbana-Champaign), and Jont Allen (Dept. of Electrical and Computer Engineering and Beckman Institute, Univ. of Illinois at Urbana-Champaign)

Although each hearing-impaired individual benefits differently from the use of a hearing aid, most listeners who have sensorineural hearing loss (SNHL) are dissatisfied by amplification provided by traditional hearing aids. In particular, these users have noted that some speech sounds amplified through the hearing aids are audible and understandable while others are not. Despite much research on new hearing aid technology and thus highly improved devices, it is still unclear why two individuals with a similar configuration of hearing loss reveal significant differences in their ability to understand speech. The purpose of current study is to examine how much speech perception ability in hearing-impaired listeners will be enhanced by amplification prescribed the NAL-R fitting formula. We hypothesize that there is a limited benefit of using NAL-R amplification as to no amplification. Twenty-six impaired ears having mild-to-moderate SNHL participated in the study. Sixteen English CV syllables were spoken by 18 native talkers at the most comfortable level for each impaired ear through ER-2 inserted earphone. They were presented in quiet and five different signal-to-noise ratios (+12, +6, 0, -6, -12 dB SNR) in speech-weighted noise. Calibration estimated the presentation levels in the ear canal to be between 75 and 90 dB SPL depending on the attenuator setting. The subjects were tested in two sessions: one was a no-amplification condition and the other was an amplified condition by applying the half-gain NAL-R formula. Each session included a total 1152 tokens (16 consonants × 12 presentations × 6 SNRs). To avoid a learning effect, the order of session was randomly presented and there was a gap of at least one week between the two sessions. Preliminary results showed that the recognition score decreased both with and without amplification, as the noise increased. However, compared to the no-amplification condition, only half of 16 CV syllables were improved in the amplified condition. Scores of /za/, /Da/, /ta/, and /Sa/ syllables generally improved after amplification, whereas /ma/, /pa/, and /ba/ scores deteriorated. Score of /Ta/ syllable was not changed significantly between two conditions. Although there was a high individual variance, the outcomes from this study suggest that the current fitting formula based on pure-tone audiogram and the half-gain rule does not improve overall speech recognition for hearing-impaired listeners. [This work was supported by NIH Grant].

C10

The effect of nonlinear frequency compression on consonant recognition as a function of acclimatization time

Danielle Glista, Susan Scollie and Jacob Sulkers, National Centre for Audiology, The University of Western Ontario

Hearing impairment is most commonly found in the high-frequencies, affecting perception of high-frequency sounds. Nonlinear frequency compression (NFC), a type of frequency lowering hearing aid technology, compresses the high-frequency output bandwidth of a signal by a specified ratio. Recent studies provide evidence of speech perception benefit with NFC for listeners with high-frequency hearing loss (Glista et al., 2009; Simpson, Hersbach, & McDermott, 2005, 2006; Wolfe, Caraway, John, Schafer, & Nyffeler, 2009). Studies of frequency compression hearing aids report some high-frequency speech sound confusions measured with a consonant- vowel nucleus- consonant (CNC) word test in adult listeners (Simpson et al., 2005, 2006). Similar reports can be found in the literature pertaining to frequency transposition (another type of frequency lowering hearing aid technology); consonant confusions around phonemes including, but not limited to, /s/, /ʃ/, /f/, and /t/ have been reported for adult and child listeners (Rees & Velmans, 1993; Robinson, Baer, & Moore, 2007; Robinson, Stainsby, Baer, & Moore, 2009). It has been suggested that auditory training and/or longer perceptual adaptation time is needed for listeners to overcome sound confusions introduced by frequency lowering technology (Auriemmo et al., 2009; Robinson et al., 2009).

Our area of interest is in the impact of NFC technology on speech recognition measured over time in pediatric listeners. This poster will present results on speech recognition performance in 5 children with sloping, high-frequency hearing loss. An acclimatization period to NFC processing of approximately 16 weeks was allotted to all participants. Speech recognition performance will be reported across time for the Distinctive Features Difference test (Cheesman & Jamieson, 1996). Degree of change measured over time will be analyzed according to overall performance, as well as errors and confusions associated with distinctive features. Preliminary results suggest an acclimatization effect associated with speech recognition performance; a change in performance for specific speech sound confusions can be observed over the specified time period. [This work was supported by The Canadian Institutes of Health Research and Phonak AG]

C11

Speech Referenced Limiting (SRL): a new approach to limiting the sound level from a hearing aid

Michael Fisher[1,2], Nicky Chong-White[1,2] and Harvey Dillon[1,2]

1 *The HEARing CRC, Melbourne, Australia*

2 *National Acoustic Laboratories, Chatswood, Australia*

Hearing aid users, with access to a volume control, typically set the volume so that conversational speech is at a comfortable level. When fixed at this volume setting everyday sounds such as a door slam, a telephone ring, music in a shop, an umpire’s whistle or the sound of washing up pots and pans can sometimes be too loud.

Conventional methods of sound level limiting in hearing aids normally involve setting the limit to just below the level at which loudness discomfort occurs for the listener. If this limiting level is low relative to the amplified speech level then the quality of the speech will be reduced. On the other hand if this limiting level is high relative to the amplified speech level then sounds of a higher level than speech will be amplified to levels above that of the speech making them less comfortable.

A new approach to sound level limiting is to use the level of the speech that the hearing aid user has recently or is currently hearing as a reference and reduce the level of non-speech sounds with respect to this reference. This novel method is called Speech Referenced Limiting (SRL). The limiting level is adaptive and is set by the loudness of the speech to which the hearing aid user is acclimatised. When done on a frequency specific basis the umpires whistle is reduced to the level of the treble of a recent conversation and the rumble of a truck to the level of its bass. This is achieved by estimating the loudness of speech at different frequencies to produce a speech reference and limiting sound that exceeds this reference. The method may be used in multi-band hearing aids that have a speech program. Details of the SRL scheme and experimental data on the effects of SRL on speech and noise are presented.

The authors acknowledge the financial support of the HEARing CRC, established and supported under the Australian Government's Cooperative Research Centres Program.

C12

Simulating the Effects of Stimulation Rate on Speech Recognition for Unilateral and Bilateral Cochlear Implant Modes

Kallel Fathi[1], Gallego Stéphane[2], Ghorbel Mohamed[1], Frikha Mondher[1] and Berger vachon Christian[2]
1 National Engineering school of Sfax-Tuni
2 Claude bernard university of lyon-France

A model was developed to simulate acoustically the effects of stimulation rate on speech recognition for unilateral and bilateral cochlear implant modes (CI). In our study, we consider two stimulation rates: 250pps and 500pps, and three CI modes: symmetric bilateral cochlear implant (SBCI), shifted Bilateral Cochlear Implant (ShBCI) and Unilateral Cochlear Implant (UCI).

The effects of stimulation rate and CI mode on speech intelligibility were simulated in normal-hearing subjects. The stimulation rate is simulated by varying the degree of overlap between successive analysis frames in the sine wave vocoder. With UCI mode, only one ear is stimulated. With SBCI, two ears are stimulated using the same frequency analysis filter. With ShBCI mode, we stimulate both two ears but using two different frequency analysis filters.

The effect of stimulation rate and the CI modes are tested in quiet and in noisy environment with three different Signals to Noise ratio (SNR). Trisyllabic French words processed via three CI modes and with two stimulation rate were presented to normal-hearing listeners for identification. Results showed significant interaction between CI modes (SBCI, ShBCI, and UCI) and stimulation rate. The effect of CI mode was minimal in quiet and in noisy environment with low SNR level but it was significant in noisy environment with height SNR level. A significant increment in performance was observed for higher stimulation rate in quiet and in noisy in environment.

This outcome is partly consistent with behavioral data obtained with cochlear implant studies in that CI users tend to do as well or better with bilateral stimulation in noisy environment and with higher stimulation rate.

C13

Can working memory capacity predict the perception of frequency compressed speech in noise in listeners with normal hearing?

R. J. Ellis and K. J. Munro, School of Psychological Sciences, The University of Manchester, UK

Nonlinear frequency compression is a method of compressing the high pitched components of a signal into a reduced bandwidth at a lower frequency. This is in order to improve the audibility of high frequency sounds for listeners with sensorineural hearing loss, who may have difficulty perceiving high frequency sounds even with the use of a conventional hearing device.

Whilst frequency compression has been shown to provide significant benefit to some hearing impaired listeners, results have been mixed and large individual differences have been reported (for example, Simpson et al., 2005; Glista et al., 2009). Currently, predictors of benefit from frequency compression devices remain elusive, although work by Glista et al. (2009) suggests that age (adult versus child) and severity/configuration of hearing loss may predict performance in some, but not all, speech perception tasks. One possible predictor of benefit may be performance in working memory tasks. In a review by Akeroyd (2008), it was reported that after audiological configuration, the most significant predictor of benefit from conventional hearing aids was working memory function. However, to date, no study has investigated the relationship between working memory capacity and the perception of frequency compressed speech. The aim of this study was therefore to investigate this relationship in listeners with normal hearing. The reading span test was administered to participants and scores were correlated with their performance on a sentence recognition in noise test. The results will be discussed in terms of their potential implications for the fitting and management of frequency compression hearing aids. Proposals for future research using hearing impaired listeners will also be outlined. [This research is being funded by an ESRC CASE PhD studentship in collaboration with Phonak Hearing Systems.]

References:

Akeroyd, M.A. (2008) Are individual differences in speech reception related to individual differences in cognitive ability? A survey of twenty experimental studies with normal and hearing-impaired adults. International Journal of Audiology, *47 (Supplement 2), 53-71.*

Glista, D., Scollie, S., Bagatto, M., Seewald, R., Parsa, V. and Johnson, A. (2009) Evaluation of nonlinear frequency compression: Clinical outcomes. International Journal of Audiology, 48, 632–644.

Simpson, A., Hersbach, A. A. & McDermott, H. J. (2005) Improvements in speech perception with an experimental nonlinear frequency compression hearing device. International Journal of Audiology, 44, 281-292.

C14

Objective characterization of binaural speech intelligility with hearing aids

Nicolas Ellaham[1], Christian Giguère[2], Wail Gueaieb[1]

1 School of Information Technology

2 School of Rehabilitation Sciences, University of Ottawa

Speech Intelligibility is typically assessed using subjective tests (e.g. HINT, SPIN, QuickSIN) performed under various acoustical conditions (e.g. babble noise, speech-spectrum noise). Such tests, however, are time consuming and limited to a few conditions per subject. In situations where large amounts of data on speech intelligibility are required, such as during the hearing-aid development process to assess the potential benefits for different strategies, objective models are needed. Standard objective measures of speech intelligibility (such as the AI, SII, and STI) were derived for monaural listening situations. In recent years, extensions of these models have been developed to deal with non-linear effects common in hearing aids (e.g. clipping, compression, noise reduction) or binaural listening conditions.

In this work we propose an objective measure based on the SII that combines both benefits, namely predicting speech intelligibility under binaural listening conditions, while accounting for non-linear processing by hearing devices. The proposed three-stage model is an extension of the binaural SII model of Beutelman and Brand (JASA 120: 331-342, 2006) to deal with non-linear acoustic input signals, and it operates as follows:

- STAGE 1: Speech and noise separation performed by the signal inversion technique of Hagerman and Olofsson (Acust. Acta Acust. 90: 356-361, 2004). It is performed through a dual presentation of speech and noise mixtures, with the phase of the noise inverted in the second mixture. The goal of this stage is to obtain separated estimates of speech and noise signals at the output of the hearing device at each ear, which incorporate the effects of non-linear processing inherent to the hearing device.
- STAGE 2: Binaural pre-processing using Equalization-Cancellation (EC) theory to model binaural release from masking (Durlach, JASA 35: 1206-1218, 1963). The output of this stage is a monaural set of signals (speech and noise) which incorporate the added benefit of binaural processing.
- STAGE 3: Intelligibility prediction based on the signals at the output of the EC model using the monaural SII (ANSI S3.5-1997 R2007).

The model was developed and tested for use with hearing aids. A software Hearing Aid Simulator is used to simulate hearing-impaired subjects under various binaural listening conditions with linear and non-linear corrections. Details of the procedure will be presented along with examples in different binaural spatial listening conditions and hearing aid processing (compression, noise reduction).

C15

Development of a novel measure of response confidence for a speech recognition task in competing noise

J. Andrew (Drew) Dundas
Vanderbilt University / Starkey Labs

Both clinical and population based studies have shown that a loss of confidence and independence may result from a decline in function, and that a high level of personal confidence is known to be correlated with increased quality of life. A reduction in confidence suggests a psychosocial impact of decreased function that may adversely affect the likelihood of performing a task or entering a situation that causes anxiety. Thus, when experienced as a result of decreased function, a loss of confidence can be considered to be a manifestation of handicap. Assessing changes in confidence due to amplification could serve as a useful measure of hearing aid outcome.

The relationship between confidence and performance has been explored in a variety of perceptual and memory based tasks, however this relationship has not been systematically investigated for a speech recognition task. Interestingly, it is well known amongst clinicians that many patients are confident in their listening performance, despite the fact that this perception may not accurately reflect their true performance. The Performance Perceptual Test (PPT) (Saunders and Cienkowski, 2002) was developed to identify this mismatch so that patients can be counseled to more closely align perception with reality. However, the PPT assesses confidence only at the 50% correct performance level, while it is not clear how confidence changes with performance.

In an effort to lay the groundwork for a series of studies investigating confidence in communication performance, the relationship between confidence and performance on a speech recognition in noise task was investigated in 21 normal hearing adults. Participants repeated randomly ordered NU-6 sentences presented in a background of NU-6 shaped multi-talker cafeteria noise. SNRs were selected to vary performance across a wide range. Following the repetition of each keyword, the participants rated their confidence in their response by placing a mark on 100mm visual analog and integer scales. Results suggest that 1) A visual analog scale resulted in significantly higher ratings of confidence for a given level of performance. 2) Normal hearing adults are typically overconfident in their performance. 3) Confidence does not increase in a linear relationship with performance. 4) Test-retest reliability was high for the confidence rating task of interest. Future work will investigate confidence/performance relationships in hearing impaired adults under aided and unaided conditions.

C16

A software suite for automatic beamforming calibration

Rob de Vries, Bert de Vries and Andrew Dittberner
GN ReSound, Research dept., Eindhoven, Netherlands

Beamforming is one of the most successful approaches to improve speech understanding in noise. Beamforming algorithms usually pass the signal from the look direction (zero degrees), while suppressing sounds from other directions. In general, fixed beamformers perform well in diffuse noise, while adaptive beamformers optimize their behavior based on the dominant noise source location. In order to avoid suppression of sources of interest, adaptive beamformers often make use of the estimated direction-of-arrival of impeding sounds so as to inhibit adaptation to sources in the region of interest. Most beamformers also employ some type of fading across frequency from omni-directional to full beamforming in order to avoid boosting microphone noise to unacceptable levels.

Based on the above, we can formulate the following most important performance metrics for beamforming.

- Acceptable white noise gain
- Good zero-degree response
- Reliable direction-of-arrival estimate
- Optimal suppression of diffuse noise (in case of fixed beamforming) or in all possible directions outside the region of interest (in case of adaptive beamforming)

Because the beamformer algorithm complexity is necessarily very limited for application in hearing aids, trade-offs need to be considered between these objectives. Moreover, the optimal algorithm settings vary from person to person. Since extensive algorithm fine-tuning is not feasible during fitting at the dispenser location, an optimal average setting needs to be found that performs well across the user population. At ReSound an automatic calibration suite has been developed that finds the optimal average settings based on measurements across a set of representative user configurations. This package will be demonstrated and explained in further detail during the presentation.

C17

The composite effect of narrowband and broadband spectral resolution on amplified speech scores

Evelyn Davies-Venn[1] and Pamela Souza[2]
[1] *University of Washington*
[2] *Northwestern University*

Some listeners with hearing loss show poor speech recognition scores, in spite of using amplification that optimizes audibility. Studies have suggested that differences in amplified speech recognition scores may be explained by suprathreshold abilities, such as spectral resolution. However, research findings on the relationship between spectral resolution and speech recognition are inconclusive. This study evaluated the effect of a composite, narrowband and broadband, measure of spectral resolution on amplified speech recognition scores.

A within-subject repeated measures design was used to evaluate speech recognition scores for linear, moderate fast-acting wide dynamic range compression (WDRC) and extreme fast-acting WDRC. Twenty-eight listeners with sensorineural hearing loss were tested. All test stimuli were first processed using a hearing aid simulation software to obtain moderate (CR 4:1) and extreme (CR 10:1) compression. Next, the stimuli were individually NAL-R amplified and presented through headphones. Audibility of the test stimuli were quantified using the aided audibility index. Subjects were instructed to select the correct stimulus heard from an on-screen display of 16 vowel-consonant-vowel /aCa/ nonsense syllables. Scores were computed based on each listener's response to 256 x 4 syllable tokens, in each test condition.

Spectral resolution was measured using a composite measure that accounted for each listener's narrowband and broadband spectral resolution. Narrowband spectral resolution was measured at 500 and 2000Hz, using the notched-noise masking procedure. Equivalent rectangular auditory filter bandwidths (ERB) values were calculated from the masked threshold data using a rounded exponential (roex) filter model (Patterson et al., 1982). Broadband spectral resolution was measured using the spectral modulation detection task by Soaji and Eddins (2007). This task measures a listener's ability to discriminate between a broadband noise with a flat spectrum and a sinusoidal modulated signal. Spectral modulation (SM) thresholds were measured by adaptively varying the depth of the modulated signal at 0.25, 0.5, 1.0 and 2.0 cycles/octave. Principal component analysis was applied to the two ERB values and four SM thresholds to determine a composite measure of spectral resolution. This composite measure approximated each listener's narrowband and broadband spectral resolution ability.

Linear regression analysis was used to evaluate the relationship between spectral resolution and amplified speech recognition scores. Results revealed that spectral resolution was a significant predictor of speech recognition scores for all three amplification conditions. These results suggest that even when provided with sufficient audibility, listeners with poor spectral resolution showed the least benefit from linear and nonlinear amplification. [Work supported by NIH F31 DC010127 and RO1 DC006014]

C18

A school-age listening landscape

Jeffery Crukley Ph.D. (Candidate), Susan Scollie Ph.D.

Hearing aid wearers commonly complain about listening in non-quiet environments. Some of the approaches used to assist adult listeners in these situations include: digital noise reduction algorithms, directional microphones, and frequency-gain shaping. Yet, there is currently no consensus on an optimal *pediatric* amplification strategy for non-quiet environments. Exploring the auditory environments experienced by children in their daily lives may be an informative preliminary step toward addressing the non-quiet listening needs of pediatric hearing aid wearers.

The purpose of this study is twofold: 1) to explore and classify the auditory environments of a pre-school (pre-K), an elementary school (K to 8), and a secondary school (9-12) 2) to evaluate the accuracy of three commercial hearing aids' sound classification systems in these environments.

Three sites participated in this study. At each site, empty room measurements were conducted; these included noise floor and reverberation levels, across the various rooms (e.g. classroom, gymnasium) frequently occupied by the participating cohorts of children. Next, the first author followed the cohorts throughout their regular school routines, recording noise levels with a sound level meter and making observations of the types and sources of sounds encountered by the students. The first author wore three hearing aids with data logging enabled throughout this observational data collection.

This exploration of school-age listening landscapes will report noise levels and types, reverberation levels, and percentages of time spent in the various listening situations. Further, results include a comparison of data from sound level measurements and researcher observations with the hearing aids' sound classification systems. These results provide valuable information for ongoing research into pediatric hearing aid fittings and these implications will be discussed.

C19

Assessing spatial hearing abilities in everyday listening situations

Mary T. Cord[1], Kim L. Block[1] and Brian E Walden[1], Andrew Dittberner[2]

1 Walter Reed Army Medical Center

2 GN ReSound Group

The primary purpose of this study was to develop and evaluate field methods for assessing spatial hearing. Experimental field trial methods based, in part, on the work of Gatehouse and Noble (2004) were used to compare the effect of different hearing aid microphone placements on spatial hearing abilities in everyday listening. Previous research suggests that an in-the-canal (ITC) microphone placement may provide improved spatial hearing ability as compared to a less-natural behind-the-ear (BTE) microphone placement.

Twelve hearing-impaired listeners compared a BTE to an ITC hearing aid microphone placement in daily life. Participants were fitted binaurally with modified hearing aids equipped with a microphone in the BTE housing as well as a microphone in the ITC shell. The ITC shells were custom made for each subject. Participants switched between the two microphones by pressing the program button on the BTE case to make direct comparisons of the two microphone configurations. In listening situations where a difference between the two settings was detected, listeners filled out a journal form describing the listening situation and made ratings of localization ability, segregation ability, externalization and sound quality for the two settings. They also rated their preference for one of the two settings. Prior to the field trial, participants received training on rating the dimensions of spatial hearing concepts assessed on the journal form, using simulations in the laboratory. Measures of spatial release from masking were also made in the laboratory with each microphone location.

Results of laboratory testing revealed that, on average, participants obtained significantly greater spatial release from masking and fewer localization errors with the ITC microphone location as compared to the BTE microphone placement. However, most participants noticed little or no difference between the two microphone configurations in real-world listening situations. Microphone placement, therefore, appeared to have little observable effect on spatial hearing in everyday life for these hearing-impaired listeners. It may be that knowledge of the listening environment and the availability of visual cues obviate subtle differences in spatial hearing abilities that may exist as a result of different hearing aid microphone configurations.

C20

The Effects of Digital Noise Reduction Algorithms on Speech and Noise

Foong Yen Chong, B. Audiology and Lorienne M. Jenstad
University of British Columbia

Digital noise reduction (DNR) is used in hearing aids to reduce noise and improve listening comfort by classifying signals as either speech or noise, and lowering gain when noise is detected. DNR may have potential adverse effects on speech sounds that have noise characteristics, such as fricatives. Phonemes /s/ and /z/ are important in English as they function as grammatical markers for plurals, possessives, and tense. Prelingual children with hearing impairment are not able to use contextual cues to help them discriminate and identify the phonemes. Thus, audibility and intelligibility of fricatives are of utmost important for hearing impaired children (Palmer and Grimes, 2005; Stelmachowics, Kopun, Mace, & Lewis, 1995). The spectral properties and duration of fricative noise are important to distinguish place of articulation of fricatives (Jogman et al, 2000). Our main goal in the present study is to investigate the effects of DNR algorithm on noise and speech using acoustic measures; specifically, 1) to quantify any detrimental effect of DNR on temporal and spectral characteristics of various speech sounds (particularly fricatives); and 2) to quantify the efficacy of DNR on various noises.

The speech stimuli were nonsense words from The University of Western Ontario Distinctive Features Differences test (Cheesman & Jamieson, 1996). Three types of noises were used: 1) steady-state speech-weighted noise, and two modulated noises: 2) speech modulated ICRA noise and 3) cafeteria noise; at 3 signal-to-noise ratios. Nine hearing aids with DNR were tested. Hearing aid gain was set to Desired Sensation Level 5.0 (DSL) targets. All advanced features other than DNR were deactivated. The stimuli were presented to, and recorded from, the hearing aid mounted on a Knowles Electronic Manikin for Acoustic Research Manikin (KEMAR). The Inversion Technique, as described by Souza, Jenstad, & Boike (2006), was used to separate speech and noise post-recording. The main acoustic measures used to quantify the effects of DNR were 1) spectral analysis using Praat to generate FFTs; and 2) temporal analysis using The Envelope Difference Index (EDI). Comparisons were made between unprocessed and processed speech and noise, both with and without DNR activated.

Our preliminary results show less spectral change caused by DNR on the phoneme /s/ when modulated noise is present, as opposed to steady-state noise, as expected. However, the specific effect differs across hearing aids. Ongoing work includes acoustic analysis of other fricatives and noises, including the attack time of the DNR circuit.

C21

Comparison and evaluation of beamforming techniques for 2-channel microphone digital hearing aids

Kyeongwon Cho[1], Jonghee Han[1], Jin Ryoul Kim[1], Heepyung Kim[1], Sung Hwa Hong[2], Sang Min Lee[3], Dong Wook Kim[4], Sun I. Kim[1] and In Young Kim[1]
[1] *Department of Biomedical Engineering, Hanyang University*
[2] *Department of ORL-HNS, Sungkyunkwan University*
[3] *Department of Electronic Engineering, Inha University*
[4] *Bio & Health Lab, Samsung Advanced Institute of Technology*

In modern digital hearing aids, acoustic directionality is implemented to suppress noise from unwanted direction. Conventional beamformings utilizing differential microphone array (DMA) have been used in many acoustic applications. But DMA shows frequency dependent response which can lead to distortion of the desired signal. To prevent such a distortion, broadband beamforming have been developed. However, most broadband beamforming algorithms have also limitations of directionality in low frequency regions when number of microphones and microphone spacing are small. Therefore, it is required to evaluate the performance of beamformers for 2-channel microphone digital hearing aids.

The broadband beamformer used here was designed by 2 stages of optimization in leastsquares sense. First, by sequential quadratic programming (SQP) in MATLAB toolbox, weighting coefficients were extracted based on arbitrary beamformer response. Second, by BFGS Quasi-Newton method, finite impulse response (FIR) filters for each microphone were calculated.

In this study, the broadband beamforming was compared to first and second-order adaptive differential microphone array (ADMA). We assumed that the desired signal comes from the front 0° and 2 omnidirectional microphones are configured in an end-fire array with a small microphone spacing of 1cm for future application in digital hearing aids. Ideal beampatterns and expected beampatterns to sinusoidal inputs for each algorithm were evaluated. The broadband beamforming showed frequency-flat response relative to the other algorithms while the low frequency region between 200~500 showed less directionality. For the low frequency region under 200Hz, the beamformer was not optimized and showed no directionality. We also simulated each algorithm with IEEE sentence and NoiseEx data. For 4 null direction settings of 100°/-100°, 120°/-120°, 150°/-150°, and 180° in azimuth, noise directions were varied from 90° to 180° in azimuth. In results, the broadband beamforming had higher improved signal-to-noise ratio (SNR) of about 12dB in maximum and similar perceptual evaluation of speech quality (PESQ) of about 3 when null direction and noise direction were aligned. All evaluation and simulation above were performed on Mathworks MATLAB and Simulink. [This work was supported by grand No. 10031741 from the Strategic Technology Development Program of Ministry of Knowledge Economy].

C22

The problems of inner-hair-cell impairment for hearing aid gain prescriptions

Ian C. Bruce, McMaster University

Historically, impairment of outer hair cells (OHCs) in the cochlea was thought to be the predominant contributor to sensorineural hearing loss (SNHL). The loss of cochlear compression and broadening of cochlear filtering that are produced by OHC impairment have consequently guided the development of compression algorithms in hearing aids and prompted the investigation of spectral sharpening schemes. More recently there has been an interest in the consequences of cochlear “dead regions”, that is regions of complete inner hair cell (IHC) dysfunction, for speech intelligibility and hearing aid amplification prescriptions. However, there is now growing evidence from physiological studies in animals and psychophysical studies in humans that *partial* IHC impairment is a substantial component of SNHL in most cases. Estimates of the IHC contribution to threshold shifts are typically around 1/3 to 1/2 of the total threshold shift. Furthermore, a recent animal study has shown that significant dysfunction of the IHC/auditory-nerve-fiber (ANF) synapse and subsequent degeneration of ANFs can occur even in cases of a temporary threshold shift (Kujawa & Liberman, J. Neurosci. 2009).

In this study we have utilized a computational model of the auditory periphery (Zilany & Bruce, JASA 2006, 2007) to study the individual and combined effects of OHC and IHC impairment on optimal hearing aid amplification gains. The mean discharge rate and spike-timing information in response to speech stimuli were both taken into consideration when analyzing the neural response patterns across the model ANFs. Simulation results show that optimal gains predicted by the model for mild impairment of just the OHCs match the NAL-R and NAL-NL1 prescriptions. In contrast, any degree of IHC impairment leads to a mismatch between optimal gains for restoring ANF mean rates and spiking-timing, and the best that linear prescriptions and wide-dynamic-range-compression schemes can achieve is a balance between restoring these two aspects of the ANF response. These results prompt the development of more sophisticated amplification schemes to better deal with the distortions in ANF spike-timing information arising from IHC and OHC impairment. [This work was supported by NSERC Discovery Grant 261736.]

C23

Using psychoacoustic thresholds to improve hearing aid fittings

Marc Brennan[1], Frederick Gallun[2] and Pamela Souza[3]

1 *University of Washington*
2 *National Center for Rehabilitative Audit*
3 *Northwestern University*

Compared to normal hearing listeners, listeners with hearing loss recognize less speech even when utilizing appropriate amplification. Additionally, aided speech recognition scores vary widely across individuals. We propose that incorporating aided psychoacoustic measures could improve hearing aid fittings. Compared to speech measures, psychoacoustic measures (1) require less top-down processing and (2) can be more reliable.

In this study the relationship between psychoacoustic thresholds, speech recognition, and hearing aid output was determined. Adult subjects with mild to moderate hearing loss were fit with binaural hearing aids. The hearing aids were adjusted to meet individual real-ear targets. Subjects identified nonsense syllables containing voiceless consonants. To focus on temporal cues, spectral cues were reduced. Subjects also detected amplitude modulations and gaps in a broadband carrier, which is similar to the carrier in voiceless speech. To determine signal audiility, real ear measures were obtained for the aided modulation and gap stimuli.

Subjects with better psychoacoustic thresholds had better speech recognition. We suspect that improving psychoacoustic thresholds for the poor performing subjects could improve speech recognition. The poor performance could be related to the bandwidth of the carrier signal in the auditory system. Therefore, the basilar membrane excitation model of Moore and Glasberg (1983) was used to predict the excitation pattern of the broadband carriers after hearing aid processing. Preliminary analyses indicate wide variability in excitation pattern across subjects. Subjects with wider bandwidth tended to have better psychoacoustic thresholds and speech recognition, but there was variability. These results indicate that including gap and amplitude modulation thresholds could improve hearing aid fittings. [Work supported by NIDCD and VA-RR&D.]

C24

Predicted effects of amplification on spatiotemporal coding of vowels in noise

Jonathan Boley[1] and Michael Heinz[1,2]

1 *Weldon School of Biomedical Engineering, Purdue University*
2 *Department of Speech, Language, and Hearing Sciences, Purdue University*

Hearing aids can restore some degree of hearing to many patients. However, the ability to understand speech in noisy conditions remains impaired relative to normal hearing. 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. 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 predict the effects of both linear and nonlinear amplification schemes on spatiotemporal coding of vowels in noise. We used a phenomenological model of the auditory nerve to calculate the correlation and relative delay of phase-locked responses as a function of cochlear place. The spatiotemporal coding of a vowel was quantified in terms of cross-fiber correlation and delay near the cochlear best places for the first three formants and troughs. In this study, we used a computational model of a normal hearing system and a model of an impaired system designed to replicate a moderate sloping hearing loss. Stimuli consisted of a single vowel presented at signal-to-noise ratios (SNRs) varying from quiet to -6dB, presented at a base level of 65dB SPL and at amplified levels (using either a linear or a compressive gain prescription).

Consistent with previous research, our model predictions indicate that sensorineural hearing loss results in an increased width of cross-fiber correlated activity and a reduction of the traveling wave delay between nearby fibers. Our results suggest that both linear and compressive amplification can reduce the spread of correlated activity, and that the width of correlation varies as a function of SNR. However, neither this spread of correlation nor the traveling wave delays are restored to normal with hearing aid amplification. Future work will focus on validating the model results with neurophysiological experiments, 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.]

C25

Sound localization with BTE and CIC hearing aids

Virginia Best[1], Susie Valentine[2], Sridhar Kalluri[2], Simon Carlile[3], Brent Edwards[2]
1 *University of Sydney*
2 *Starkey Labs*
3 *VAST Audio Pty Ltd*

Despite the increasing popularity of BTE hearing aids relative to CIC hearing aids, the perceptual consequences of the different microphone placements associated with these hearing-aid styles are not well understood. We tested the hypothesis that the CIC hearing-aid style offers superior localization ability over the BTE hearing-aid style by providing access to the high-frequency spectral cues for localization generated by the pinnae.

We studied localization ability by measuring how accurately single words presented at random from all around the head were localized. Eleven listeners with mild to moderate sensorineural hearing loss fitted with CIC hearing aids and with comparably fitted receiver-in-canal style BTE hearing aids participated in the experiment. The fitted hearing aids provided gains according to standard clinical practice. We measured localization ability immediately following fitting for both hearing-aid styles and again after a 6-week acclimatization period to each hearing-aid style.

We quantified localization accuracy in terms of errors in elevation judgments, errors in lateral-angle judgments and confusions of front and back locations. Prior to acclimatization, the ability to distinguish front from back locations was superior with CIC hearing aids compared to BTE hearing aids, while the two styles gave comparable results for judging elevation and lateral angle. Post-acclimatization, confusions of front and back locations reduced for both styles, but the CIC still style still had superior front-back discrimination. Acclimatization had no effect on elevation and lateral-angle judgments for either style. Additionally, the listeners with hearing loss never reached the same performance level as a control group of normal-hearing listeners.

To investigate if limited audibility of the high frequencies was a factor in why the hearing-impaired listeners were not able to achieve aided performance similar to the normal-hearing listeners, we conducted a follow-up study. We repeated the protocol with eleven new participants and fit the hearing aids with the CAMEQ prescriptive formula to provide additional high frequency information. Results again showed that participants had fewer front-back reversals when using the CIC style hearing aid when compared to the BTE style hearing aid. However, performance was still poorer than the normal-hearing participants.

These results indicate that CIC hearing aids are superior to BTE hearing aids for three-dimensional sound localization. The inability of both hearing-aid styles to restore localization ability to the level for normal listeners suggests an area in which all hearing-aids can be further improved in order to restore normal abilities.

C26

Simultaneous localization and identification of environmental sounds in auditory scenes by normal hearing and hearing impaired listeners

Thomas Behrens[1,2] and Florian Kramer[1,3]
1 *Oticon A/S, Research Centre Eriksholm, Snekkersten, Denmark*
2 *Oticon A/S, Audiology and Algorithms, Smørum, Denmark*
3 *Jade Hochschule, Institute of Hearing Science and Audiology, Oldenburg, Germany*

In everyday listening situations, it is necessary to simultaneously localize and identify concurrent sound sources. A test paradigm was developed to investigate the abilities of normal hearing persons and hearing impaired test-persons wearing hearing aids in simultaneous identification and localization of environmental sounds. Nine normal hearing and six hearing impaired test subjects took part in initial evaluation of the test paradigm. The influence of the scene, number of signals, test-person and hearing impairment were analyzed by measuring the mean absolute localization error and the identification accuracy.

The environmental sounds were selected to form four auditory scenes consisting of six signals. An example is a “Park scene” with animal sounds such as a dog barking. Accuracy of identification and localization in each scene was tested when presented with of three, four and five sounds. The sounds were presented concurrently with a spatial separation of 30 degrees in the frontal horizontal plane, symmetrically aligned around the listener’s position. The duration of the sounds was five seconds. The test took place in an anechoic room.

Localization error for normal hearing listeners was on the order of 2-4 degrees in the three signal condition dependent on the scene. For the hearing impaired group mean localization error of 10-15 degrees was typical for the three signal condition. The mean identification performance of the normal hearing and hearing impaired test persons was around 90% and 80% correct in the three signal condition. The performance decreased with increasing number of signals for both groups by 7-12 degrees (localization) and 10-20% (identification). Furthermore, factors such as scene, number of signals and test-person were found to have a highly significant influence on the results for both localization and identification ($p<0.01$).

The results further hint at a difference in listening strategy between the two subject groups.

C27

External receiver provides superior gain in open fittings

Elin Barrock and Vivian Amdal Hals
Sør-Trøndelag University College

Open fitting hearing aids have become desirable for many clients because of their cosmetical appeal -and comfortable fit. Open fittings reduces occlusion to a minimum, and are therefore a good alternative for clients with normal hearing in the low frequencies.

It has been suggested that open fittings with the Receiver In The Ear (RITE) may provide more gain than open fittings with the Receiver In The Aid (RITA) due to the greater distance between the microphone and the receiver. Longer distance between the two transducers enables the hearing aid to give more gain before feedback occurs. Also, earlier studies suggest that replacing the thin-tube with a thin-wire eliminates possible tube-resonances, which may result in a smoother frequency response.

However, research on whether or not the location of the receiver has an impact on how much gain an open fitting can produce is sparse. In this study, the main objective was to compare the high-frequency gain between the two solutions.

Two open fitting hearing aids were used, Phonak’s microSavia Art dSZ (RITA) and microSavia Art CRT dSZ (RITE). The hearing aids were fitted based on a simulated high-frequency hearing loss. NAL-NL 1 was used as fitting rationale. The processing algorithms were turned off to ensure that the calculated gain would be the same. Real ear measurements on 14 subjects (28 ears) with normal hearing were used to measure gain.

The results revealed that RITE provided significantly more gain than RITA for all frequencies between 1 kHz to 6.3 kHz ($p < .001$). The maximum difference was obtained at 6.3 kHz (7.8 dB), whereas the minimum difference was obtained at 2 kHz (0.8 dB). At 3.4 and 5 kHz, RITE gave 5 dB more gain than RITA.

C28

Laboratory evaluation of directional preference: Effects of stimulus type and location

Shilpi Banerjee, Starkey Laboratories, Inc.

Directional benefit has been unequivocally demonstrated to improve speech understanding in background noise in the laboratory. However, real-world directional advantage can be best described as lukewarm. Studies have shown that success with directional microphones cannot be reliably predicted from laboratory measures of directional advantage. While surveys of hearing aid users indicate that 38% are dissatisfied with, and 95% desire improvement in, the performance of hearing aids in noisy situations, the prevalence of directional microphones in the marketplace is only ~25%. Nonetheless, the good news is that hearing aid users do report a directional advantage in several environments.

The disconnect between laboratory and real-world findings is typically attributed to the acoustics of the environment. This includes: (1) the presence, location and distance of signal and noise, (2) reverberation, and (3) typical input levels. It is conceivable that the hearing aid wearer's internal criterion in a particular listening situation would influence directional preference. That is, some situations may result in conflicting outcomes depending on whether maximizing speech understanding or optimizing listening comfort is most important to the individual. Preferences may further be confounded by binaural considerations. Specifically, in some situations, omnidirectional may be preferred in one ear while directional is preferred in the other.

The effects of these factors were assessed in the laboratory by 20 adults with mild-to-moderate sensorineural hearing loss. Participants were fitted bilaterally with BTEs. Directional benefit in daily life is often assessed on the basis of the patient's response to "Which program did you prefer?" In keeping with this theme, subjective preference for directionality was evaluated in a paired-comparison format using standard laboratory (i.e., controlled) and simulated real-world stimuli. Various configurations of speech and noise location were used. Bilaterally symmetrical and asymmetrical settings were judged. Finally, preferences for optimized speech understanding and maximized listening comfort were obtained separately. The preference data were analyzed using the Bradley-Terry model. The results and their clinical implications will be discussed.

C29

Evaluation of a pediatric outcome measure to assess auditory development

Bagatto, M, Moodie, S, Brown, C, Malandrino, A, Keene, K, Seewald, R, Richert, F, Bartlett, D, Clench, D, Scollie, S
University of Western Ontario

The primary goal of Early Hearing Detection and Intervention (EHDI) programs is to provide effective intervention by six months of age to maximize the infant's natural potential to develop language and literacy skills. Intervention with hearing aids is a common choice among families and audiologists have access to scientifically-based strategies and clinical tools to ensure the hearing aids are fitted appropriately to the infant. Outcome evaluation is a key component of the pediatric hearing aid fitting process, however, there is little research related to what a typical outcome might be for an infant who wears hearing aids and how to systematically track the child's auditory development and performance over time. This is in part due to the lack of normed and validated outcome measures available for use with infants and children who wear hearing aids. Research has focused on the communication outcomes of children involved in EHDI programs and what factors may impact outcome[e.g.,1]. Limitations of this work include extensive test batteries which are impractical to administer and score in a typical clinical situation.

The current work focuses on the use of an outcome measure that is suitable for use with infants and children and that is feasible for clinical use. The LittlEARS® Auditory Questionnaire is a 35-item questionnaire that assesses auditory development[2]. Normative values were derived from normal-hearing children from German-speaking families and validation was completed with German and Italian children who wear cochlear implants[3,4]. The short questionnaire was administered by pediatric audiologists at 4 clinics across Ontario, Canada to parents of over 250 children ranging in age from 1 month to 6 years. The clinicians followed protocols as part of Ontario's EHDI program, which included fitting the hearing aids to the Desired Sensation Level (DSL) prescriptive algorithm[5]. Profiles of the children vary and include: normal hearing, unaided hearing loss, aided hearing loss and the presence of other comorbidities. Data from this population study will be compared to existing norms and presented by subpopulation to illustrate the impact of hearing loss, hearing aid fitting, and comorbidities in a representative clinical population.

References

1 *Sininger, Y, Grimes, A, Christensen, E. 2010. Auditory development in early amplified children: Factors influencing auditory-based communication outcomes in children with hearing loss, Ear Hear, 31(2):166-185.*

2 *Tsiakpini, L, Weichbold, V, Kuehn-Inacker, H, Coninx, F, D'Haese, P, Almadin, S. 2004. LittlEARS Auditory Questionnaire, MED-EL, Innsbruck, Austria.*

3 *Coninx, F, Weichbold, V, Tsiakpini, L, Autrique, E, Bescond, G, Tamas, L. 2009. Validation of the LittlEARS® Auditory Questionnaire in children with normal hearing, Int J Pediatr Otorhi, 73:1761-1768.*

4 *Kuehn-Inacker, H, Weichbold, V, Tsiakpini, L, Coninx, F, D'Haese, P. 2003. LittlEARS Auditory Questionnaire: Parents questionnaire to assess auditory behaviour, MED-EL, Innsbruck, Austria.*

5 *Scollie, S, Seewald, R, Cornelisse, L, Moodie, S, Bagatto, M, Laurnagaray, D, et al. 2005. The Desired Sensation Level Multistage Input/Output Algorithm, Trend Amplif, 9:159-197.*

C30

The influence of spectral and spatial characteristics of early reflections on speech intelligibility

I. Arweiler, J. M. Buchholz, T. Dau
Technical University of Denmark (DTU), Centre for Applied Hearing Research (CAHR)

The auditory system employs different strategies to facilitate speech intelligibility in complex listening conditions. One of them is the integration of early reflections (ER's) with the direct sound (DS) to increase the effective speech level. So far the underlying mechanisms of ER processing have mostly been addressed by manipulating the temporal characteristics of ERs. To achieve a more complete picture, the present study investigates the influence of the spectral and spatial characteristics of ERs on speech intelligibility.

Speech intelligibility tests were performed with 9 normal-hearing and 8 hearing-impaired listeners in a virtual auditory environment. In this setup with 29 loudspeakers, the amplitude of the DS and the ERs could be varied independently. The ER pattern was taken from a classroom simulated with the room acoustic software Odeon. Thus, the spectral, spatial and temporal characteristics of the ERs were preserved. The DS of the speech signal was always presented from the front and the ERs were either presented from the front or spatially distributed. Speech intelligibility was measured monaurally and binaurally for different types of interferers.

It was found for both groups of listeners that speech intelligibility improved with added ER energy, but less than with added DS energy. An efficiency factor was introduced to quantify this effect. The difference in speech intelligibility could be mainly ascribed to the differences in the spectrum between the speech signals with and without ERs. As the ERs were changed from spatial to frontal presentation the speech intelligibility increased in the same manner for monaural and binaural listening. This indicates that the integration of ERs and DS depends on the direction of the ER's, but not on the listening mode (monaural vs. binaural). The direction-dependency could be explained by the spectral changes introduced by the pinna, head, and torso. The results will be important with regard to the influence of signal processing strategies in modern hearing aids on speech intelligibility, because they might alter the spectral, spatial and temporal cues important for the benefit from ERs.

C31

Sound quality ratings for the International Speech Test Signal (ISTS)

Kathryn Arehart, James Kates and Peter Moats
University of Colorado at Boulder

The International Speech Test Signal (ISTS) was developed by the European Hearing Instrument Manufacturing Association (EHIMA) to be used in the standardized evaluation of hearing aids. The ISTS was designed to be a speech-like test signal that is representative of multiple languages but which is not intelligible. As such, the ISTS might be an appropriate signal to use for evaluations of hearing aid sound quality, especially as objective measures of hearing aid sound quality become available. However, previous research [Preminger and Van Tasell, J. Speech Hear. Res. 38, 714-725] has shown that quality judgments are correlated with speech intelligibility, especially for low intelligibility scores. Thus, it cannot be assumed a priori that a non-linguistic signal such as the ISTS will yield sound quality ratings that are the same as for linguistically-meaningful test signals. The purpose of the present experiment was to determine whether sound quality ratings for the ISTS are similar to sound quality ratings obtained for linguistically-meaningful speech.

A simulated hearing aid was used to process the ISTS modified by (1) additive noise and nonlinear processing, (2) linear filtering, and (3) combinations of noise, nonlinear processing, and linear filtering, for a total of 100 test conditions. The sound quality of the test conditions was rated by a group of 20 listeners with normal hearing (NH) and by a group of 15 listeners with hearing loss (HL) using a five point rating scale. Listeners' ratings were then modeled using the Hearing Aid Sound Quality Index [Kates and Arehart, J. Audio Eng. Soc., in press]. HASQI includes a model of the auditory periphery that incorporates aspects of impaired hearing, and the index was fit to quality ratings of American English sentences. The correlation between the HASQI predictions and the listener ratings indicates how closely the ISTS sound quality ratings are to listener ratings of speech. In the NH group, correlations between HASQI predictions and listener ratings were 0.90 for nonlinear processing, 0.78 for linear processing, and 0.88 for combined processing. In the HI group, correlations between HASQI predictions and listener ratings were 0.96 for nonlinear processing, 0.94 for linear processing, and 0.96 for combined processing. These results demonstrate that the ISTS is comparable to American English as a test signal for hearing aid sound quality ratings, and that ISTS combined with HASQI is an effective procedure for objective quality measurements. [Supported by a grant to University of Colorado from GNResound.]

C32

Older adults expend more effort to understand audiovisually presented speech in noise

Anderson Gosselin, Penny[1,2], Gagné, Jean-Pierre[1,2]

1 École d'orthophonie et d'audiologie, Université de Montréal
2 Institut universitaire de gériatrie de Montréal

<u>Purpose</u>: Listening in noisy situations is a challenging and exhausting experience for many older adults. Hearing is a sense but listening is a skill that requires attention and intention to access and use the information that is heard. We hypothesize that older adults exert increased listening effort compared to younger adults. Listening effort refers to the attention and cognitive resources required to understand speech. Our goal is to quantify and compare the amount of listening effort that older and younger adults expend when they try to understand audiovisually presented speech in noise.

<u>Methods</u>: We used a dual task paradigm to objectively evaluate the listening effort of 25 normal hearing young adults and 20 normal hearing older adults. The primary task involved a closed-set sentence-recognition test and the secondary task involved a vibro-tactile pattern recognition test. Participants performed each task separately and concurrently under two experimental conditions: 1) when the level of noise was the same – equated level, and 2) when baseline word recognition performance did not differ between groups – equated performance.

To compare across age groups, proportional dual task costs (i.e., pDTC = (dual task – single task)/single task) were calculated for each dependent variable (i.e., primary task: percent correct and response time, secondary task: percent correct and response time). In addition, using a scale from 0-100, participants subjectively rated how accurate they thought their responses were and how much effort they required to perform each of the tasks under dual task conditions.

<u>Results</u>: For both experimental conditions, we found that older adults expended more effort to understand speech as shown by significantly larger pDTC's on the concurrent tactile task percent correct scores and response time measures. Further, while subjective ratings of task accuracy correlated with the relevant percent correct dual-task measures, effort estimates did not correlate with any of the objective dual-task measures.

<u>Conclusions</u>: Our preliminary results indicate that older adults require more processing re-

sources to understand audiovisually presented speech in noise. In addition, the subjective estimates of effort did not correlate with any of the dual-task measures. As a result, caution should be advised if listening effort is measured by subjective means only. An objective measure of listening effort could be used by clinicians 1) as an assessment tool, 2) as an outcome measure, 3) to target clients that would benefit from aural rehabilitation and, 4) to optimize hearing aid settings to improve speech understanding.

C33

Sensitivity to temporal fine structure in older adults using the TFS1 test

Melinda C. Anderson[1], Kathryn H. Arehart[1], James M. Kates[1,2] and Naomi H. Croghan[1]

Both hearing loss and age can affect the ability of listeners to process important components of sound. Some of these components relate to the temporal structure of sound, and include slowly varying changes (temporal envelope) and much faster-changing components (temporal fine structure). The purpose of this poster is to quantify sensitivity to temporal fine structure changes in older adults using the TFS1 test [Moore & Sek, *Int J Aud*, 48,161-174]. The working hypothesis is that older adults with normal to near-normal hearing will be less sensitive to temporal fine structure changes than young adults with normal hearing.

Our results show that age negatively impacts the ability perceive TFS, although significant intersubject variability exists. This may partly explain why some older adults experience increased difficulty understanding speech in a complex environment.

The results will be discussed in terms of the differential effects of age and hearing loss on the ability to utilize temporal fine structure as a measured by the TFS1 test. [Work supported by the Institute of Cognitive Science at the University of Colorado, Boulder and GN ReSound]

C34

Effect of binaural tone vocoding on recognising target speech presented against spatially separated speech maskers

M. R. Andersen, M. S. Kristensen, T. Neher and T. Lunner
Oticon A/S, Research Center Eriksholm, Kongevejen 243, DK-3070 Snekkersten, Denmark.

In the studies of *Hopkins et al.* [J. Acoust. Soc. Am. 123, 1140-1153, (2008)] and *Lunner et al.* (2010) the benefit from monaural Temporal Fine Structure (mTFS) information was measured in terms of Speech Reception Thresholds (SRTs) using speech-in-speech tests. TFS was either kept or substituted using a tone-vocoder in a filterbank with 32 1-ERB_N wide bands. Results showed that hearing-impaired subjects were not able to utilise TFS information to the same degree as normal-hearing (NH) subjects. A first step towards a more ecological situation compared to these experiments, would be to exploit the tone-vocoder processing paradigm in a simulated spatial setup, and measure binaural TFS (bTFS) benefit. Like in the monaural case, the intention is to make the bTFS information unusable - equivalent to making the Interaural Time Difference (ITD) cues unusable - by use of a binaural tone-vocoder processing paradigm.

By the introduction of the binaural tone-vocoder processing paradigm, a concern arose that not only will the original ITD cues be removed, but artificial ITD cues pointing to a direction determined by the phase difference between the carriers of the two sides, will also be introduced. These artificial ITD cues may disturb the spatial perception and hence distort the measurements. The purpose of this experiment was to explore ways to minimise the artificial ITD cues.

The experiment was carried out with headphones utilising generic Head Related Transfer Functions to spatially separate a target speaker and two masker speakers. SRTs were measured for six NH subjects in a fixed spatial condition with either no vocoder-processing (unprocessed) or vocoding carried out before (vocBefore) or after (vocAfter) spatially separating the speakers. For the vocAfter condition three different binaural phase combinations with different artificial ITD cues were tested.

Results of the unprocessed condition and the vocBefore condition indicate the cost of removing mTFS in a spatial setup, which is quite similar to earlier experiments measuring the benefit of mTFS. Furthermore, comparison of the results from the vocBefore condition and the vocAfter conditions indicates the cost of removing the original ITD cues and introducing artificial ITD cues. A discussion about the different conditions and how binaural tone-vocoding affects the spatial perception will be given.

Reference:

Lunner T, Hietkamp RK, Andersen, MR, Hopkins K, Moore BCJ (2010). Effect of speech material on the benefit of temporal fine structure information in speech for normal-hearing and hearing-impaired subjects. Submitted to J. Acoust. Soc. Am.

C35

Development of hearing loss simulator based on characteristics of humans auditory filter

Hongsub An, Gyuseok Park, Yuyong Jeon, Sangmin Lee, Department of Electronic Engineering, Inha University, Korea

The algorithm test for digital hearing aid should be performed with hearing impaired listeners. However it takes a lot of time and high cost, and clinical test for impaired listeners must obtain IRB approval. So it is not easy to test often for impaired listeners. Therefore it is required to simulate the hearing loss of hearing impaired listener and perform the algorithm test using hearing loss simulation. The study of humans auditory filter is required before applying characteristics of auditory filter for hearing loss simulator because hearing loss simulator imitates human perception process of hearing. Until now, there have been a lot of researches on auditory filter shape. However, what is the best way was not yet concluded. In this study, we propose a hearing-loss-simulator using modified ROEX(Rounded Exponential) filters to apply different degree of hearing impairment to speech. The ROEX filters are provided with a simple mathematical function and be expected to estimate the shape of auditory filters very well. The filter parameters were derived from previous studies that have been done by Mashshi Unoki et al. However the parameters are modified for our purpose. In the hearing loss simulator, speech is converted from time domain to frequency domain through FFT. Then based on the selected degree of hearing loss on 125Hz, 250Hz, 500Hz, 1KHz, 2KHz, 4KHz, 6KHz, filter shape and level is decided. Decided filter is applied to frequency bin of speech, and converted to time domain. To evaluate our simulator, we perform a word discrimination score test with hearing impaired listeners using the original speech, and then with normal hearing listeners using speech processed by hearing loss simulator. As a result, we are able to verify the correlation between score by normal hearing listeners and score by hearing impaired people. [This work was supported by grant No. 10031764 from the Strategic Technology Development Program of Ministry of Knowledge Economy]

C36

Measurements of ITD thresholds for partially masked noises in older and hearing-impaired listeners

M.A. Akeroyd and S.L. Suller, MRC Institute of Hearing Research (Scottish Section), Glasgow Royal Infirmary, Glasgow, UK

In most auditory circumstances there will be background sounds that can partially mask the sound to which a listener is attending. These maskers could be entirely separate sounds or delayed copies of the target from reflections. Either masker will potentially “dilute” the interaural time differences (ITDs) of the target and so reduce performance in directional hearing. This experiment therefore measured thresholds for changes in the ITD of a target sound as a function of the amount of partial masking. We intend to test a large group of older hearing-impaired listeners — currently eight older listeners have participated (age = 64-73; better-ear hearing level = 16 - 49 dB HL [0.5/1/2/4-kHz average]), along with two young normal-hearing listeners (aged 26-28).

A one-interval, two-alternative forced choice design was used. A noise stimulus was presented over headphones and listeners were asked if it was to the left or right of the center of their head. The stimulus consisted of a target mixed with a masker. The target was a 500-ms speech-shaped noise given an ITD that was either positive (to the right) or negative (left). The masker was an interaurally-uncorrelated noise, again 500 ms and with the same spectrum. The two were gated after addition so that they were perceived as one. The target-to-masker ratios (TMRs) were either +3, 0, -3, or -6 dB, giving interaural correlations of the target-plus-masker sum of 0.67, 0.50, 0.34, or 0.20. Thresholds werederived from a curve fitted to four-point psychometric functions.

The results showed that the older group consistently gave the highest ITD thresholds: the average across the four TMRs was 140 microseconds, compared to 46 microseconds for the younger group. Moreover, for every listener the ITD threshold was found to increase as the TMR was reduced. The relationship was accurately described by ITD threshold being proportional to interaural correlation raised to a power *p*. The mean *p* was -1.5 for the older listeners and -1.2 for the two control listeners. Within the older group hearing level was found to have no effect on *p*. Excepting the caveat of the small group sizes, the results demonstrate that neither age nor hearing impairment affect the rate of change of ITD with partial masking.

IHCON 2010
Attendees

● -- ● -- ●

NAME	ORGANIZATION	COUNTRY
Richard Abela	Sound Design Technologies	Canada
Harvey Abrams	Starkey Labs	USA
Mike Abry	Knowles Electronics	USA
Jayne Ahlstrom	Medical University of South Carolina	USA
Michael Akeroyd	MRC Institute of Hearing Research	UK
Jont Allen	University of Illinois	USA
Akiko Amano	Oregon Health & Science University	USA
Vivian Amdal Hals	Sør-Trøndelag University College	Denmark
Hongsub An	Inha University	South Korea
Martin Rune Andersen	Oticon A/S - Research Centre Eriksholm	Denmark
Ture Anderson	University of Southern Denmark	Denmark
Penny Anderson Gosselin	Institut universitaire de gériatrie de Montréal	Canada
Kathy Arehart	University of Colorado	USA
Steve Armstrong	SoundsGood Labs	Canada
Iris Arweiler	Technical University of Denmark	Denmark
Marlene Bagatto	The University of Western Ontario	Canada
Geoffrey Ball	Vibrant Med-El	Austria
Shilpi Banerjee	Starkey Laboratories, Inc.	USA
Elin Barrock	Sør-Trøndelag University College	Norway
Thomas Behrens	Oticon A/S	Denmark
Joel Beilin	Oticon A/S	Denmark
Keri O'Connell Bennett	Nat’l Center for Rehabilitative Auditory Research	USA
Nikolai Bisgaard	GN ReSound A/S	Denmark
Martin Blecker	EUHA Mainz	Germany
Jonathan Boley	Purdue University	USA
Marc Brennan	University of Washington	USA
Ian C. Bruce	McMaster University	Canada
Jorg M. Bucholz	Centre for Applied Hearing Research	Denmark
Maja Bülow	Widex A/S	Denmark
Pamela Burton	Siemens Hearing Instruments, Inc.	USA
Sepp Chalupper	Siemens Audiologische Technik	Germany
Jing Chen	University of Cambridge	UK
JinHo Cho	Kyungpook National University	South Korea
Kyeongwon Cho	Hanyang University	South Korea
Foong Yen Chong	University of British Columbia	Canada

NAME	ORGANIZATION	COUNTRY
George Cire	Cochlear Americas	USA
Charles Cooper	Florida Atlantic University	USA
Mary Cord	Walter Reed Army Medical Center	USA
Bram Cornelis	Katholieke Universiteit Leuven	Belgium
Catherine Creaser	MED-EL	Canada
Jeff Cruckley	The University of Western Ontario	Canada
Jerry DaBell	Sonic Innovations, Inc.	USA
Huangtin Dai	University of Arizona	USA
Evelyn Davies-Venn	University of Washington	USA
Keith Davis	Sonic Innovations, Inc.	USA
Piers Dawes	University of Manchester	UK
Joannes A.P.M. de Laat	Leiden University Medical Center	The Netherlands
Bert Devries	GNReSound	The Netherlands
Rob de Vries	GNReSound	The Netherlands
Peter Derleth	Phonak AG	Switzerland
Norbert Dillier	University of Zurich	Switzerland
Harvey Dillon	National Acoustics Laboratory	Australia
Andrew Dittberner	GN ReSound	USA
Amy Donahue	NIDCD	USA
Judy Dubno	Medical University of South Carolina	USA
Drew Dundas	Starkey Laboratories, Inc.	USA
Brent Edwards	Starkey Hearing Research Center	USA
Nicolas Ellaham	University of Ottawa	Canada
Rachel Ellis	University of Manchester	UK
Nurgun Erdol	Florida Atlantic University	USA
Brenda Fedor	Sonic Innovations, Inc.	USA
Manuela Feilner	Phonak AG	Switzerland
Michael Fisher	National Acoustic Laboratories	Australia
Mary Florentine	Northeastern University	USA
Todd Fortune	GN ReSound	USA
Tom Francart	ExpORL, K.U.Leuven	Belgium
Johnny Fredsgaard	GN Resound	Denmark
Daniel Freed	EarLens Corp.	USA
Robert Fretz	Intricon, Inc.	USA
Shigekiyo Fujii	Panasonic Shikoku Electronics Co., Ltd.	Japan
Christian Füllgrabe	University of Cambridge	UK
Jason Galster	Starkey Laboratories	USA
Bernd Gerharter	Vibrant Med-El	Austria
Ira Gerson	Telacoustics	USA
Danielle Glista	The University of Western Ontario	Canada
Alexander Grafenberg	Siemens	Germany
Jonghee Han	Hanyang University	South Korea
Woojae Han	University of Illinois at Urbana-Champaign	USA
Jörg Haubold	ciAD - competence in AudioDesign	Germany

NAME	ORGANIZATION	COUNTRY
Jovina Havard	NCRAR Portland VA Medical Center	USA
Michael Heinz	Purdue University	USA
Adam Hersbach	Cochlear Limited	Australia
Andrew Hines	Trinity College Dublin	Ireland
Inga Holube	Jade University of Applied Sciences	Germany
Kathryn Hopkins	University of Manchester	UK
Amy Horwitz	Medical University of South Carolina	USA
David Hotvet	David Hotvet Consulting	USA
Semi Hwang	Inha University	South Korea
Yoon Su Hwang	Hanyang University	South Korea
Motokuni Itoh	Panasonic	Japan
Arthur Jampolsky	Smith-Kettlewell Eye Research Institute	USA
Sofie Jansen	ExpORL, K.U.Leuven	Belgium
Ole Dyrlund Jensen	GN ReSound	Denmark
Lorienne Jenstad	University of British Columbia	Canada
Yuyong Jeon	Inha University	South Korea
Morten L. Jepsen	Technical University of Denmark	Denmark
Rene Johannesson	Oticon A/S - Research Centre Eriksholm	Denmark
Earl Johnson	US Department of Veterans Affairs	USA
Christine Jones	Phonak	USA
Euisung Jung	Kyungpook National University	South Korea
Sridhar Kalluri	Starkey Hearing Research Center	USA
Hyundeok Kang	Inha University	South Korea
James M. Kates	GN ReSound	USA
Robert Keefer	Acousticon, Inc.	USA
Gitte Keidser	National Acoustic Laboratories	Australia
Anna Kharlamova	Nat'l Center for Rehabilitative Auditory Research	USA
Dongwook Kim	Kyungpook National University	South Korea
Heepyung Kim	Hanyang University	South Korea
Jinryoul Kim	Hanyang University	South Korea
Jon Kindred	Starkey Labs, Inc.	USA
Martin Kinkel	KIND Hoergeraete	Germany
Ulrik Kjems	Oticon A/S	Denmark
Birger Kollmeier	Universität Oldenburg	Germany
Petri Korhonen	Widex	USA
Mike Kramer	Phonak Hearing Systems	U.S.A.
Abigail Kressner	Georgia Tech	USA
Michael Kristensen	Oticon	Denmark
Francis Kuk	Widex Hearing Aid Company	USA
Martin Kuster	Phonak AG	Switzerland
Julie Ladiges	The University of Western Ontario	Canada
Ariane Laplant-Lévesque	University of Queensland	Australia
Matthias Latzel	Siemens Audiologische Technik	Germany
Søren Laugesen	Oticon A/S - Research Centre Eriksholm	Denmark

NAME	ORGANIZATION	COUNTRY
Stefan Launer	Phonak AG	Switzerland
Jangwoo Lee	Kyungpook National University	South Korea
Jyung Hyun Lee	Kyungpook National University	South Korea
Arne Leijon	KTH - Sound and Image Processing Lab	Sweden
Anneke Lenssen	ExpORL, K.U.Leuven	Belgium
Suzanne Levy	EarLens Corporation	USA
M. Samantha Lewis	Nat'l Center for Rehabilitative Auditory Research	USA
Chunjian Li	Widex A/S	Denmark
Sascha Liebe	Phonak AG	Switzerland
Nicholas Liimatta	Nat'l Center for Rehabilitative Auditory Research	USA
Thomas Lunner	Oticon A/S - Research Centre Eriksholm	Denmark
Alan Lupin	University of British Columbia	Canada
Nilesh Madhu	ExpORL, K.U.Leuven	Belgium
Gil Magilen	Hearing Aid Center	USA
Jorge Marin	Georgia Tech	USA
Stefan Mauger	Cochlear Limited	Australia
Matthew McKee	GN ReSound	USA
Martin McKinney	Starkey Labs	USA
Anjali Menon	University of Illinois at Urbana Champaign	USA
Roger Miller	NIH/ NIDCD	USA
Michelle Molis	Nat'l Center for Rehabilitative Auditory Research	USA
Brian Moore	University of Cambridge	UK
Gus Mueller	Vanderbilt University	USA
Hannes Muesch	Dolby Labs	USA
Martin Müller	Univ. Hospital and Fed.l Instutitue of Technology	Switzerland
Kevin Munro	University of Manchester	UK
Atsunobu Murase	Panasonic	Japan
Markus Nagl	Vibrant Med-El	Austria
Hari Natarajan	Starkey Labs, Inc.	USA
Graham Naylor	Eriksholm Research Centre	Denmark
Stephen Neely	Boystown National Research Hospital	USA
Tobias Neher	Oticon A/S - Research Centre Eriksholm	Denmark
Hoi Ning Ng	Linnaeus Centre HEAD, Linköping University	Sweden
Jakob Nielsen	Widex A/S	Denmark
Michael Nilsson	EarLens Corp.	USA
Kazuteru Nishiyama	Rion Co., Ltd.	Japan
Nazanin Nooraei	Starkey Hearing Research Center	USA
Morten A. Nordahn	Widex A/S	Denmark
Peter Magnus Norgaard	Widex A/S	Denmark
Myriel Nyffeler	Phonak AG	Switzerland
Lars Oddershed Sunesen	Widex A/S	Denmark
Henrick Olsen	Oticon A/S	Denmark
Ole Fogh Olsen	Oticon A/S	Denmark
Per Östli	Chalmers University of Technology	Sweden

NAME	ORGANIZATION	COUNTRY
Andrew Oxenham	University of Minnesota	USA
Jon Oygarden	HiST	Norway
Manasa Panda	University of Essex	United Kingdom
GyuSeok Park	Inha University	South Korea
Aaron Parkinson	Cochlear Americas	USA
Rodney Perkins	EarLens Corporation	USA
Shawn Perman	GNResound A/S	Denmark
Srdjan Petrovic	GN Resound / Beltone	USA
Erin Picou	Vanderbilt University	USA
Jonathan Pietrobon	University of Western Ontario	Canada
Niels Pontoppidan	Research Centre Eriksholm, Oticon	Denmark
Gerald R. Popelka	Stanford University	USA
Thomas Powers	Siemens Hearing Instruments, Inc.	USA
Tim Proulx	Sonitus Medical, Inc.	USA
Sunil Puria	Stanford University	USA
Karrie Recker	Starkey Labs	USA
Todd Ricketts	Vanderbilt University	USA
Reimer Rohweder	Deutsches Hoergeraete Institut GmbH	Germany
Jerker Rönnberg	Linnaeus Centre HEAD, Linköping University	Sweden
Joyce Rosenthal	Starkey Labs	USA
Betty Rule	Unitron	Canada
Jim Ryan	Sound Design Technologies	Canada
Sébastien Santurette	Technical University of Denmark	Denmark
Gabrielle Saunders	Nat'l Center for Rehabilitative Auditory Research	USA
Susan Scollie	The University of Western Ontario	Canada
Jack Scott	The University of Western Ontario	Canada
Sangwan Seo	Hanyang University	South Korea
Kiwoong Seong	Kyungpook National University	South Korea
Roger Serwy	University of Illinois	USA
Anu Sharma	University of Colorado	USA
Helen Simon	Smith-Kettlewell Eye Research Institute	USA
Dan Sklare	NIDCD, NIH	USA
Karolina Smeds	ORCA Europe, Widex A/S	Sweden
Zachary Smith	Cochlear Ltd.	USA
Niels Søgaard Jensen	Oticon A/S - Research Centre Eriksholm	Denmark
Sig Soli	House Ear Institute	USA
Pam Souza	Northwestern University	USA
Brenton Steele	Dynamic Hearing	Australia
Andre Steinbuss	Siemens	Germany
Stefan Stenfelt	Linköping University	Sweden
Georg Stiefenhofer	Widex A/S	Denmark
Kevin Still	Nat'l Center for Rehabilitative Auditory Research	USA
Michael Stone	University of Cambridge	United Kingdom
Olaf Strelcyk	Starkey Hearing Research Center	USA

NAME	ORGANIZATION	COUNTRY
Christopher Struck	CJS Labs	USA
David Suelzle	The University of Western Ontario	Canada
Masahiro Sunohara	Rion Co., Ltd.	Japan
Hamidreza Taghavi	Chalmers University of Technology	Sweden
Elizabeth Talmage	Bay Pines VA	USA
Chin-Tuan Tan	New York University, School of Medicine	USA
Michael Tetelbaum	Sound Design Technologies	Canada
Anne Marie Tharpe	Vanderbilt University	USA
Thilo Thiede	Widex A/S	Denmark
Steve Thompson	Penn State University	USA
Celina Ann Tobias	Philippine National Ear Institute	Philippines
Georg M. Traeland	Soerlandet Hospital	Norway
Tim Trine	Starkey Labs	USA
Richard Tyler	University of Iowa	USA
Michael Ungstrup	Widex	Denmark
Bas Van Dijk	Cochlear Technology Centre	Belgium
Aart van Halteren	Sonion Nederland B V	The Netherlands
Niek J. Versfeld	VU University Medical Center	The Netherlands
Michael Vorländer	Technical University (RWTH)	Germany
Bernd Waldmann	Phonak AG	Switzerland
Daniel Warren	Knowles Electronics	USA
Charles Watson	Indiana University	USA
Thomas Weddell	University of Sussex	United Kingdom
Adam Westermann	Technical University of Denmark	Denmark
Søren Westermann	Widex A/S,	Denmark
William Whitmer	MRC Institute of Hearing Research	United Kingdom
Victoria Williams	Bay Pines VA & University of South Florida	USA
Mario Wolframm	Vibrant Med-EL	Austria
Bill Woods	Starkey Hearing Research Center	USA
Ke Xu	West China Hospital of Sichuan University	China
Yoshiyuki Yoshizumi	Panasonic Shikoku Electronics Co., Ltd.	Japan
Bill Yund	VA NCHCS	USA
Justin Zakis	Dynamic Hearing	Australia
Timothy Zeyl	McMaster University	Canada
Tao Zhang	Starkey Labs	USA