IHCON 2008

International Hearing Aid Research Conference

August 13 – 17, 2008

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LAKE TAHOE, CALIFORNIA

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The House Ear Institute

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Akiko Amano-Kusumoto	Oregon Health & Sciences University
Stephanie Bor	University of Washington
Suzanne Carr	Boston University
Jeff Crukley	University of Western Ontario, Canada
Stefan Fredelake	University of Applied Sciences, Germany
Christian Fullgrabe	University of Cambridge
Kathryn Hopkins	University of Cambridge
Benedict Leung	McMaster University
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Melissa Polonenko	University of Western Ontario
Anne Schlueter	University of Applied Sciences, Germany
Yi Shen	Indiana University
Thamar Van Esch	AMC Amsterdam

Daily Schedule

WEDNESDAY, AUGUST 13, 2008

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 14-16, 2008

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/Poster Session Continues**

SUNDAY, AUGUST 17, 2008

7:00 AM	**Breakfast and Checkout**
8:00 AM	**Morning Session**
9:10 AM	**Break**
9:30 AM	**Morning Session Continues**
10:40 AM	**Adjournment (busses leave for airport with boxed lunches for passengers)**

PROGRAM SUMMARY

WEDNESDAY, AUGUST 13

WELCOME AND KEYNOTE ADDRESS

7:30 PM – 8:45 PM

Moderator: Peggy Nelson

Welcome Remarks: Sig Soli, Brian Moore

KEYNOTE ADDRESS

Barbara Shinn-Cunningham	Listening in complex environments, with special emphasis on the effects of hearing impairment

THURSDAY, AUGUST 14

SESSION ONE

8:00 AM – 9:45 AM

MECHANISMS UNDERLYING HEARING IMPAIRMENT AND INDIVIDUAL DIFFERENCES IN PATTERNS OF HEARING LOSS

Moderator: Judy Dubno

Jochen Schacht	Mechanisms and prevention of acquired hearing impairment
Morten Jepsen	Modeling auditory perception of individual hearing-impaired listeners
Su-Hyun Jin	Spectral resolution and interrupted speech perception

POSTER SESSION A 9:45 AM – 11:00 AM

SESSION TWO

11:10 AM – 12:25 PM

TREATMENT OPTIONS FOR CONDUCTIVE AND MIXED LOSSES

Moderator: Karolina Smeds

Stefan Stenfelt	Mechanisms of bone conduction and its use for hearing-impaired persons
Bill Hodgetts	Patient-derived versus audibility-derived fittings in BAHA users: a validation study

SESSION THREE

5:15 PM – 7:00 PM

COGNITIVE FACTORS INFLUENCING SPEECH PERCEPTION IN NOISE

Moderator: Virginia Best

Mary Rudner	Aided speech recognition in noise, percei effort and explicit cognitive capacity
Anastasios Sarampalis	Understanding speech in noise with hearing loss: Measures of effort
Gabrielle Saunders	Performance Perceptual Test (PPT) and the Acceptable Noise Level (ANL): what do they measure?

FRIDAY, AUGUST 15

SESSION FOUR

8:00 AM – 9:45 AM

THE LEGACY OF STUART GATEHOUSE

Moderator: Michael Akeroyd

Introduction by Graham Naylor

Quentin Summerfield	Stuart Gatehouse's legacy for hearing research: past, present, and future
William Noble	A self-rating measure with emphasis on binaural hearing function: The Speech, Spatial, and Qualities of Hearing scale
Adrian Davis	Measuring quality in audiology: A global (or Gatehouse) framework

POSTER SESSION B 9:45 AM – 11:00 AM

SESSION FIVE

11:10 AM – 12:20 PM

WHY YOU LOSE IT AND WHAT HAPPENS WHEN YOU DO

Moderator: Jan Wouters

Karen Avraham	Human genetics of hearing loss
Brian Moore	The role of temporal fine structure in pitch and speech perception by people with normal and impaired hearing

SESSION SIX

5:15 PM – 6:50 PM

PHYSIOLOGICAL AND PHYSICAL FACTORS RELEVANT TO AUDITORY FUNCTION IN NORMAL AND IMPAIRED HEARING

Moderator: Joanna Robinson

Charles Liberman	Cochlear nerve degeneration after "temporary" noise-induced hearing loss
Faheem Dinath	Hearing aid gain prescriptions balance restoration of auditory nerve mean-rate and spike-timing representations of speech
Ryan McCreery	Use of forward pressure level (FPL) to minimize the influence of acoustic standing waves during probe-microphone hearing

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SATURDAY, AUGUST 16

SESSION SEVEN

8:00 AM – 9:45 AM

CURRENT AND FUTURE TRENDS IN SIGNAL PROCESSING FOR HEARING AIDS

Moderator: Brent Edwards

DeLiang Wang	Computational auditory scene analysis and its potential application to hearing aids
Stefan Launer	Future trends in hearing instrument technology
Joshua Alexander	Effects of frequency lowering in wearable devices on fricative and affricate perception

POSTER SESSION C 9:45 AM – 11:00 AM

SESSION EIGHT

11:10 AM – 12:20 PM

LARGE-SCALE STUDIES OF SIGNAL PROCESSING AND AUDITORY FUNCTION

Moderator: Pamela Souza

Heleen Luts	Evaluation of signal enhancement strategies for hearing aids: a multicenter study
Thamar van Esch	Evaluation of the 'Auditory Profile' test battery in an international multi-centre study

SESSION NINE

5:15 PM – 6:50 PM

OCCLUSION AND OWN-VOICE PERCEPTION

Moderator: Michael Stone

Jorge Mejia	Acoustically transparent hearing aids: an electronic vent for hearing aids
Soren Laugesen	A large-scale substantiation of own-voice issues in hearing-aid users, part I: There is more than just occlusion
Gitte Keidser	The effect of openness of the fitting on the relative loudness perception of low and high frequency sounds

SUNDAY, AUGUST 17

SESSION TEN

8:00 AM – 9:10 AM

AUDITORY LEARNING AND TRAINING

Moderator: Christian Füllgrabe

Larry Humes	Improving aided speech communication through auditory training: A review of current approaches and future applications
Monica Hawley	Intervention for restricted dynamic range and reduced sound tolerance: Clinical trial using modifies Tinnitus Retraining Therapy

BREAK 9:10 AM – 9:30 AM

SESSION ELEVEN

9:30 AM – 10:40 AM

PERCEPTUAL AND PHYSICAL MEASURES OF HEARING AID SIGNAL PROCESSING

Moderator: Leonard Cornelisse

Kathryn Arehart	Effects of linear, nonlinear and combined linear and nonlinear distortion on perceived speech quality
Thomas Rohdenburg	Objective quality measures for (binaural) hearing aids
Inga Holube	Development and Analysis of an International Speech Test Signal (ISTS)

ORAL PROGRAM

Wednesday, August 13

KEYNOTE ADDRESS

7.45 PM **AUDITORY OBJECT FORMATION, SELECTIVE ATTENTION, AND HEARING IMPAIRMENT**

Barbara G. Shinn-Cunningham, Boston University Hearing Research Center

A common complaint amongst hearing-impaired listeners is that they have difficulty communicating in everyday social settings when there are multiple, competing talkers. In contrast with hearing-impaired listeners, normal-hearing listeners are relatively good at focusing attention on a source of interest and switching attention rapidly from one talker to another. We can gain insight into the problems of the hearing-impaired by studying the ability of normal-hearing listeners to selectively attend in a complex acoustic scene. Results suggest that the ability of normal-hearing listeners to focus and switch attention depends on the ability to analyze the acoustic scene and form perceptual auditory objects properly. Unfortunately, sound features important for auditory object formation and selection may not be robustly encoded in the auditory periphery of hearing impaired listeners. Degraded acoustic cues are likely to interfere with auditory object formation, reducing the ability to filter out competing sound sources. Moreover, these peripheral degradations may also reduce the salience of higher-order auditory cues such as location, pitch, and timbre that enable normal-hearing listeners to select a desired sound source out of a sound mixture. Both of these effects are likely to contribute to the difficulties that hearing-impaired listeners experience in social settings with competing sound sources.

Thursday, August 14

SESSION ONE

Mechanisms Underlying Hearing Impairment and Individual Differences in Patterns of Hearing Loss

Moderator: Judy Dubno

8.00 AM MECHANISMS AND PREVENTION OF ACQUIRED HEARING IMPAIRMENT

Jochen Schacht, Kresge Hearing Research Institute, University of Michigan, Medical Sciences Bldg I, Ann Arbor, MI 48109-0506

Exposure to noise, treatment with ototoxic medications, and living too long are major causes of acquired hearing impairment today. Drug- and noise-induced hearing loss affect up to 5% of the world population and put about 10% at risk. Noise trauma is the most prevalent occupational disorder worldwide, and the drugs of highest concern are the anticancer agent cisplatin and the aminoglycoside antibiotics. Age-related hearing loss has both genetic predisposition and "aging" components. Presbycusis may start when people reach their 40s and 50s (earlier in men than in women). Approximately half the population suffers from a significant hearing impairment in their sixties, 66% are afflicted by age 69, and 90% after age 80. This triad of acquired hearing loss has been well investigated experimentally since animal models have long existed for each of these conditions.

Studies on the underlying molecular mechanisms have shown certain similarities between drug-, noise-, and possibly age-related loss of hair cells. An important contributor appears to be oxidant stress, created by free radicals, leading to cell injury and triggering cell death. In addition, each disorder shows specific homeostatic (survival) responses and variations on pathways to hair cell loss. Oxidant stress, for example, may take different forms, being primarily caused by oxygen free radicals or nitrogen free radicals. Likewise, both apoptotic and necrotic pathways operate in the sensory cells of the inner ear and the precise signaling pathways will vary with the type of insult and its severity.

Notwithstanding individual differences, the notion of oxidant stress as a common denominator is supported by the fact that the application of antioxidants is an effective protective therapy. This has clearly been shown for drug- and noise-induced hearing loss and, to a limited extent, for age-related hearing loss. The presentation will discuss details of the underlying causes of these forms of acquired hearing impairment and the emerging strategies for prevention.

Acknowledgement: The author's research on acquired hearing loss is supported by research grants DC 03685, DC 06457 and AG 025164 from the National Institutes of Health.

8.45 AM MODELING AUDITORY PERCEPTION OF INDIVIDUAL HEARING-IMPAIRED LISTENERS

Morten L. Jepsen and Torsten Dau, Danmarks Tekniske Universitet

Models of auditory signal processing and perception allow us to generate hypotheses that can be quantitatively tested, which in turn helps us to explain and understand the functioning of the auditory system. Here, the perceptual consequences of hearing impairment in individual listeners were investigated within the framework of the computational auditory signal processing and perception (CASP) model of Jepsen *et al.* [J. Acoust. Soc. Am., in press]. Several parameters of the model were modified according to data from psychoacoustic measurements. Parameters associated with the cochlear stage were adjusted to fit the basilar membrane input/output function estimated from forward masking experiments. The absolute sensitivity of the model was adjusted according to the pure-tone audiogram, and the variance of the internal noise in the model adjusted to predict measured just noticeable differences in intensity discrimination tasks. Simultaneous- and forward-masking experiments with noise maskers were used to test to what extent the model can account for the recovery from forward masking. Notched-noise masking was considered to test the model's ability to account for individual frequency selectivity. Three groups of listeners were considered: (a) normal hearing listeners; (b) listeners with a mild-to-moderate sensorineural hearing loss; and (c) listeners with a severe sensorineural hearing loss. A fixed set of model parameters were derived for each hearing-impaired listener. The simulations showed that, in most cases, the reduced or absent cochlear compression, associated with outer hair-cell loss, quantitatively accounts for broadened auditory filters, while a combination of reduced compression and reduced inner hair-cell function accounts for decreased sensitivity and slower recovery from forward masking. The model may be useful for the evaluation of hearing-aid algorithms, where a reliable simulation of hearing impairment may reduce the need for time-consuming listening tests during development.

9.15 AM SPECTRAL RESOLUTION AND INTERRUPTED SPEECH PERCEPTION

Su-Hyun Jin, Peggy B. Nelson and Chag Liu, University of Texas, Austin, TX,University of Minnesota, Twin Cities, MN, University of Texas, Austin, TX, USA

In a previous study of hearing-impaired (HI) listeners' speech perception in noise (Jin and Nelson, 2004), two factors emerged as highly related to amplified sentence recognition in the presence of modulated noise: low frequency audibility and auditory filter bandwidths. Nine young adult listeners with sensorineural hearing loss and eight young adults with normal hearing (NH) sensitivity as controls participated in the series of experiments. Amplified speech recognition performance of the HI listeners was equal to that of the NH listeners in quiet and in steady noise, but was significantly poorer in modulated noise. Thus, even when amplification was adequate for full understanding of speech in quiet and in steady noise, HI listeners experienced significantly less masking release from the modulated maskers.

The results indicated that those listeners with greatest hearing losses in the low frequencies were poorest at understanding amplified sentences in modulated noise. In addition, those HI listeners with wider auditory filters (in the 2k – 4k Hz region) were poorer than HI listeners with near-normal auditory filter bandwidths. These two findings are consistent with the hypothesis that strong spectral representation of voice pitch is necessary for auditory segregation of speech from noise (e.g., Qin and Oxenham, 2003). Additional results from HI and NH listeners will be presented; first, spectral resolution of HI and NH listeners was measured using the notched-noise method. This approach is to attempt to relate performance of current participants on psychophysical measures of spectral resolution to speech recognition. Second, we systematically vary the audibility of different frequency regions of speech by filtering. Sentences are interrupted by either speech-shaped noise or silence gap while measuring the percent of sentence recognition. The purpose of the current study is to examine contribution of different spectral regions to the auditory segregation/integration of interrupted speech. Implications for noise-reduction signal processing algorithms will be discussed. (This work was supported by a Summer Research Assignment program from the University of Texas at Austin.)

Thursday, August 14

SESSION TWO

Treatment Options for Conductive and Mixed Losses

Moderator: Karolina Smeds

11.10 AM **MECHANISMS OF BONE CONDUCTION AND ITS USE FOR HEARING IMPAIRED PERSONS**

Stefan Stenfelt, Linköping University, Sweden

Although bone conduction as a means to transmit sound to the hearing organ has been used for diagnose and (re)habilitation of persons with impaired hearing for nearly two centuries, its mechanisms are not fully understood. Because bone conduction hearing thresholds are relatively insensitive to the status of the middle ear, the general assumption is that direct stimulation of the cochlea (inertia of the cochlear fluids and alteration of the cochlear space) is the overall dominant contribution; for the healthy ear sound radiated in the outer ear canal, inertial effects of the middle ear ossicles, and sound pressure transmission from the cranial interior may also contribute to the perception of bone conducted sound. The use of bone conduction for amplified hearing has gained interest with the development of semi-implantable bone conduction hearing aid systems (eg. the Bone Anchored Hearing Aid). Such systems have shown great benefit for patients with conductive or mixed hearing losses or patients with a hearing impairment where use of conventional hearing aids are excluded due to ear canal problems. Bone conduction hearing aids differ from normal ear canal positioned hearing aids as they transmit the amplified sound to both cochleae (transcranial transmission). This transcranial transmission reduces the binaural benefit when the aids are fitted bilaterally but enables the usage of bone conduction hearing aids in unilateral deaf persons.

11.55 AM **PATIENT-DERIVED VERSUS AUDIBILITY-DERIVED FITTINGS IN BAHA USERS: A VALIDATION STUDY**

Bill Hodgetts, University of Alberta Sigfrid Soli, House Ear Institute, Bo Håkansson, Chalmers Technology University

Current approaches to fitting Baha rely heavily on patient feedback of “loudness” and “sound quality.” Audiologists are limited to this approach for two reasons: (1) the technology in current models of Baha does not allow for much fine-tuning of frequency response or maximum output on an individual basis, and (2) there has not been a valid approach to verifying the frequency response or maximum output on an individual basis.

To circumvent problem 2, we have developed a method of verifying speech audibility by measuring all hearing parameters (auditory dynamic range) and hearing aid parameters (aided output, MPO) in the same units at the same reference point: acceleration levels at the Baha abutment. This study addressed problem 1 through the use of a computer-controlled Master Baha hearing aid. There were two fitting approaches under investigation: Patient-Derived (PD) and Audibility-Derived (AD). For the PD fitting, the user's current Baha settings were matched with the Master Baha. For the AD fitting, a modified DSL m[i/o] fitting strategy was used (Scollie et al, 2005) to map all hearing aid output levels (in acceleration) into each user's dynamic range (in acceleration). The following parameters were under control on the Master Baha: frequency shaping (3 bands), compression (3 channels), overall gain and MPO.

Electro-mechanical testing revealed significantly better audibility with the AD fitting, especially in the high frequencies. Subjects were also tested on the following outcome measures: HINT (quiet and in noise), consonant recognition in noise, aided loudness, and subjective percentage of words understood. Subjects performed significantly better in all outcome measures with the AD fitting approach except when testing aided loudness and subjective perception at medium and high speech levels, where the differences were non-significant. Significant advantages for the AD fitting were found on these tests when the input level was soft.

Thursday, August 14

SESSION THREE

Cognitive Factors Influencing Speech Perception in Noise

Moderator: Virginia Best

5.15 PM **AIDED SPEECH RECOGNITION IN NOISE, PERCEIVED EFFORT AND EXPLICIT COGNITIVE CAPACITY**

Mary Rudner, Catharina Foo, Thomas Lunner, and Jerker Rönnberg,

The Swedish Institute for Disability Research, Linköping University, Sweden, Department of Behavioural Sciences and Learning, Linköping University, Sweden, Department of Medical and Experimental Medicine, Linköping University, Sweden, Oticon A/S, Research Centre Eriksholm, Snekkersten, Denmark

Speech recognition in noise is an effortful process requiring explicit cognitive processing. It may be influenced by level and type of noise and by the signal processing algorithms employed when hearing is aided. These complex relationships may be understood in terms of the working memory model for Ease of language Understanding (ELU, Rönnberg et al., in press). This model predicts that under challenging listening conditions, explicit cognitive processing demands will be high and that persons with good explicit cognitive capacity will be better listeners. Previous work has suggested that they may also find listening less effortful (Behrens et al., 2004; Larsby et al., 2005; in press). We studied this issue by including subjective effort ratings in a larger study designed to investigate aided speech recognition in noise and cognition. 32 experienced hearing aid users participated. Effort was rated using a visual analogue scale and the speech material was the Hagerman sentences presented in three fixed speech to noise ratios of +10 dB, +4 dB and -2dB. Effort was rated in modulated and unmodulated noise with fast and slow compression release settings, after each of two nine week training sessions with the same settings. Speech recognition performance was tested objectively under the same conditions using an adaptive procedure. Order of testing was balanced. Explicit cognitive capacity was measured using the reading span test. ANOVAs and correlations were computed. Preliminary results showed that decreasing SNR led to greater perceived effort and that the difference in perceived effort between the highest and the lowest SNR was greater in unmodulated noise than in modulated noise. Speech recognition performance in unmodulated noise generally correlated with effort ratings under similar conditions but in modulated noise generally it did not. Effort ratings correlated with reading span performance at the lowest SNR (-2dB) but only in unmodulated noise after the first training session. These preliminary findings show that subjective ratings of the effort involved in aided speech recognition covary with noise level and performance but that these effects are reduced by noise modulation. Further, the perceived effort of aided speech recognition at low SNR may be related to explicit cognitive capacity as measured by the reading span test. However, we only find evidence of this in unmodulated noise after the first training session. These findings extend previous work on perceived effort and cognitive capacity and provide further evidence that type of noise is an important factor in this relationship.

5.50 PM **UNDERSTANDING SPEECH IN NOISE WITH HEARING LOSS: MEASURES OF EFFORT**

Anastasios Sarampalis, Sridhar Kalluri, Brent Edwards, Ervin Hafter, University of California at Berkeley, Dept of Psychology, Berkeley, CA, Starkey Hearing Research Center, Berkeley, CA.

This paper investigates the hypothesis that listening effort is increased in the presence of noise and that digital noise reduction (NR) reduces effort with hearing-impaired listeners. It is well-documented that listeners with hearing impairment experience great difficulty understanding speech in noisy environments, even when amplification is provided. Traditional speech reception threshold (SRT) measures capture the difference between normal-hearing and hearing-impaired listeners in terms of information transmission, but are largely insensitive to the relative contributions of auditive and cognitive processes involved in speech communication. With this in mind, we have in the past reported a series of experiment that use a dual-task method to measure speech intelligibility scores as well as listening effort with normal-hearing listeners. The results from those experiments suggested that the presence of noise affects not only the ability to identify speech but also the ability to perform a simultaneous short-term memory or speed of processing task. Performance in these cognitive tasks improved as the signal-to-noise ratio (SNR) was increased from -6 to +2 dB. What is more, when a digital noise reduction (NR) algorithm was used to counteract the effects of noise, its effects were not in improving speech intelligibility, but in improving performance in the competing, cognitive task. We suggested that this was evidence in favor of the hypothesis that NR reduces listening effort in certain noisy situations and that this could explain anecdotal reports of NR being more comfortable. In the experiments presented here, we report results from the continuation of this work. More specifically, listeners with mild to moderate sensorineural hearing loss were tested in their ability to understand IEEE sentences at different SNRs and with or without NR. As before, listening effort was assessed using a dual-task method, with listeners performing a simultaneous, visual reaction time (RT) task. The results indicate that, just like with normal-hearing listeners, performance in the RT task was negatively affected by the presence of noise. Unlike with normal-hearing listeners, however, the effect was much greater, and largely unaffected by SNR or NR processing. These results are in line with the hypothesis that with hearing loss (and indeed aging) there is greater reliance on top-down processing when listening to speech in noise.

6.25 PM **PERFORMANCE PERCEPTUAL TEST (PPT) AND THE ACCEPTABLE NOISE LEVEL (ANL): WHAT DO THEY MEASURE?**

Gabrielle Saunders, National Center for Rehabilitative Auditory Research (NCRAR), Portland VA Medical Center, Portland, Oregon.

Conventional measures of speech in noise are often not well correlated to hearing aid use or to reported satisfaction, benefit or residual reported activity limitation or participation restriction. Two newer measures, the Performance Perceptual Test (PPT, (Saunders, Forsline, & Fausti, 2004)) and the Acceptable Noise Level (ANL, (Nabelek, Tampas, & Burchfield, 2004) have been developed in an effort to combine objective speech understanding with subjective perception. The PPT measures a signal-to-noise ratio (SNR) at which listeners can repeat back sentences presented in noise (the Performance condition) and a SNR at which listeners believe they can understand speech in noise (the Perceptual condition). By subtracting the Perceptual SNR from the Performance SNR the discrepancy between measured and perceived ability to hear is obtained. This value, known as the PPDDIS, provides a variable that has been shown to provide information additional to that provided by conventional measures of speech in noise. The ANL is a procedure to quantify listener's willingness to accept background noise in the presence of speech. Similar to the Perceptual condition of the PPT, listeners set the level of noise they can tolerate while listening to ongoing speech.

In this study, both the PPT and ANL were measured, along with a variety of hearing questionnaires, including the International Outcome Inventory (IOI-HA, the Abbreviated Profile of Hearing Aid Benefit (APHAB) and the Hearing Handicap Inventory for the Elderly/Adults (HHIE/A) to assess the relationships between these tools. Data from seventy-four participants with hearing impairment will be presented that describes the various relationships between the ANL, PPT and questionnaire subscales.

Nabelek, A., Tampas, J., & Burchfield, S. (2004). Comparison of speech perception in background noise with acceptance of background noise in aided and unaided conditions. *Journal of Speech , Language and Hearing Research, 47*, 1001-1011.

Saunders, G., Forsline, A., & Fausti, S. (2004). The Performance-Perceptual Test (PPT) and its relationship to unaided reported handicap. *Ear and Hearing, 25*, 117-126.

FRIDAY, AUGUST 15

SESSION FOUR

The Legacy of Stuart Gatehouse

Moderator: Michael Akeroyd

INTRODUCTION by Graham Naylor

8.00 AM **STUART GATEHOUSE'S LEGACY FOR HEARING RESEARCH: PAST, PRESENT, AND FUTURE**

Quentin Summerfield, Department of Psychology, University of York, Heslington, York YO10 5DD, UK

Stuart Gatehouse appreciated that auditory handicap arises from the interplay between ear, brain, and environment, and that influence over policy-makers is achieved by engaging them on their own terms. I will illustrate these characteristics with two projects which originated in discussions with Stuart and which have been taken forward since his death.

Policy makers want the clinical effectiveness of interventions to be demonstrated using generic scales that can be applied to all diseases and disabilities. Such scales include a questionnaire which maps a patient onto a multi-dimensional state of health which has previously been valued by members of the public using a formal technique such as the time trade-off. The preferred scale in the UK, the EuroQol EQ-5D, is largely insensitive to hearing loss, even in its most extreme form. Stuart and I hypothesised that the limitation arises primarily from the omission of the ability to communicate as a key concomitant of well-being. We extended the EQ-5D to include dimensions relevant to communication and we demonstrated that hearing-impaired people value the additional dimensions. Now, Michael Akeroyd and I have shown that members of the public also value the additional dimensions using the time trade-off. This work, therefore, starts to rectify the inequity wherein hearing-impaired people must compete for resources within systems of health-care, despite hearing loss being only rarely a manifestation of ill health.

Stuart demonstrated that individual differences in self-reported difficulties in listening are explained not only by variation in hearing sensitivity but also by variation in attention, both auditory and visual. Padraig Kitterick and I have extended this work by demonstrating that performance on a demanding task that requires participants to switch attention among multiple concurrent talkers correlates not only with hearing sensitivity, self-reported difficulties in everyday listening, and measures of visual attention, but also with the power in neuro-magnetic signals generated in the brain at key moments when attention must be focused and distraction must be resisted. Potentially, these measures of brain activity could provide indices of success in implementing Stuart's suggestion that interventions to arrest or reverse the age-related decline in attentional capacities could form a component of auditory rehabilitation in addition to the provision of amplification.

[Work in Glasgow supported by MRC UK and in York by RNID and Deafness Research UK.]

8.45 AM A SELF-RATING MEASURE WITH EMPHASIS ON BINAURAL HEARING FUNCTION: THE SPEECH, SPATIAL, AND QUALITIES OF HEARING SCALE

William Noble, University of New England, Australia

Why develop a self-rating scale in a domain such as binaural hearing function which is traditionally covered using laboratory test techniques? Two arguments are offered: 1) only through direct inquiry of, for example, people with impaired hearing, can the consequences for binaural function in everyday terms be assessed; 2) self-ratings provide a cost-effective way to find out about connections among auditory functions. This can lead to development of functional test designs that allow those connections to be better understood. The Speech, Spatial and Qualities of Hearing scale (SSQ) was developed with Stuart Gatehouse and initial results demonstrated its utility in showing ways in which dynamic aspects of spatial hearing connect with dynamic aspects of speech hearing. This result has had valuable influence on the design of a dynamic test of masking. The scale has also provided insights about where benefits do and do not lie in provision of bilateral amplification.

Recent applications of the SSQ are described, that demonstrate the sensitivity of the measure to small differences in overall hearing ability, and to different forms of masking (energetic versus informational). These data can help form the basis of a future "normal binaural hearing standard". An extension of the scale to inquire about aspects of size and speed discrimination will be described, with preliminary data from a normal hearing and a cochlear implant sample.

9.15 AM MEASURING QUALITY IN AUDIOLOGY: A GLOBAL (OR GATEHOUSE) FRAMEWORK

Adrian Davis, Martin Evans, Pauline Smith, Margaret Martin
MRC Hearing and Communication Group, Manchester University

Abstract
A standards based framework for audiology services has been developed over the last few years with input from professionals, clients, patients and others in the UK. Stuart Gatehouse was instrumental in starting this programme of work to devise and put in place a framework against which it might be possible to assess the quality and performance of adult and paediatric audiology services. This framework has been piloted and validated in the first national audit of audiology services conducted in Scotland. The audit has had considerable impact on policy and investment. The framework enables self-assessment and tracking of quality and performance over time.

Background
Clinical and research audiology programmes traditionally focus on patient outcomes, e.g. 'Glasgow Hearing Aid Benefit Profile'. However, working initially with the late Professor Gatehouse, we developed a quality framework against which quantitative measures could be made. We piloted this framework across all adult and some paediatric services in Scotland. We tested whether the framework was valid and sensitive in showing if services were responsive to needs, empowered patients/parents to be good partners in meeting those needs. It also enabled services to establish whether they made the best use of staff skills and resources.

Design
The audit was undertaken by a multidisciplinary team across all adult services in Scotland and half the paediatric services. The process followed the patient journey and consisted of three stages: a **questionnaire** survey, self assessment using a **quality rating tool** devised by the MRC Hearing & Communication Group, followed by on-site **visits** to verify the responses.

We developed
- new draft standards against which services could be measured
- the questionnaire and visit protocol
- the quality rating tool and framework.

Results and conclusions
The audit resulted in 169 specific recommendations for improvement in clinical practice, local service organisation, leadership and skills development. The impact of operational targets on audiology services often resulted in pressure to sacrifice quality and a "one size fits all" service. The

quality rating tool used was a useful way to benchmark quality, identify areas of improvement and indicate value for money. We think it is ready to be adapted in other countries and across other areas of services.

FRIDAY, AUGUST 15

SESSION FIVE

Why You Lose It and What Happens When You Do

Moderator: Jan Wouters

11.10 AM THE GENETICS OF HEARING LOSS: A QUIET REVOLUTION

Karen B. Avraham, Department of Human Molecular Genetics and Biochemistry, Sackler School of Medicine, Tel Aviv University, Tel Aviv, Israel

The past two decades have brought remarkable advances in our understanding of the mechanisms governing inner ear function, built on studies in comparative physiology and anatomy. We now know the DNA sequences of 45 different genes responsible for nonsyndromic hearing loss and 32 genes responsible for syndromic hearing impairment. At least 105 additional genes responsible for hearing are known to exist and geneticists are searching actively for their identities. Hearing is probably the human trait best defined by modern genomic analysis. Discovery driven by genomics offers the opportunity for molecular diagnosis of hearing impairment, and in turn, appropriate choice of therapy. Each known gene harbors many different mutations. Many of these mutations are private and therefore specific to one family. For increasing numbers of affected individuals, we understand the connection between the mutation at the level of DNA and the nature of the hearing loss. Early detection of hearing loss can guide the choice of therapy, including hearing aids or cochlear implants. Precise genetic characterization enables a far greater understanding than in the past of whether a child will, or will not, develop syndromic features that accompany some forms of hearing loss. Genomic techniques are being developed to screen genes more effectively. This ‘quiet revolution’ will continue to change our perception of hearing loss, and can alter the appropriate care for children and adults with hearing impairment.

11.55 AM THE ROLE OF TEMPORAL FINE STRUCTURE IN PITCH AND SPEECH PERCEPTION BY PEOPLE WITH NORMAL AND IMPAIRED HEARING

Brian C. J. Moore, Department of Experimental Psychology, University of Cambridge, Downing Street, Cambridge CB2 3EB, England

Any complex sound that enters the normal ear is decomposed by the auditory filters into a series of relatively narrowband signals. Each of these signals can be considered as a slowly varying envelope (E) superimposed on a more rapid temporal fine structure (TFS). I consider the role played by TFS in a variety of psychoacoustic tasks. I argue that cues derived from TFS may play an important role in the ability to "listen in the dips" of a fluctuating background sound. TFS cues also play a role in pitch perception, the ability to hear out partials from complex tones, and sound localisation. Finally, and perhaps most importantly, TFS cues may be important for the ability to hear a target talker in the spectral and temporal dips of a background talker.

Evidence will be reviewed suggesting that cochlear hearing loss reduces the ability to use TFS cues for both pitch perception and speech perception. The perceptual consequences of this, and reasons why it may happen, will be discussed. Finally, possible applications of these findings to the choice of compression speed in hearing aids will be discussed.

This work was supported by the Medical Research Council (UK).

FRIDAY, AUGUST 15

SESSION SIX

Physiological and Physical Factors Relevant to Auditory Function in Normal and Impaired Hearing

Moderator: Joanna Robinson

5.15 PM COCHLEAR NERVE DEGENERATION AFTER "TEMPORARY" NOISE-INDUCED HEARING LOSS

Sharon Kujawa and M. Charles Liberman, Massachusetts Eye and Ear Infirmary, Boston, MA

The presence of excitotoxic damage to cochlear nerve terminals under inner hair cells (IHCs) has long been recognized as an acute result of acoustic overexposure. However, it has been assumed that such damage is reversible. We have recently shown, in both mice and guinea pigs, that noise exposures titrated to produce threshold shifts at the limits of reversibility actually cause loss of up to 50% of the afferent synapses on IHCs within a few days post-exposure, despite complete recovery of otoacoustic emission thresholds and suprathreshold responses by 1 week post-exposure. Within a few months post-exposure, degeneration of up to 50% of the spiral ganglion cells is seen, despite no loss of inner or outer hair cells. Thresholds for wave 1 of the ABR also return to normal within 1 week post-exposure, whereas suprathreshold amplitudes are reduced by more than 50%, reflecting the primary neuronal degeneration. Results suggest that ABR thresholds are a poor metric of primary neurodegeneration and that existing exposure guidelines may seriously underestimate the deleterious effects of noise.

6.00 PM **HEARING AID GAIN PRESCRIPTIONS BALANCE RESTORATION OF AUDITORY NERVE MEAN-RATE AND SPIKE-TIMING REPRESENTATIONS OF SPEECH**

Faheem Dinath and Ian C. Bruce, McMaster University

Linear and nonlinear amplification schemes for hearing aids have thus far been developed and evaluated based on perceptual criteria such as speech intelligibility, sound comfort, and loudness equalization. Finding amplification schemes that optimize all of these perceptual metrics has proven difficult. Using a physiological model, Bruce et al. (ISAAR 2007) investigated the effects of single-band gain adjustments to linear amplification prescriptions. Optimal gain adjustments for model auditory-nerve fiber responses to speech sentences from the TIMIT database were dependent on whether the error metric included the spike timing information (i.e., a time-resolution of several microseconds) or the mean firing rates (i.e., a time-resolution of several milliseconds). Results showed that *positive* gain adjustments are required to optimize the mean firing rate responses, whereas *negative* gain adjustments tend to optimize spike timing information.

The discrepancy in optimal gain adjustments between the two neural representations is not clear and warrants further investigation. In particular, it is difficult from visual inspection of the spike train patterns to determine in exactly what ways the hearing impairment, prescribed linear amplification and gain adjustments affect the spike-timing information. Possible contributions include: i) changes in which frequency components of speech auditory nerve fibers are synchronizing to, and ii) changes in the phase of these synchronized responses.

In this study we examine this issue in more depth using a similar optimization scheme applied to a synthetic vowel /ɛ/. The periodic nature of the synthesized vowel allows for quantitative analysis of synchronization and phase responses to different frequency components of the vowel. It is found that negative gain adjustments (i.e., below the linear gain prescriptions) minimize the spread of synchrony and deviation of the phase response to vowel formants in responses containing spike-timing information. In contrast, positive gain adjustments (i.e., above the linear gain prescriptions) normalize the distribution of mean discharge rates in the auditory nerve responses. Thus, linear amplification prescriptions appear to find a balance between restoring the spike-timing and mean-rate information in auditory-nerve responses. Furthermore, the negative gain adjustments found to optimize the spike-timing representation of speech in these studies are similar to the gain adjustments made by wide dynamic range compression algorithms. This suggests that loudness recruitment, and the resulting need for compression algorithms in hearing aids, is determined by the spike-timing response of the auditory nerve rather than the mean discharge rate.

[This work was supported by NSERC Discovery Grant 261736.]

6.25 PM **USE OF FORWARD PRESSURE LEVEL (FPL) TO MINIMIZE THE INFLUENCE OF ACOUSTIC STANDING WAVES DURING PROBE-MICROPHONE HEARING AID VERIFICATION**

Ryan W. McCreery, Andrea L. Pittman, Stephen T. Neely, and Patricia G.Stelmachowicz, Boys Town National Research Hospital, Omaha, NE, USA

Probe-microphone measurements provide audiologists with a valid and reliable method of verifying hearing aid sound pressure level in the ear canal for frequencies between 0.25 and 4 kHz. The presence of acoustic standing waves in the ear canal and the individual variability in ear canal acoustics combine to reduce the validity and reliability of probe microphone measurements for frequencies above 4 kHz. Recent data suggest that speech information at frequencies up to 10 kHz may enhance speech perception, particularly for children (Stelmachowicz et al., 2007). As manufacturers begin to extend the upper bandwidth of hearing aids, there is a crucial need to develop accurate measures of sound pressure level (SPL) at frequencies above 4 kHz. Farmer-Fedor and Rabbitt (2002) suggested a method for separating the incident and reflected components of acoustic intensity in the ear canal and recommended the use of incident (or forward) intensity as a measure of stimulus level. A similar approach, forward pressure level (FPL), is theoretically unaffected by standing waves and has been successfully applied to DPOAE measurements in the ear canal (Scheperle et al., in press). In Part I of the present study, conventional SPL measurements were completed with probe placements at the eardrum and at 2 mm distal to the eardrum and were compared to SPL measurements taken at 4 mm and 2 mm from the end of the earmold in 10 adult subjects. Although the within subject test-retest reliability was good, probe microphone measures in SPL at the eardrum reduced the presence of standing waves up to 10 kHz in only 2 of the 10 participants. In most subjects, probe placement at the eardrum simply shifted the acoustic minima upward in frequency, and did not eliminate the presence of standing waves. Part II of the present study was to determine if the use of FPL in probe microphone measurements can reduce the presence of standing waves in the ear canal for frequencies up to 10 kHz. Preliminary data from a group of normal-hearing children using FPL for high-frequency probe-microphone measurements will be presented.

SATURDAY, AUGUST 16

SESSION SEVEN

Current and Future Trends in Signal Processing for Hearing Aids

Moderator: Brent Edwards

8.00 AM **COMPUTATIONAL AUDITORY SCENE ANALYSIS AND ITS POTENTIAL APPLICATION TO HEARING AIDS**

DeLiang Wang, The Ohio State University

The acoustic environment is typically composed of multiple simultaneous events. A remarkable achievement of auditory perception is its ability to disentangle the acoustic mixture and group the sound energy that originates from the same event or source. This process of auditory organization is referred to as auditory scene analysis (ASA). Psychoacoustic research in ASA has motivated the study of computational auditory scene analysis (CASA), which aims at sound source separation based on ASA cues, including pitch, location, amplitude/frequency modulation, and onset/offset. This presentation will give an overview of CASA, in particular research on speech segregation. A typical CASA system produces a binary time-frequency mask as its output, which attempts to retain time-frequency regions of the acoustic mixture where target speech dominates and discard other regions where interference dominates. The presentation will also discuss recent work on speech intelligibility evaluation of binary time-frequency masks with both normal-hearing and hearing-impaired listeners. These evaluation results demonstrate the promise of CASA for improving speech intelligibility in noise.

8.45 AM **FUTURE TRENDS IN HEARING INSTRUMENT TECHNOLOGY**

Stefan Launer, Phonak AG, 8712 Staefa, Switzerland

Digital technology offers many possibilities to optimally fit hearing instruments to the individual listening needs of hearing impaired persons. Today's hearing instruments contain many different adaptive control functions which automatically adapt the hearing instruments operational parameter settings according to the requirements of the respective acoustic environment.

Identifying the acoustic environment for selecting the optimal signal processing strategy requires an intelligent decision making process about the acoustic environment relying on different physical characteristics of the sound field. The automatic identification of the acoustic environment is a very important pre-requisite for the application of sound cleaning features, i.e. signal processing techniques improving the listener's communication abilities in adverse listening conditions. Such techniques include the adaptive multi-microphone technology which significantly improves the performance of noise reduction systems. Latest technologies also include means for reducing the very detrimental effect of reverberation on speech intelligibility especially for hearing impaired people. Recent studies clearly show an improvement in subjective and objective speech intelligibility in a variety of difficult listening situations. Recently Frequency Compression has been successfully re-introduced into hearing instruments specifically improving speech intelligibility and speech production in subjects with profound hearing losses. Furthermore wireless links between left/right hearing instruments and also to external devices have been introduced opening up a range of new applications. Finally, hearing instrument technology has significantly made progress regarding the mechanical design, robustness and wearing comfort offering new solutions for age old problems such as occlusion and cosmetic appearance.

These technologies are being applied in today's hearing instruments and have been shown to be very helpful to the end user in a number of studies. The goal of this talk is to discuss the state of the art focusing on user benefit of modern hearing instruments and furthermore to discuss the perspectives of hearing instrument technology.

9.20 AM **EFFECTS OF FREQUENCY LOWERING IN WEARABLE DEVICES ON FRICATIVE AND AFFRICATE PERCEPTION**

Joshua M. Alexander, Dawna E. Lewis, Judy G. Kopun, Ryan W. McCreery, and Patricia G. Stelmachowicz, Boys Town National Research Hospital, Omaha, NE, USA

Work by Stelmachowicz and colleagues demonstrate that limited hearing aid bandwidth prevents useful high-frequency speech information from being transmitted. This is especially problematic for children with mild-to-moderate hearing loss who are learning speech and language. Because conventional hearing aids limit audible bandwidth to 5-6 kHz, children with hearing loss often cannot hear their own productions of some fricatives and affricates, which can have energy out to 9-10 kHz. The purpose of this study is to test the efficacy of frequency lowering features in devices currently on the market in adults with mild-to-moderate hearing impairment and in normal-hearing controls.

Participants listened monaurally through headphones to recordings of nine fricatives and affricates spoken by three women in a vowel-consonant (VC) context. Stimuli were mixed with speech-shaped noise at a 10 dB SNR. During recording, VCs were preceded by a 2-second speech-in-noise carrier and presented at 62 dB SPL. Recordings were made offline with a probe microphone in a 2cc coupler coupled to Widex's Inteo (IN-9) and Phonak's Naída (V) BTE hearing aids. Both devices were set to DSL-Adult prescriptive targets with all advanced features off. Frequency lowering in the Inteo works by identifying peaks in the high-frequency spectrum, which are then transposed down one octave. Frequency lowering in the Naída works by nonlinearly compressing a high frequency band to a lower frequency range. Frequency lowering in both devices was set to occur for spectral components above 4 kHz, but neither device provided a full lowering of the VCs 10-kHz bandwidth. Each device was tested under four conditions.

Both devices had control, wideband, and frequency lowering conditions. For the control condition, recordings were made with frequency lowering turned off. The wideband condition was constructed by mixing recordings of control stimuli with high-pass filtered versions of the original stimuli so that gain at 5-10 kHz was approximately equal to the DSL target at 4 kHz. For the Inteo, the fourth condition consisted of recordings made with settings the same as the control, but with the noise reduction feature turned on, because this feature cannot be disengaged when in transposition mode. For the Naída, the fourth condition was the same as the control except input stimuli were first processed by a custom frequency compression algorithm that allowed for a complete lowering of the 4-10 kHz input range within the amplified range of the device. Results from an informational analysis of feature errors will be presented.

SATURDAY, AUGUST 16

SESSION EIGHT

Large-Scale Studies of Signal Processing and Auditory Function

Moderator: Pamela Souza

11.10 AM EVALUATION OF SIGNAL ENHANCEMENT STRATEGIES FOR HEARING AIDS: A MULTICENTER STUDY

Heleen Luts, Koen Eneman, Sofie Jansen, Jan Wouters, Michael Büchler, Norbert Dillier, Wouter Dreschler, Matthias Froehlich, Giso Grimm, Niklas Harlander, Volker Hohmann, Rolph Houben, Arne Leijon, Anthony Lombard, Dirk Mauler, Marc Moonen, Henning Puder, Michael Schulte, Ann Spriet, Matthias Vormann
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Over the past decades many digital signal enhancement techniques have been developed. However, most of these techniques have been evaluated only in a limited way. Within the framework of the European HearCom project different signal enhancement strategies were studied and five promising techniques were selected and further developed for future use in hearing aids: single-channel noise suppression based on perceptually optimized spectral subtraction, Wiener-filter-based single-channel noise suppression, broadband blind source separation, multi-microphone adaptive beamforming based on multi-channel Wiener filtering and binaural coherence dereverberation filtering. All algorithms were evaluated in a wide variety of listening environments with a set of advanced physical performance measures. Five signal processing groups collaborated in these developments. The selected and optimized algorithms were implemented on a single common real-time hard- and software test platform using the Master Hearing Aid (MHA) and custom-designed hearing aids. This facilitates a thorough comparative perceptual validation of the algorithm performance.

For the perceptual evaluation a multicenter study has been set up. Perceptual tests are carried out at five different test-sites in Belgium, the Netherlands, Germany and Switzerland. In total 150 subjects with different hearing profiles will participate: 50 normal-hearing subjects, 50 hearing-impaired subjects with a flat perceptual hearing loss and 50 hearing-impaired subjects with a sloping perceptual hearing loss. Adaptive speech reception tests, as well as listening effort and preference rating are assessed. Tests and retests are carried out in two types of acoustical environments, a living room and a highly reverberant room. Speech is presented in three different listening scenarios: with one interfering noise source, with three uncorrelated noise sources and in quiet. For both noise conditions multitalker babble of 65 dB(A) is used.

First results of the listening tests indicate that for the scenario with three interfering noise sources large differences across different strategies are obtained. The largest improvement in speech intelligibility relative to the unprocessed condition is obtained with the multi-microphone adaptive beamforming algorithm. Certain algorithms appear to offer limited intelligibility improvements in the tested scenarios, or sometimes even fail to enhance the speech intelligibility.

The unique feature of this study, compared to similar evaluation efforts, is the intra-subject comparison by listeners with different hearing profiles of real-time implementations of a number of advanced signal enhancement techniques. Results of this completed study will be presented.

[This work was supported by grants from the European Union FP6 Project 004171 HearCom.]

11.45AM **EVALUATION OF THE 'AUDITORY PROFILE' TEST BATTERY IN AN INTERNATIONAL MULTI-CENTRE STUDY**

Thamar van Esch, Wouter Dreschler, AMC Clinical & Experimental & Audiology, Amsterdam, NH

HEARCOM (Hearing in the Communication Society) is a European project that aims at reducing limitations in auditory communication caused either by hearing loss or by poor environmental conditions. Two of the focus areas of HEARCOM are on the identification and characterization of auditory communication limitations and on the development of standardized testing and evaluation procedures for hearing-impaired persons. In this context, an 'Auditory Profile' will be developed. This auditory profile should be a diagnostic tool that complements the pure-tone audiogram and can be assessed for each individual listener, using a standardized battery of audiological tests. It will be assessable in (specialized) hearing centers or clinics or in audiological research. It will be used to characterize the individual's auditory impairment profile in a comparable way within Europe, and should be usable for a broad population of hearing impaired subjects.

Therefore, the auditory profile should include all necessary measures to describe details and differences between different hearing impairments, but should minimize redundancy between measures.

Within HEARCOM, a preliminary auditory profile has been defined, that contains measures of loudness perception, spectral and temporal resolution, speech perception, cognition, binaural hearing, and subjective judgments, including listening effort. This preliminary auditory profile has been evaluated in an international multi-center study in which approximately 100 subjects (both normally hearing and hearing impaired) were measured in five research centers in Germany, Sweden, UK and The Netherlands. All tests of the preliminary auditory profile were conducted in test and retest on different days.

The focus of the multi-centre field trial was to investigate the clinical applicability (test-retest reliability, reproducibility, learning effects) and the relevance of the test results to communication problems. In this presentation, results of the multi-centre study will be presented. We investigated effects of test-retest and learning effects, effects of ears (left or right, poorer or better), and effects of center/language. Subsequently we examined the per-ear variables (outcomes of tests that were conducted monaurally) in hearing-impaired listeners (e.g. factor analysis, regression analysis) and selected the most relevant per-ear variables. In a similar manner we also selected the most relevant per-subject variables (outcomes of all other objective tests). Next we evaluated the communication-performance outcome measures and their relation to the selected per-ear and per-subject variables. The results show a clear pattern of underlying factors that will help us to select the most relevant tests and parameters for a broad clinical application of the auditory profile. This approach will be validated in a second multi-center study. [The authors gratefully acknowledge the contributions of Hörzentrum Oldenburg; Institute for Sound and Vibration Research, Southampton; Linköping University; VU University Hospital, Amsterdam]

SATURDAY, AUGUST 16

SESSION NINE

Occlusion and Own-Voice Perception

Moderator: Michael Stone

5.15 PM ACOUSTICALLY TRANSPARENT HEARING AIDS: AN ELECTRONIC VENT FOR HEARING AIDS

Jorge Mejia, Harvey Dillon, and Michael Fisher, National Acoustic Laboratories and CRC Hear, 126 Greville Street, Chatswood NSW 2067, Australia

The occlusion effect can be described as the low-frequency amplification of own-voice bone conducted sounds occurring inside blocked ear canals. This amplification arises from differential vibrations of the jaw relative to the skull. It is commonly reported as unnatural and annoying and can deters usage of hearing aid devices. A novel active vent was investigated as a strategy to increase the wearability of hearing aids by reducing the level of the bone-conducted sounds present directly inside the ear canal. With a hearing aid including an active vent level reduction ranging from 10 to 18 dB, listeners rated their own voices as significantly better and more natural than when provided with a passive 1 mm vented hearing aid. In addition, for minimally vented hearing aids, the active vent strategy also cancels the passive vent-transmitted sounds, thus greatly widening the frequency range over which directional microphones and adaptive noise suppression systems operate.

6.00 PM A LARGE-SCALE SUBSTANTIATION OF OWN-VOICE ISSUES IN HEARING-AID USERS, PART II: REDUCING OCCLUSION PROBLEMS IS STILL IMPORTANT

Niels Søgaard Jensen, Søren Laugesen, Patrick Maas, Marie Louise Kamp González Cárdenas, and Sidsel Mørch Rysager, Eriksholm, Oticon Research Centre, Oticon España S.A., and ViSP, Resource Centre for Special Needs Education.

In a companion presentation (part I), Laugesen et al. report on a questionnaire study (utilizing the Own Voice Qualities (OVQ) questionnaire) where the main hypothesis under test was that hearing-aid users have other issues and concerns related to their own voice besides the well-known problems caused by occlusion. This hypothesis was strongly confirmed by the questionnaire data.

In the same study, a secondary hypothesis was that hearing-aid users who are exposed to occlusion (due to an unfortunate combination of hearing loss and vent size) will experience more own-voice issues than hearing-aid users, who are not exposed to occlusion. Accordingly, the 187 participating hearing-aid users were recruited so one third could be included in a group expected to suffer from occlusion problems (due to small low-frequency hearing losses and small hearing-aid vents) while the remaining two thirds could be included in a group not expected to suffer from occlusion problems (due to either large low-frequency hearing losses or large hearing-aid vents).Surprisingly, the questionnaire data did not support the secondary hypothesis. The group expected to suffer from occlusion did not in fact report about more own-voice issues than the other group. Rather than questioning the evidence that open hearing-aid fittings provide major improvements on occlusion-related issues, the data indicate that 'self-selection' has played a significant role in the recruiting of test subjects, since all test subjects evaluated the own-voice perception with their own hearing aids. It is therefore quite likely that the test subjects who actually had decided to buy and use small-vent hearing aids are people who are simply not bothered by occlusion.

These results led to a follow-up study where 43 test subjects with small low-frequency hearing losses compared open fittings with small-vent fittings (using the same type of hearing aid) in a balanced cross-over design. Each type of fitting was used for a period of one month before the OVQ questionnaire was filled in. The data showed that significantly more own-voice issues were reported with small-vent fittings than with open fittings. This finding supports both the secondary hypothesis (i.e., reducing occlusion problem reduces own-voice issues) and the explanation for the observations made in the first study. Data from both studies will be presented and discussed.

6.25 PM THE EFFECT OF OPENNESS OF THE FITTING ON THE RELATIVE LOUDNESS PERCEPTION OF LOW AND HIGH FREQUENCY SOUNDS

Gitte Keidser, Anna O'Brien, Ingrid Yeend, and Lisa Hartley, National Acoustic Laboratories

This study was sponsored by Siemens Instruments, Germany.

In a loudness balancing test conducted in the late 1990s (Keidser et al., 2000) it was found that normal-hearing listeners selected, on average, 10 dB higher levels at the eardrum when listening to a 0.5 kHz octave band noise with the ear occluded than when listening with the open ear. Loudness was referenced to that of a 1.5 kHz octave band noise. Similar levels were selected when balancing loudness of a 3.0 kHz octave band noise to that of the reference stimulus. No physical explanation for the discrepancy has been found, but if the finding is transferable to hearing-impaired listeners, the effect could have consequences for hearing aid prescriptions when using hearing aids with different degree of openness.

The aim of this study is to investigate the effect of openness of the hearing aids on the relative loudness perception of low and high frequency sounds. Ten listeners with normal hearing and 24 hearing-impaired listeners are balancing loudness of a 0.5 and 3.0 kHz pure tone to that of a 1.5 kHz pure tone under three conditions. In two of the test conditions, the tones are generated in the hearing device and presented to the ear canal via an open dome and a closed dome mould. The tones are also presented to the ear occluded with a foam ear tip using an insert headphone. In all cases levels are measured with a probe tube microphone positioned within 6 mm from the eardrum. The level of the reference stimulus is 73 dB SPL for the normal-hearing listeners and the NAL-NL1 prescribed REAG for a 65 dB SPL input for hearing-impaired listeners. Preliminary data suggest that for both subject groups the level difference for equal loudness of the 0.5 and 3.0 kHz tones is greater when listening through the open mould than when listening through the insert tip and closed mould. However, the difference in the equally loud levels for the low- and high frequency sounds presented through the open and closed moulds is less prominent for the hearing-impaired than for the normal-hearing listeners. The full data set will be presented and discussed, including the implication for open-ear fittings.

Reference:

Keidser G, Katsch R, Dillon H, and Grant F. (2000) Relative loudness perception of low and high frequency sounds in the open and occluded ear. JASA 107(6): 3351-3357.

SUNDAY, AUGUST 17

SESSION TEN

Auditory Learning and Training

Moderator: Christian Füllgrabe

8.00 AM **IMPROVING AIDED SPEECH COMMUNICATION THROUGH AUDITORY TRAINING: A REVIEW OF CURRENT APPROACHES AND FUTURE APPLICATIONS**

Larry Humes & Matthew Burk, Indiana University

Auditory training and aural rehabilitation techniques have been at the forefront of the audiology literature in recent years, but the history of aural rehabilitation programs dates back several decades. Although there were exceptions, many of the early auditory-training programs for hearing-aid wearers focused on children with severe or profound hearing loss. The majority of hearing aids in the U.S. and many other countries, however, are sold to individuals over 60 years of age and several more recent auditory-training methods have targeted this age group. The degree of hearing impairment in older adults is typically less than that of the children originally targeted by auditory-training programs in the past, but there is also greater likelihood for concomitant central or cognitive deficits in older adults. Further, with advances in hearing aids, it would seem that auditory training would be less necessary now than in years past, particularly as directional microphones and noise-reduction algorithms have improved. However, the opposite may hold true as older listeners are more active than ever and expect more from their hearing-aids as digital technologies improve. This presentation will provide a broad overview of past and present auditory-training methodologies and principles, as well as encouraging data from one particular lexically based training program under development at the Audiology Research Laboratory at Indiana University. (This work was supported, in part, by NIH research grant R01 AG08293.)

8.45 AM **INTERVENTION FOR RESTRICTED DYNAMIC RANGE AND REDUCED SOUND TOLERANCE: CLINICAL TRIAL USING MODIFIED TINNITUS RETRAINING THERAPY**

Monica L. Hawley, LaGuinn P. Sherlock, Susan Gold, Craig Formby University of Maryland Tinnitus and Hyperacusis Center, University of Maryland School of Medicine, Baltimore, MD, Department of Communication Disorders, University of Alabama, Tuscaloosa 35487.

Hyperacusis is the intolerance to sound levels that normally are judged acceptable to others. The presence of hyperacusis (diagnosed or undiagnosed) can be an important reason that some persons reject their hearing aids. Tinnitus Retraining Therapy (TRT), originally proposed for the treatment of persons with debilitating tinnitus, offers the significant secondary benefit of increased Loudness Discomfort Levels (LDLs) in many persons. TRT involves both counseling and the daily exposure to soft sound from bilateral noise generator devices (NGs). We implemented a randomized, double-blind, placebo-controlled clinical trial to assess the efficacy of TRT as an intervention to improve sound tolerance in hearing-aid eligible persons with hyperacusis and/or restricted dynamic ranges. Subjects were assigned to one of four treatment groups: 1) full treatment, both counseling and NGs, 2) counseling with placebo NGs, 3) NGs without counseling, and 4) placebo NGs without counseling. They were evaluated at least monthly, typically for five months or more, on a variety of audiometric tests, including LDLs, the Contour Test for Loudness for tones and speech, word recognition measured at each session's comfortable and loud levels, and with electrophysiological measures. Over 80% of the subjects assigned to full treatment achieved significant treatment effects (defined as shifts of greater than 10 dB in LDL or Contour Test for Uncomfortable Level), whereas most subjects assigned to a partial treatment group did not meet these criteria. Case studies of hearing aid use by these subjects will also be presented. Supported by NIH R01 DC04678.

SUNDAY, AUGUST 17

SESSION ELEVEN

Perceptual and Physical Measures of Hearing Aid Signal Processing

Moderator: Leonard Cornelisse

9.30 AM **EFFECTS OF LINEAR, NONLINEAR AND COMBINED LINEAR AND NONLINEAR DISTORTION ON PERCEIVED SPEECH QUALITY**

Kathryn H. Arehart, James M. Kates, Melinda Anderson, Lewis O. Harvey, Jr., CU Boulder, GN ReSound and CU Boulder, CU Boulder, CU Boulder

The purpose of this experiment was to measure subjective quality ratings for speech subjected to signal processing typical of that encountered in real hearing aids. Using a hearing aid simulation programmed in MAT-

LAB, the experiment quantified perceived quality of speech subjected to a) noise, distortion, and nonlinear processing, b) linear filtering, and c) combinations of noise, distortion, and linear processing. Quality ratings were obtained using a five-point rating scale from 15 listeners with normal hearing and 15 listeners with hearing loss.

The processing conditions included 32 conditions for the noise and nonlinear processing, 32 conditions of linear filtering, 36 conditions combining noise and nonlinear processing with linear filtering, and 12 unprocessed reference conditions, giving a total of 112 different conditions of which 100 represented the different forms of hearing aid processing. The noise and nonlinear conditions included stationary speech-shaped noise, multi-talker babble, peak clipping, quantization noise, dynamic-range compression (with both speech in quiet and speech in babble), spectral subtraction (with speech in babble), and combined compression plus spectral subtraction (with speech in babble). The linear conditions included high-pass filtering, low-pass filtering, band-pass filtering, spectral tilt with positive and negative slopes, a single spectral peak, multiple spectral peaks, and multiple spectral peaks combined with a low-pass filter. Six of the nonlinear conditions were combined with six linear conditions to give the 36 combined linear and nonlinear conditions. Each listener carried out quality ratings on four complete repetitions of the processing conditions: two with the stimuli from the male talker and two with the stimuli from the female talker.

This paper compares the perceived speech quality across listener groups and within and across types of signal processing (linear, nonlinear and combined linear-plus-nonlinear). Analyses of the quality judgments indicate that they are based on an underlying multidimensional perceptual space. The nature of this space will be discussed. The results provide a comprehensive data set giving the perceived quality for a variety of stimuli representative of real-world hearing-aid processing conditions. This data set will be useful in developing models and predictive metrics of quality perception for hearing aid applications and in comparing the different processing options available in contemporary hearing aids.

[Work funded in part by a research grant from GNResound].

9.55 AM OBJECTIVE QUALITY MEASURES FOR (BINAURAL) HEARING AIDS

Birger Kollmeier, T. Rohdenburg, R. Beutelmann, V. Hohmann
Medizinische Physik, Universität Oldenburg & HörTech gGmbH, Oldenburg, Germany

Since the ultimate goal of hearing-aid development is a positive (subjective) judgment of the individual hearing-impaired listener, time-consuming tests with the end user are indispensable. However, time- and effort-saving objective methods to assess the potential benefit of different versions and parameter sets of hearing aid algorithms are an attractive alternative approach if they are applied to conditions that are validated with experimental data. This contribution reviews previous approaches to predict the hearing-impaired judgement and speech reception performance for monaural hearing aids and tests the possible extensions towards algorithms employed in binaural hearing aids, such as, e.g., head-worn beamformer systems with a binaural output.

For monaural noise reduction schemes, the objective perceptual similarity measure (PSM) from PEMO-Q (Huber & Kollmeier, 2006) yields high correlations with subjective data (Rohdenburg, Hohmann, & Kollmeier, 2005).

It evaluates the similarity between a tested condition and an "ideal" reference condition not on the physical level, but rather on the perceptual level at the output of a perception model for the individual hearing-impaired listener.

For noise reduction schemes with binaural output, the binaural speech intelligibility measure BSIM (Beutelmann & Brand, 2006) appears promising: It employs a binaural pre-processing stage (based on the Equalization and Cancellation (EC)-model) followed by a speech intelligibility index (SII)- based prediction scheme. BSIM is capable of predicting the relative benefit of binaural signal presentation and signal enhancement in complex spatial signal and noise source configurations in terms of the speech-reception threshold (SRT). It can also quantify the "effective" perceptual degradation if binaural information is distorted. A combination of BSIM and PSM can be used to assess the effect of noise reduction algorithms (such as adaptive beamformers) and to optimize their respective performance for different acoustical situations.

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Eindhoven.

10.20 AM DEVELOPMENT AND ANALYSIS OF AN INTERNATIONAL SPEECH TEST SIGNAL (ISTS)

Inga Holube, Stefan Fredelake, Marcel Vlaming, Institute of Hearing Technology and Audiology at the University of Applied Sciences and Center of Competence HoerTech, Oldenburg, Germany, VU University Medical Center, Amsterdam, The Netherlands

At present measurement procedures for hearing instruments according to ANSI and IEC standards apply sinusoids or stationary noises as test signals. Due to the nonlinear signal processing in modern hearing instruments, e.g. multi-channel dynamic compression, noise reduction, and feedback cancellation, these test signals are not very adequate to reflect the effect of hearing instrument gain and output level for real life situations. For analyzing the processing of speech (as being the most relevant real-life signal) by hearing instruments, a standard test signal is necessary which allows for reproducible measurement conditions and which contains all or the most relevant properties of natural speech. The most important properties are e.g. the modulation spectrum and the fundamental frequency with harmonics. Existing artificial signals simulating speech, e.g. the ICRA5-signal or the P50 (International Telecommunication Union, ITU) fulfill these requirements inadequately whereas recordings from natural speakers represent only one language and therefore are not internationally applicable. Hence, the European Hearing Instrument Manufacturer Association (EHIMA) has set up the ISMADHA working group which has initiated a project resulting in an International Speech Test Signal (ISTS). The ISTS is reflecting the most important characteristics of natural speech, e.g. the long-term average speech spectrum, the ratio of voiced and unvoiced fragments, the modulation spectrum, and the distribution of speech pauses and speech intervals. It is based on natural recordings which were made with female speakers speaking American English, Arabic, French, German, Mandarin, and Spanish as their mother-tongue. All recordings were segmented. The segments were concatenated in random order to construct the ISTS. During this process, several statistically motivated restrictions were respected in order to result in an unintelligible test signal while preserving most relevant speech characteristics. Special attention was paid to the duration of speech pauses and speech intervals, as well as pitch shifts when switching between the speakers. This contribution will show the development of the ISTS as well as its characteristics compared to existing signals. It is planned to include the ISTS together with a new measurement method into a new ANSI and IEC standard for characterizing hearing instruments.

[This work was supported by EHIMA and AGIP/EFRE]

Poster Program

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

POSTER SESSION A

Thursday 8:00 AM –10:00 PM

A1

Observed and Predicted Benefit of Bilateral Hearing Aids

Jayne B. Ahlstrom, Amy R. Horwitz, Judy R. Dubno, Medical University of South Carolina, Department of Otolaryngology-Head and Neck Surgery

Individuals with hearing loss have difficulty understanding speech in noise, have reduced ability to detect and localize signals in space, and derive less benefit from spatial separation of speech and noise sources. Bilateral amplification should benefit speech recognition in noise by increasing speech audibility, improving directional hearing, and improving spatial benefit by restoring the availability of interaural level and timing cues. Here, observed and predicted hearing-aid benefit and the benefit attributable to spatial separation were measured as a function of low-pass cutoff frequency of speech and babble with and without bilateral hearing aids. Listeners were older adults with cochlear hearing loss fit with digital hearing aids programmed according to NAL. Predictions of hearing-aid benefit and spatial benefit for each cutoff frequency were determined by an importance-weighted speech-audibility metric (aided audibility index, AAI). Three to six months following hearing-aid fitting, speech levels corresponding to 50%-correct recognition of sentences from the Hearing in Noise Test (HINT) were measured in a multi-talker babble with two loudspeaker configurations: (1) sentences and babble at 0^{o} azimuth and (2) sentences at 0^{o} azimuth and babble at 90^{o} azimuth. Speech and babble spectra for all conditions were digitally recorded using a probe microphone placed in each ear canal of each subject. Spectra and levels of speech and babble, and unaided thresholds, were used to calculate the AAI. Finally, self-report questionnaires were administered to assess each listener's perspective on the success of their hearing aids and to determine the association between subjective and objective measures of speech recognition.

Thresholds for HINT sentences in babble improved significantly when aided and when speech and babble were spatially separated. Specifically, hearing-aid benefit improved significantly as cutoff frequency increased from 1.8 to 3.6 kHz, but only when speech and babble were spatially separated; likewise, spatial benefit improved significantly from 1.8 to 3.6 kHz, but only in the aided condition. No further improvement in hearing-aid or spatial benefit was observed when cutoff frequency was increased from 3.6 to 5.6 kHz, although improvement in hearing-aid benefit was predicted. With negligible predicted spatial benefit, unaided and aided spatial benefits were significantly greater than predicted and aided spatial benefit was greater than unaided. Few significant correlations were observed between self-report measures and objective measures of speech recognition; nevertheless, some significant correlations supported an association be-

tween objective and subjective binaural performance.
[Work supported by NIH/NIDCD]

A2

New Developments in Middle Ear Implants

Eberhard Aigner, Prof. Dr. med. Matthias Tisch, Prof. Dr. med Heinz Maier, Hörgeräte Iffland, Ulm, Germany, Military Hospital Ulm, Germany, Head and Neck Surgery, Military Hospital Ulm, Germany

Middle ear implants are an established method in the rehabilitation of sensorineural hearing loss. An invaluable addition to conventional hearing systems, they are also particularly suitable for patients intolerant of foreign bodies in the ear canal or for patients with inner ear hearing impairment with steep high frequency loss. Future applications will likely lie in a new, very interesting field, i.e. mixed hearing losses. Satisfaction with and patient acceptance of these systems are very high. In the following report the middle ear implant currently most frequently used the Vibrant Soundbridge®, is presented with clinical data.

Introduction:
Since the first implantation of a Soundbridge system by Prof. Dr. Fisch in September 1996 at the University ENT Clinic Zürich, approximately 2,000 implantations have been performed worldwide. Clinical studies in Europe and the USA have demonstrated significant improvement of hearing performance in the implanted patients. These improvements have been substantiated by objective and subjective measurements. Implantable hearing systems are particularly important in military medicine due to their special indication range. In addition, they complement and expand conventional hearing aid options and – with advanced computer technology and miniaturization – will become increasingly important over the next years in the rehabilitation of hearing-impaired individuals.

Method:
The semi-implantable hearing system consists of two components. The external part, the Audio Processor, held in place on the head under the hair with a magnet, consists of the microphone, battery and hearing aid chip. Environmental sounds are processed into electrical signals and transcutaneously transmitted to the implanted part of the Vibrant Soundbridge. The internal receiver sends the signal via a conductor link to the Floating Mass Transducer (FMT), which vibrates and thus enhances the natural vibrations of the ossicles by its direct contact to the incus (Fig. 1).

Discussion and prospects:
The Vibrant Soundbridge has particularly proven its value as a middle ear implant at the ENT department of the Military Hospital Ulm. This technology is very suitable for patients who cannot use hearing aids. Exceptional results were obtained with high frequency hearing losses, which are typical in soldiers subjected to recurrent noise traumas. To improve speech understanding in noise and improve directional hearing, bilateral implantation will be increasingly performed to treat inner ear hearing loss. The Vibrant Soundbridge will be an important addition to device-based rehabilitation of inner ear hearing loss. New surgical techniques indicate that its design (very small transducer) will make the Vibrant Soundbridge very successful in treating complicated middle ear conditions (mixed hearing losses).

A3

A Survey of the Relationship between Cognitive Ability and Speech Perception in Noise

Michael A. Akeroyd, MRC Institute of Hearing Research (Scottish Section), Glasgow, U.K.

Since the publication of the CHABA report in 1988 (J. Acoust. Soc. Am., 83, 859-895), there have been 18 published reports of experimental measures of speech recognition in noise in normal or hearing-impaired adults, some aspect of cognition (defined somewhat broadly), and then looked at the links or relationships between the two. Here a quantitative survey of these studies is presented. There was a wide variety of speech tests, cognitive tests, and statistical methodologies across the studies. Overall, it was found that (1) there is a link between cognition and speech reception, but it is weak, and secondary to hearing loss as a predictor; (2) no speech test (or type of masking noise) always gives a significant link; (3) no cognitive test always gives a significant link, although (4a) measures of working memory (especially "reading span", but also "visual letter-monitoring") are mostly effective, (4b) as are measures of visual speech identification using partially-masked written sentences, (4c) but measures of general ability, such as IQ, are mostly ineffective. Six of the studies included aided listening and two reported the benefits from aided listening: again mixed results were found, and in some circumstances cognition was a useful predictor of hearing-aid benefit. [This work was supported by MRC (U.K.) and CSO (Scotland)].

A4

The Role of Nonlinear Cochlear Processing in Human Speech Perception

Jont Allen, Marion Regnier, Sandeep Phatak, University of Illinois, ECE, Walter Reed Hospital, Washington DC

Little is know about how the auditory system decodes speech. We may think of speech communication re Shannon's source−channel model, thus viewed, the most complex part of the speech communication channel is the auditory system (the receiver). In our speech−perception research, we have fallen back on Shannon's basic source−channel model. The basic tool is the confusion matrix (CM) for isolated natural consonant and vowels (CV), as a function of the speech to noise ratio (SNR), with several types of masking noise. We have used large numbers of talkers and listeners (i.e., 20). We selectively remove islands of speech in time−frequency, and then correlate the resulting modified speech against subject scores. We will show that speech perception is very nonlinear. Possible reasons for this are forward masking and the upward spread of masking. Live demos will be played, including ''edge−enhanced'' speech signals, having a greater robustness to noise. Our most important conclusions are:

1) The across−frequency onset transient portion of the signal is typically the most important.

2) The spectral regions of these transient are used to code different consonants.

3) Compact spectral−temporal amplitude modulations components (e.g., a 10 Hz modulation) do not seem to play a significant role, at least above 1−2 kHz.

A5

The Effect of Formant Trajectories and Phoneme Durations on Vowel Perception

Akiko Amano-Kusumoto and John-Paul Hosom, Center for Spoken Language Understanding (CSLU) at Oregon & Health Science University, USA

Picheny et al. (1985) found that the intelligibility of clear (CLR) speech, which is spoken deliberately clearly as if talking to a hard-of-hearing listener, is higher than that of conversational (CNV) speech, which is spoken as if talking with a colleague. A number of acoustic features have been recognized to be different between CLR and CNV speech (Picheny et al. 1986, Krause and Braida 2004). In this work, we focus on phoneme duration and the vowel space (a two-dimensional representation of F1 and F2). According to the previous research, phoneme durations of CLR speech are longer, especially for the tense vowels, and the vowel space of CLR speech is larger for the lax vowels, compared with CNV speech. In our previous work, when only phoneme duration was manipulated, a significant improvement over the intelligibility of CNV speech was not observed. However, when phoneme duration and short-term spectrum were manipulated by hybridizing those two features from CLR speech with the remaining features from CNV speech, a significant improvement was observed (Kusumoto et al. 2007). We hypothesized that lengthening or shortening phoneme durations without appropriately modifying formant trajectories has a negative impact on vowel identity, because formant dynamics are altered. This hypothesis motivates our current study, making modifications to formant targets and/or formant transitions to improve speech intelligibility. The application of this research to improve speech intelligibility has the potential to be developed for hearing aids or assistive listening devices. One male talker elicited the words wheel, will, well, wail, and tool for test words, and heed, hid, head, hayed, and who for reference words, all in a carrier sentence. CNV speech was recorded first, followed by CLR speech. The speaking rate was self regulated and was subsequently measured at 363 wpm for CLR speech and 163 wpm for CNV speech. We created four types of hybrid (HYB) speech: (1) formant trajectories from CLR speech applied to CNV speech, with phoneme duration adjusted linearly, (2) formant slope and targets from CLR speech applied to CNV speech, (3) formant slopes from CLR speech applied to CNV speech, and (4) formant slopes and targets from CNV speech applied to CLR speech. We will present the method to modify formant trajectories and phoneme durations, the results from perceptual experiments using these four types of HYB speech, and the results of acoustic analysis.

A6

Digital Signal Processing Algorithm Arrangement in Hearing Aids: Parallel Versus Series

Melinda C. Anderson, University of Colorado at Boulder, Kathryn H. Arehart, University of Colorado at Boulder, James M. Kates, GN Resound, University of Colorado at Boulder

Little is known about the perceptual consequences of interactions when signal processing algorithms are combined within hearing aids. For example, when dynamic range compression and spectral subtraction are used to process noisy speech, the compression increases the intensity of low level sounds, while spectral subtraction works to decrease the intensity of many of those same sounds. It is possible for these two signal processing algorithms to work together, but it is likely that they will at times work against each other. Of particular interest are the differences between parallel and series configurations of dynamic range compression and spectral subtraction when processing noisy speech. In parallel processing the signal is modified by each algorithm independently and the gains added together. In series processing a noisy signal is first processed by the spectral subtraction routine and the modified signal is then sent to the dynamic range compression routine, where further gain adjustments take place. Parallel processing shows greater suppression of noise regions than series processing, due to the fact that some of what is done by the spectral subtraction routine is undone by the subsequent compression routine. The perceptual differences in these configurations were investigated by examining speech understanding and speech quality in listeners with normal hearing and with hearing loss. Speech quality has been measured using HINT sentences in a paired-comparison task for parallel and series processing at a range of signal-to-noise ratios (SNRs) from +6 to 0 dB. Speech intelligibility has been measured using sentences from the IEEE corpus at SNRs ranging from +6 to -6 dB. Linear processing was included as a reference condition. To date, thirty listeners have participated, 15 with normal hearing and 15 with hearing loss. Results show significant effects of processing. Linear processing shows the best speech intelligibility scores, while series outperforms parallel processing in intelligibility. Complete statistical analysis of the perceptual data will be presented. Acoustic analysis using spectrograms and cochlear modeling of inner hair cell firing rates will be used to investigate relationships with the perceptual findings. Results will be discussed in terms of optimal processing configurations, as well as assessment of trade-offs in combining signal processing algorithms in different ways.

A7

Age-Related Deficits in F0 Processing: Use of Periodicity and Fine-Structure Cues

Kathryn H. Arehart, University of Colorado at Boulder Pamela E. Souza University of Washington, Seattle, Christi Wise Miller, University of Washington, Seattle, Ramesh Kumar Muralimanohar, University of Colorado at Boulder

Recent studies suggest that older listeners have difficulty processing information related to the fundamental frequency (F0) of voiced speech. The purpose of this study is to explore the mechanisms that underlie this reduced ability. Specifically, we examine the extent to which degradations in F0 processing are due to a decreased ability to use a) fine structure cues provided by the harmonic structure of voiced speech sounds and/or b) high-rate envelope fluctuations (periodicity) cues. F0 processing is considered for stimuli processed in three ways on four different F0 tasks. The first processing condition, which provides a baseline for F0 processing abilities, is unprocessed speech. The second processing condition, which is designed to remove fine structure cues and leave high-rate envelope (periodicity cues), is speech subjected to envelope vocoding (8-channel noise vocoding with a 300 Hz cutoff frequency within each channel). The third processing condition, which provides low-frequency fine structure cues and higher-frequency periodicity cues, is a "hybrid" condition in which speech is unprocessed below 656 Hz and envelope vocoded above 656 Hz. The experimental tasks are designed to assess F0 processing with both steady-state and time-varying stimuli presented with and without competing speech. The tasks include 1) F0 discrimination using steady state vowels 2) intonation perception using synthetic diphthong glides 3) concurrent-vowel identification and 4) competing sentence perception. We are currently using these stimuli and tasks to measure F0 processing in a group of young normal-hearing listeners and a group of older adults with normal or near-normal auditory thresholds. Stimuli are presented at

70 dB SPL for listeners with normal thresholds through 6000 Hz. For older listeners with slightly elevated high-frequency thresholds, the 70 dB SPL input signals are custom amplified using the NAL-R prescriptive formula. Results to date demonstrate that a) on average, the older listeners have more difficulty on the F0 processing tasks b) the patterns of performance across processing conditions differ between the two groups and c) in contrast to the younger listeners, there is substantial variability among the older listeners. The results have implications for hearing aid design, including combined acoustic and electric hearing systems for older adults. [Work supported by grants from the University of Colorado, the University of Washington's Virginia Merrill Bloedel Hearing Research Center, and NIHNIDCD.]

A8

Development and Evaluation of a Paediatric Audio-Visual Speech Test in Noise

Laure Arnold, David Canning, Patrick Boyle
Advanced Bionics Europe, Rixheim, France
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The importance of audio-visual speech integration in the communication of children with cochlear implants may sometimes be underestimated. There are not many tools available to assess lip-reading abilities. The objective was to develop a paediatric audio-visual speech test to evaluate the benefit obtained from adding lip-reading information to auditory signal.

The materials from the existing McCormick and English as a Second Language (E2L) toy tests were selected. The 26 words available in total (14 in one test, 12 in the other) were recorded from male and female children, with and without a carrier phrase. Video and audio editing was performed through VirtualDubMod and Cooledit 2000 respectively, to add competing noise to the speech signal. A display adaptor capable of driving two screens simultaneously ("subject" screen allowing lip-reading and "clinician" control screen) was created and software prepared to drive it. Pilot testing was performed on twelve paediatric cochlear implant users who were tested on the E2L words spoken by the male child, first with lip-reading, then with auditory signal only.

The pilot testing showed that the video recordings provided acceptable quality. The preliminary version of the software platform is functional. Words are randomly presented to the subjects whose task is to indicate the corresponding picture or toy, or to repeat the word. Testing may be conducted in quiet or noise, with an adaptive signal to noise ratio. The clinician can select lip-reading only, or lip-reading with audio or audio only. Controls are available for scoring and automatic report generation indicating the signal to noise benefit of lip-reading. Feedback was collected on how to improve the ergonomics of the interface. The preliminary data showed differences in scores obtained in the audio-visual condition compared to the auditory only condition, highlighting the benefits that might be obtained from adding lip-reading information to auditory signal.

The feasibility of using recorded audio-visual material to assess lip-reading abilities was confirmed. The option to test in noise allows for a better representation of real-life conditions. This aspect will be further developed through the implementation of a roving option. In addition to the possibility of assessing speech understanding in difficult listening situations, this tool may also have the potential to assist in determining rehabilitation options for implanted children whose lip-reading ability is important in the choice of educational settings. The next steps will involve finalizing the interface and testing a larger group of children.

A9

The Demographics of Hearing Aid Users in the United States

Justin M. Aronoff & Sigfrid D. Soli, House Ear Institute, Los Angeles, CA

Although the validity and generalizability of hearing aid research depends on having a representative sample, it is difficult to determine whether a representative sample has been obtained in a given study because little is known about the demographics of hearing aid users. This retrospective study aimed to address this problem by determining the detailed demographics of the population of hearing aid users in the United States.

Data were acquired from two databases: One provided by a major hearing aid manufacturer, and the second provided by the House Clinic, a tertiary care otological clinic located in Los Angeles. Together, these two databases contained data on 134,019 hearing aid ears for hearing aids acquired in 2006 or 2007. The manufacturer's database contained data for ITE, ITC, and CIC hearing aid users, whereas the House Clinic database contained data for BTE and OEC hearing aid users, as well as limited data for ITE and CIC hearing aid users. The inclusion of ITE and CIC data in both databases allowed a direct comparison, which suggested that there were small differences between the two populations. The House Clinic clients had slightly less hearing loss (approximately 5 dB HL less) and were slightly younger (approximately 5-10 years younger) than the manufacturer's clients.

The age distribution of hearing aid ears was skewed towards older individuals and had a median age that ranged from 70 to 81 years old across hearing aid types. Hearing loss had a roughly normal distribution within each hearing aid type, with a median hearing loss of 42-62 dB HL across hearing aid types. Hearing aid type was not independent from hearing loss, with OEC users having the mildest hearing losses and ITE users having the severest, although there was considerable overlap. The vast majority of hearing aid users (66-85% across hearing aid types) had disproportionate high frequency hearing loss, defined as more than a 10 dB HL increase in hearing loss for the 2-4 kHz frequencies compared to the .5-1 kHz frequencies. For all hearing aid types, starting at approximately forty-five years of age, hearing loss severity increased and hearing loss variability decreased as a function of age. For younger individuals, hearing loss was very idiosyncratic. These results provide guidance for researchers in selecting hearing aid patients and in interpreting results for studies of hearing aid users.

This research was supported by the Mobile Manufacturers' Forum and the GSM Association.

A10

Frequency Transposition on a Linear or Warped Frequency Scale: Potential Benefits for Listeners with High-Frequency Dead Regions

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

Hearing-impaired listeners with high-frequency dead regions (DRs) benefit from amplification of frequencies up to about 1.7 times the edge frequency (f_e) of the DR, indicating a contribution of off-frequency listening. Previous studies simulated on- and off-frequency listening by DR listeners in normal-hearing (NH) listeners by lowpass filtering speech either at f_e of the simulated DR or at 1. f_e. Intelligibility improved when the f_e - 1. f_e band (reference band) was added to the band up to f_e (base). However, better intelligibility was obtained when the base was combined with a band of identical width to that of the reference band, but centered at higher frequencies. Thus, intelligibility in DR listeners may be improved if information contained in the higher band could be transposed to replace that in the reference band.

The present study was designed to explore this potential by simulating a DR with f_e =0.75 kHz in six NH listeners. In the first experiment, consonant intelligibility for vowel-consonant-vowel utterances was measured for the base alone, the base plus the reference band, and the base plus a band centered at one of four different higher frequencies. The bandwidths of the added bands were identical to that of the reference band either in ERB_N number (ERB_N condition) or in Hertz (linear condition). Highest scores were obtained for the added band centered near 4 kHz, and intelligibility was always higher for the ERB_N condition at identical center frequencies.

In the second experiment, consonant intelligibility was remeasured in six naïve NH listeners for the base alone, the base plus reference band, and the base plus the best added band from the ERB_N and linear conditions in experiment 1. In addition, intelligibility was assessed when the latter added bands were transposed to the reference band. Transposition was performed using (i) an FFT-based technique for the linear condition, and (ii) frequency warping, followed by FFT-based transposition and unwarping for the ERB_N condition. Results were similar to experiment 1 for the conditions without transposition. With transposition, intelligibility was lower than that with the reference band but similar to that with the base alone. Thus, transposition without training does not seem to provide any benefit. Further studies are warranted to assess the role of training.

[Supported by MRC (UK) and Marie-Curie Fellowship.]

A11

Relating Patient Complaints To Hearing Aid Behavior

Shilpi Banerjee, Katherine Teece, Eric McCabe, Starkey Laboratories Inc, Starkey Labs, University of Minnesota,

In audiological practice, clinicians are routinely called upon to make fine-tuning adjustments to hearing aids based on patients' qualitative reports. Through a survey of 311 audiologists and 24 experts, Jenstad et al. (2003) showed that patient complaints can be related to basic hearing aid parameters – compression, gain, output, and frequency response. The outcome of this survey formed the basis of an expert system for troubleshooting hearing aid fittings. But, hearing aids are evolving rapidly. Over a period of 10 years, digital signal processing (DSP) has gone from virtual non-existence to accounting for more than 90% of hearing aid fittings (Kochkin, 2002, 2005). In contrast, satisfaction with hearing aids has remained relatively constant at 60-70% over the same period of time. This is in spite of the fact that clinicians believe that their patients are more satisfied with DSP in comparison with older technology. One explanation for the disparity is that the nature of the problems has changed. Indeed, the top 10 correlates of hearing aid satisfaction are more subtle today (e.g., richness or fidelity of sound) than in 2000 (e.g., use in noisy situations).

Numerous studies have investigated preference for and/or benefit from DSP features in hearing aids. However, to the author's knowledge, there are no studies that specifically scrutinize the relationship between complaints and the behavior of DSP features in hearing aids. The objective of the present study was to examine the relationship between patient complaints and the activation of DSP features. Ten adults with bilateral, mild-to-moderate sensorineural hearing loss participated in the study. They were fitted bilaterally with Starkey Destiny 1200 BTEs – equipped with expansion, automatic directionality and noise management – and asked to appraise hearing aid performance in everyday environments.

Although participants chose when and where to appraise performance, they were specifically instructed to do so while in the environment under evaluation. The hearing aids were connected to a PDA, which logged objective data from the hearing aids as well as subjective responses to survey questions. Over the course of 3-4 weeks and 184 evaluations, analysis of complaints showed differences in feature activation. These data offer a glimpse into real-world experiences. The outcomes and their clinical relevance will be discussed.

A12

Psychophysical Approach to Investigating Relative Loudness of Self-Generated Speech

Dragana Barac-Cikoja, Jose Reyes III and Sarah Sonnemann, Gallaudet University, Washington, DC, USA

Measurements of the perceived loudness of speech feedback have been obtained using a psychophysical approach that involves direct comparisons between hearing one’s own speech as feedback, during speech production, and as a recording, in a listening-only condition. With the air-borne component of the speech signal recorded and under experimental control, it is possible to implement adaptive stepwise changes in its intensity. The participant is asked to indicate which of the two signals, the speech feedback or the recording, sounded louder. Based on his/her responses, the participant’s intensity difference threshold, IDT is obtained. IDT is defined here as the difference in sound pressure level (dB SPL) of each signal (feedback and replay), when the two are experienced as equally loud. Several studies investigating the perceived loudness of one’s own speech feedback have been conducted using this procedure. The experiments on normally hearing individuals revealed significant individual differences in IDT values. When both live and recorded speech was presented through insert earphones average IDT values varied across the participants from 0 dB to 4 dB. In other words, for some individuals, at the point of subjective equality, the SPL of the recorded speech signal exceeded the SPL of the live (feedback) signal by as much as 4 dB. The role of a voice activated acoustic reflex in the observed reduction in the feedback loudness, and the related effect of occlusion will be discussed. Significance for hearing aid (HA) users will be examined. We can expect level differences between self- and other-generated speech to vary depending on the residual hearing of an individual, the amount of occlusion created by an earmold, and the amount of speech feedback amplification provided by a HA fitting. Empirical data on how these factors influence self-hearing could provide an objective basis for considering HA fittings specifically designed for self-hearing. They could also facilitate decisions on the canal length and the amount of venting required to enhance hearing aid benefits during speech production.

Support provided by the Rehabilitation Engineering Research Center (RERC) on Hearing Enhancement and the Gallaudet Research Institute.

A13

Phonemic Restoration with Hearing-Impaired Listeners of Mild to Moderate Hearing Loss

Deniz Başkent, Cheryl Eiler, and Brent Edwards, Starkey Hearing Research Center, Berkeley, CA

In phonemic restoration with normal-hearing listeners, when silent intervals of interrupted speech are filled with loud noise bursts, speech is perceived as continuous. This ability to perceive sounds with moments of inaudibility as continuous likely plays a role in cocktail party-like conditions, where target signals are made temporarily inaudible by masking from nearby interfering sounds, yet the target speech is perceived as continuous. Under specific conditions, benefit from phonemic restoration is not only perceived continuity but also an increase in speech intelligibility.

Başkent et al. (ISAAR 2007) showed that undershoot that may happen due to release times common in hearing-aid compression may reduce phonemic restoration benefit. The undershoot effect was simulated and the subjects were normal hearing (NH). These results would be more crucial for hearing-impaired (HI) listeners who might encounter such problems in real life as hearing-aid users. However, it has not been previously shown if HI listeners benefit from phonemic restoration similar to NH listeners. Factors such as adverse effects of background noise on speech perception and increased forward masking observed with HI listeners might affect the potential benefit from phonemic restoration.

The present study explored phonemic restoration with HI listeners who had mild to moderate levels of hearing loss of sensorineural origin, and NH listeners participated as the control group. A method similar to the previous study was used where speech perception was measured with interrupted sentences and with interrupted sentences combined with noise bursts. The difference between the two measures showed the benefit due to phonemic restoration. IEEE sentences, interrupted at the rate of 1.5 Hz, with duty cycles of 50% and 67%, and at the rate of 2.2 Hz, with 50% duty cycle, were used as stimuli. A half-gain rule was used for amplification and with an additional volume control program comfortable listening levels were ensured for the HI listeners.

The preliminary results showed that many HI listeners, especially with mild hearing loss, were able to benefit from phonemic restoration, similar to NH listeners. Hence, the results from the simulations of the previous study would likely apply to hearing aid users. There was substantial variability in the results and in all three subject groups (NH, mild HI, and moderate HI) a proportion of the subjects did not show the phonemic restoration effect. However, in general, less benefit was observed as the degree of the hearing loss increased.

A14

Effects of Frequency Translation on Speech Perception By Listeners With Severe High-Frequency Hearing Loss

Deniz Başkent, Kelly Fitz, Brent Edwards, Nazanin Nooraei, Matt Burk, Karrie Recker, Starkey Laboratories, Berkeley, CA, Eden Prairie, MN

Amplification through a hearing aid has been helpful for hearing-impaired listeners, especially for mild to moderate hearing impairment. For listeners with steeply sloping hearing loss, however, fitting a hearing aid can be problematic. The high frequencies would have to be amplified substantially to make high-frequency sounds, such as consonants audible. However, these listeners would have a higher chance to hear potential distortions, such as off-place listening, or feedback due to relatively better hearing thresholds at lower frequencies, which would limit the gain that can be applied at high frequencies. Frequency translation (FT), where the high frequency information is moved to lower frequencies, has been suggested for these listeners as an alternative.

The present study shows speech perception results with a new FT algorithm, with listeners who have severe high frequency hearing loss. FT is applied only to high frequency components of speech with a warping function that linearly transforms high frequencies to lower frequencies. The strength of the algorithm is controlled with a number of parameters so that it can be customized for individual listeners. The results will be presented for two experiments where performance was measured with stimuli that were only amplified and with a number of FT settings where the stimuli were both amplified and FT was applied. In the first experiment, detection /s/ was measured by using words that were presented in the singular or plural form. The subject's task was to identify if the presented stimulus was singular or plural. The preliminary data from this experiment showed an improvement for most subjects. In the second experiment, consonant confusions were measured for the same conditions. The preliminary data showed that listeners had difficulty identifying high-frequency consonants before the FT was applied, as expected from the audiograms. When FT was applied, detection for some consonants, such as plosives and affricates, improved. However, there were also a number of new confusions. For example, /s/ was made more detectable by lowering its frequency, but now it was occasionally confused with /□/ which has lower frequency content compared to /s/. The present study presents acute effects and these new confusions are likely to be reduced over time when the listener has the opportunity to listen with the FT algorithm through extended periods and adapt to the remapped acoustic speech cues. Overall, preliminary results showed a potential benefit of the FT algorithm for the listeners with severe high frequency hearing loss.

A15

Spatial Hearing Abilities of Bilaterally Fitted Hearing Aid Users Assessed Using Objective and Subjective Outcome Measures

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Spatial hearing in aided hearing-impaired persons remains a rather sparsely explored topic. Therefore we do not know enough about the ability of hearing aid users to exploit auditory cues for spatial hearing. In an attempt to advance our knowledge the following experiment was set up. A group of 21 experienced hearing aid users took part in a test requiring them to wear modern completely in the canal hearing aids in the field for at least 3 weeks. After acclimatization to the devices they were tested on a number of outcome measures. These included spatial unmasking, an interview administered selection of questions from the Speech Spatial and Qualities of hearing scale, and a baseline speech in noise measure. Spatial unmasking was assessed using three concurrent female talkers, with the target talker always presented directly ahead and the maskers presented either at +/- 50 degrees or both at 180 degrees.

This presentation will discuss aspects of approaches for testing spatial hearing in hearing impaired subjects and provide results from the study described above along with possible relations to auditory and non-auditory predictors of spatial hearing performance. These results indicate that both age and degree of hearing loss play a significant role in spatial hearing performance as assessed by the test of spatial unmasking. Finally the results will be used to discuss the need for testing hearing aid performance in environments with spatial complexity.

A16

Hearing Loss and Divided Listening

Virginia Best, Frederick Gallun, Christine Mason, Gerald Kidd, and Barbara G Shinn-Cunningham

In crowded listening environments, selective attention enables information to be extracted from a talker of interest. However, in many cases it is desirable to retrieve information from a talker who is outside the immediate focus of attention (e.g. when two people talk at once). Broadbent (Q J Exp Psychol 9:1-11, 1957) postulated that auditory immediate memory allows listeners to process simultaneous inputs in a serial fashion. In his model, sensory inputs are stored temporarily in a sensory trace, and selective attention allows an object to be processed further. For simultaneous inputs, it is possible to process one input and then use the sensory trace (if it is still available) to process the other. Given that the sensory trace is a relatively volatile form of storage, it was hypothesized that processing of a message via the sensory trace would be more sensitive to the quality of the acoustic input than processing of an attended message. If correct, this conclusion has some important implications. For example, because sensorineural hearing loss degrades the spectrotemporal representation of an auditory stimulus, hearing impairment may severely disrupt divided listening even more than selective listening.

In this experiment, listeners were asked to respond to two spoken messages presented simultaneously (one to each ear). The two messages were equal in level but were systematically degraded by adding noise. In selective listening trials, listeners reported two keywords from one message. In divided listening trials, there was an additional (secondary) task of reporting two keywords from the other message. For all listeners, errors in selective listening trials were more frequent as the noise level increased. In divided listening trials, results for the primary task were similar to those from the selective task, whereas performance in the secondary task was poorer and more affected by the addition of the noise. This finding is consistent with the involvement of a volatile sensory trace in divided listening.

Listeners with sensorineural hearing loss showed a deficit compared to their normal-hearing counterparts on selective listening trials, but showed an even greater deficit in divided listening trials due to poor performance on the secondary task. Their pattern of performance was similar to that of the normal-hearing group listening at a poorer signal-to-noise ratio. Thus, the difficulties experienced by hearing impaired listeners in situations involving divided listening (such as dynamic conversations involving several people) may be partly explained by the poor quality of their sensory trace.

A17

Vowel Identification, Hearing Loss Configuration and Hearing Aid Processing

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Hearing aid (HA) users with sloping versus flat sensorineural loss may perform differently in the same vowel identification task. Listeners with sloping loss are predicted to have more high frequency formant (F2) vowel confusions corresponding to broad high frequency auditory filters, while listeners with flat loss are predicted to have both low and high frequency formant (F1 & F2) confusions corresponding to broad auditory filters across frequencies. The interaction between loss configuration and hearing aid processing is examined in this study using auditory filter measures and vowel identification. It is hypothesized that vowel errors in both groups will increase in the same frequency dependent manner as vowel stimuli become degraded by amplitude compression. First, auditory filter bandwidths were measured at 500 and 2000 Hz using a simultaneous masking notched-noise paradigm (Stone et al., 1992, Br J Audiol, 6:329-334). Equivalent rectangular bandwidths (ERB) were calculated from the notched-noise data using a rounded exponential (roex) filter shape model (Patterson et al., 1982, J Acoust Soc Am, 72:1788-1803) implemented via the POLYFIT procedure (Baker & Rosen, 2006, J Acoust Soc Am, 119:454-462). Second, vowel identification was tested using eight naturally-produced vowels digitally processed through sixteen HA compression channels. These vowels have been previously analyzed and have various amounts of spectral flattening (Bor et al., in press, J Speech Lang Hear Res). An unprocessed condition served as a control. All vowel stimuli were individually frequency-shaped to provide sufficient audibility. Preliminary results in each hearing loss category (sloping, flat & normal control) suggested a systematic vowel error pattern which corresponds with auditory filter measurements. Sloping loss subjects have a sharply tuned filter at 500 Hz, whereas flat loss subjects have a more broadly tuned filter at 500 Hz. Auditory filters at 2000 Hz for both groups have a high ERB value, suggesting the presence of broadly tuned filters. Results of the vowel identification test indicated that 1) listeners with hearing loss scored more poorly compared to normal, 2) performance was slightly better with sloping than with flat loss, 3) the number of vowel errors were positively correlated with increasing frequency and filter bandwidth at F2, and 4) all error patterns were exaggerated after multichannel compression processing. The results of this study provide empirical evidence for the relation between vowel errors and cochlear frequency selectivity, and stress the importance of considering individual hearing and processing characteristics. [Work supported by NIH RO1 DC006014 & DC00626.]

A18

Interactive Fitting Using Audiovisual Simulations of Real World

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There are different approaches for fitting hearing aids: prescriptive approaches (with insertion gain measurements) and interactive approaches (for example with video fragments). We compared both approaches in an extensive multi-center evaluation study in 80 subjects, using a cross-over design. For the interactive approach we applied the Amplifit II system that inventories the subject's difficulties in auditory communication using an individually-tailored selection of video fragments designed to simulate real world conditions. The subject judges speech intelligibility, listening comfort, and sound quality. After hearing aid fitting, the same simulations can be judged in the aided condition and the subjective benefits of using hearing aid(s) can be assessed. A well-structured feedback of the responses is given in six dimensions, which can be used to choose between different hearing aids and/or between different settings in the same hearing aid.In this study the hearing aid dispenser used the Amplifit II system to fit and fine tune the hearing aids. The results were compared to a second fitting made at Audiological Centers, based on Insertion Gain measurements matching the NAL-NL1 prescription rule. The subjects were asked to use one fitting for six weeks followed by the other fitting for another six weeks in a cross-over design. The order of fittings was randomized. After each trial-period the settings were evaluated objectively by insertion gain measures. The performance was evaluated by speech tests in quiet, continuous noise, and time reversed speech, both presented at 0 degrees and with spaciously separated sound sources. The subjective results were evaluated using extensive questionnaires (SSQ and AVETA). Preliminary results show that the prescriptive method gives the better speech intelligibility and can be regarded as an intelligibility driven approach. However, in the domain of subjective results some benefits were found for the interactive fitting. The interactive fitting can be regarded as a comfort driven approach that complements the prescriptive procedure.

A19

Preferred Signal Path Delay and High-Pass Filtering In Open Fittings

Lars Bramsløw, Oticon A/S

All digital hearing aids (HA) introduce a processing delay in the amplified signal path. The combination of delayed sound from the hearing aid with direct sound through an open or vented fitting can lead to measurable comb filter effects if the direct and amplified contributions have similar intensities. These interactions can potentially degrade the sound quality due to audible changes in timbre and/or perception of echo.

Whereas increased signal path delay may degrade sound quality, it provides opportunities for increased flexibility in the hearing and more complex signal processing. One possible way to reduce the side effects of increased delay is to avoid equal contributions from HA and vent by either decreasing or increasing HA gain in the critical frequency regions, e.g. by adjusting a high-pass cutoff for the HA gain.

The present study was designed to test a number of delay and high-pass combinations under worst-case (i.e. most sensitive) conditions. 16 normal-hearing and 16 mildly hearing-impaired subjects performed the test in a paired comparison (A/B) task. The subjects were asked to select preferred setting with respect to sound quality. The test was set in an anechoic chamber using speech, own voice and environmental sounds. Experimental hearing aids were used and these were fitted with open domes thus providing maximum ventilation.

The preference data have been processed using a statistical model that derives a ratio-scale (Ellermeier et al, 2004). A preliminary analysis of group data indicates that there is no strong preference for delay but that high-pass filtering is preferred for normal-hearing listeners due to the absence of comb filtering artifacts.

Ellermeier, W.; Mader, M.; Daniel, P.: Scaling the unpleasantness of sounds according to the BTL model: Ratio-scale representation and psychoacoustical analysis. Acta acustica, Vol. 90 (2004), no.1, 101-107

A20

Validation of Objective Sound Quality Models for Hearing Aids

Lars Bramsløw and Marcus Holmberg, Oticon A/S

Objective estimation of sound quality has been standardized and successfully applied in the field of telecommunication. More recently, three sound quality models have been developed for use with hearing aids. This was a joint project by Hörtech (Oldenburg) and a number of hearing aid manufacturers. The models include hearing impairment and can be used to assess the sound quality of a hearing aid. As part of the development, the models have already been validated on existing hearing aids.

In order to have faith in such models it is crucial to validate them on a large range of test conditions, as close to the future (unknown) applications as possible. In the present study, the sound quality models were applied to existing in-house subjective test data for a diverse set of different signal-processing algorithms and acoustic conditions. These include noise reduction, frequency manipulation, compression, bandwidth limitation, clipping, and both open and closed fittings in a number of combinations. Data from both normal-hearing and hearing-impaired listeners were available.

This is a typical future application of the models, evaluating types of algorithms or test conditions that were not used in the development of the models. The results of the validation were mixed, with some predictions highly correlated with subjective ratings and others poorly correlated. On average, the models seemed to perform quite equal, although specific models performed better on specific data sets. The new validation data are presented and discussed, and potential limitations of the models are highlighted.

A21

Speech Intelligibility in Normal-Hearing and Hearing-Impaired Listeners: The Roles of Pitch, Spatial Separation, and Reverberation

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Understanding speech in noisy and reverberant environments is a task which normal listeners can perform quite well and with which the hearing impaired have great difficulty. By comparing performance of normal listeners to hearing impaired listeners in carefully designed complex environments, the auditory deficiencies of the hearing impaired can be studied. The ultimate goal is being able to improve a hearing-impaired person's performance in real-life listening situations. This may involve the use of an aid that, through signal processing, can attempt to restore performance to that of a normal-hearing listener. In an effort to this end, it is important to find and investigate the perceptual effects of cues that may or may not be useful in segregating speech in complex environments, such as pitch and spatial separation.

The ability to use speech segregation cues, pitch and spatial separation, to understand a target sentence amidst masking sentences is tested in a simulated reverberant environment. Speech reception thresholds, defined here as the signal-to-noise ratio needed to perform at 50% of target words correct, are measured for a variety of conditions, for both normal-hearing and hearing-impaired listeners. A main parameter varied is the amount of simulated reverberation: anechoic, moderate reverberation (T60=.35s), high reverberation (T60=.7s). Another is the spatial separation of the sources: the target is always in front of the listener, and the masker sentences are either both co-located at zero degrees as well, symmetrically placed at +/- 60 degrees from the center, or co-located at +60 degrees. The pitch contours of the sentences are flattened using PRAAT software, and the target is separated by 0, 1 or 4 semitones from the masking sentences. Preliminary results show that normal-hearing listeners get a benefit from having pitch separation, and that the normal-hearing listeners get a benefit from spatial separation. The preliminary results indicate that impaired listeners get less benefit from spatial separation than normal listeners, and that the benefit from having pitch separation varies between individual impaired listeners. [This work was supported by NIH/NIDCD grant R01 DC00100].

A22

Technique to Characterize the Differential Treatment of Speech and Noise in Adaptive Signal Processing Devices: Using Signal Inversion to Extract Estimates of the Modified Speech and Noise Components

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The primary goal of adaptive signal processing algorithms, in a hearing instrument, is to treat the speech component differently than the noise component in a combined speech plus noise signal. Generally, the intention is to either improve the speech understanding or to reduce the annoyance of noise. Different adaptive algorithms provide different advantages when confronted with a speech plus noise signal. For example, directional microphone techniques reduce competing sounds (i.e., noise) that are spatially separated from the front target source (i.e., speech). Noise cancellation algorithms reduce the level of ambient noise (independent of spatial location), while attempting not to attenuate the target speech signal. Speech enhancement algorithms place emphasis on only the speech component of a combined speech plus noise signal.

It is difficult to assess, optimize, or verify the performance of adaptive signal processing algorithms, because the algorithm must be tested with the signal it is intended to improve, namely a combined speech plus noise signal. Since the speech and noise are mixed in the test signal it is difficult to determine if the signal processing block treat the speech component differently from the noise component. Signal inversion (Hagerman et.al., 2002, Souza et.al., 2006) allows the tester to separate out the speech component and the noise component in a combined speech plus noise signal. For acoustic measures on hearing aids the only requirement, for signal inversion to work, is that the hearing aid behaviour over the time course of the test signal is the same for the original and inverted version of the test signal. This is generally true; one possible exception may be feedback cancellation algorithms. The data presented is obtained on acoustic measures of commercially available hearing instruments, using a semi-pro off-the-shelf external firewire sound card.

With separate estimates of the speech and noise components it is possible to characterize the differential effect that the signal processing block had. One technique is to measure the post-processing SNR. However, a simple SNR measure does not reveal whether the signal processing affected the speech component, the noise component, or both. An alternative approach is to measure the differential impact on the extracted speech component and the extracted noise component. The extracted speech and noise components can be used to generate long-term average 1/3-octave band data (taken after the processing algorithm has stabilized), which are then used to measure a long-term average change. The presentation will briefly review signal inversion and then describe the analysis techniques.

A23

Effects of Age and Background Noise on Human Sound Localization

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Sound localization accuracy was assessed for three groups of normal hearing listeners. Participants included 12 children between 7 and 12 years of age, 9 children between 13 and 18 years of age, and 10 adults between the ages of 22 and 30 years. Subjects were asked to localize sound stimuli in two conditions; 1) a 300ms burst of white noise in a quiet hemi-anechoic chamber, and 2) the perceived location of a car horn amidst a stereo recording of traffic noise presented at ±90º within the chamber. Target stimuli were presented from one of nine locations 22.5º apart, spanning 180º in the frontal-horizontal plane. Subject responses were collected with a head-mounted electromagnetic tracking unit which monitored the position of the subjects' head in space. Localization performance was assessed by comparing the average root-mean-square (RMS) error between groups and test conditions. Results indicated that subjects made significantly more localization errors in the presence of background noise than in a quiet environment. Additionally, the RMS error of the youngest group of children was significantly higher than that of the adult subjects. Preliminary data of hearing impaired subjects with and without the use of hearing instruments is also presented.

A24

How Auditory Feedback Can Change Speech Production: Implications For Hearing Aids

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Previous research has shown that talkers will spontaneously compensate for perturbations to the auditory feedback of their voice such as compensating for pitch- and formant-shift manipulations. Thus, the modified auditory feedback experienced by users wearing hearing aids may have unintended consequences on their speech production. Two experiments that demonstrate the role of auditory feedback in controlling speech production are presented. The first explores the effect of wearing headphones while the second explores the effect of formant shifting.

In the first experiment, 17 undergraduate females with clinically normal audiograms were prompted to say the word "head" into a microphone a total of 90 times with their normal speaking voice. The talkers received normal auditory feedback (i.e., no headphones) during the first 30 utterances. For the next 30 utterances, talkers wore circumaural headphones through which they heard their own voice at 85 dB SPL. During these utterances, the talkers heard their own voice from the microphone played back over the headphones at 85 dB SPL. The talkers removed the headphones for the final 30 utterances. For each utterance, the vowel was segmented and measures of duration, fundamental (F0), and the first two formants (F1, F2) were calculated. While wearing headphones did not affect F0 or F1, it did increase the duration of the vowel and decreased F2.

In the second experiment, 15 undergraduate females with clinically normal audiograms were prompted to say the word "head" into a microphone a total of 120 times with their normal speaking voice. The talkers received normal auditory feedback (i.e. no headphones) during the first 20 utterances. For the next 20 utterances, the talkers wore circumaural headphones through which they heard their own voice at 85 dB SPL. The talkers kept wearing the headphones for another 40 utterances. However, during these utterances, a real-time signal processing system was used to increase F1 by 200 Hz. Thus, when the talker said the target vowel in the word head, she heard herself saying a different vowel (which sounded more like "had"). Finally, the talkers were asked to remove the headphones and produce 40 more utterances of the word "head". The results showed that talkers spontaneously compensate for the formant perturbation, by decreasing the F1 of their utterances.

The results will be discussed in terms of the relationship between speech perception and production and the implications for hearing aid design.

A25

Fast Model-Based Fitting through Active Data Selection

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The current trend of model-based fitting refers to setting hearing aid tuning parameters (e.g. compression ratios, thresholds and time constants) to values that maximize a utility metric averaged over a representative database. To this end, new utility models are being developed that aim to predict perceived speech quality and/or intelligibility ratings for hearing aid patients, e.g. [Beutelmann et al., 2006], [Kates, 2005], [Rhebergen et al., 2005]. As a rule, large databases of preference data are needed to train these complex utility models. Still, individual patient preferences may differ substantially from (on average mostly correct) learned model predictions, [Kleijn, 2008]. This would indicate that a large set of preference data for an individual patient is needed to train his unique utility model. Clearly, this is a situation that is not conducive to an efficient audiology practice.

In this paper, we present a novel approach to accurate model-based fitting that needs few measurements from an individual patient. Our method is based on the observation that, for fitting purposes, we are not interested in an accurate utility model for all possible tuning parameter values. Instead, we are only interested in the values for the tuning parameters that maximize the utility model. This implies that away from the maximum, much less model accuracy and hence much less training data is needed. Moreover, rather than learning utility models from a large fixed database of preference data, we actively select only training data that contributes to finding the location of maximum expected utility.

We will show that our active data selection procedure enables accurate utility-model based fitting for individual patients in much less time than the current approaches.

Acknowledgment:

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A26

Combination Open Ear Instrument for Tinnitus Sound Treatment

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Tinnitus is a serious condition reducing the quality of life for a large number of people. In 2006 Del Bo et al. reported promising results by integrating open ear amplification with Tinnitus Retraining Therapy (TRT). The aim of this study was to clinically evaluate the efficacy of sound stimulation delivered by a new open ear combination prototype hearing instrument integrated with TRT. This instrument consists of an amplification system with state-of-the-art signal processing and an advanced sound generator. This combination has been chosen to obtain optimal compensation for the subjects hearing loss and provide the most effective sound enrichment for use in e.g. TRT. The sound generator included a number of unique features; a white noise sound generator with flexible frequency shaping capabilities and manual level control; a random amplitude modulation feature and an environmental steering feature. The amplitude modulation feature was designed to make the noise signal less monotone, while the environmental steering feature ensures that the noise signal is only applied in certain quiet situations. The study was designed as a multi center study with 39 tinnitus patients (15 female and 24 male with mean age of 59 years) falling within Jastreboff 's tinnitus category 1 and 2 (Henry et al. 2002). The mean of hearing loss (PTA) was 25 dB HL. After fitting the instruments TRT was administrated for 6 months and the effect of the treatments was evaluated using the Structured Interview (Jastreboff et al. 2002) and THI self-administered questionnaire (Newmann et al. 1996) after 3 and 6 months.The evaluation results show that significant improvements were obtained within 3 months for the THI scores and all Structured Interview scores with exclusion of “Life effect” which requires 6 months to reach a significant improvement. The sound generator construction and detailed evaluation results will be presented.

A27

Predictors of Hearing-Aid Ownership and Success by Older Adults

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Most older adults who could derive benefit from amplification are not successful hearing-aid users. Two studies were undertaken to provide audiologic descriptions of hearing-aid users and non-users, and successful hearing-aid users, and to determine factors that predict hearing-aid ownership and success. The first study was an analysis of a large database of older adults (N=530). Predictor variables included demographics, peripheral and central auditory function, speech recognition, and self-assessment of hearing handicap. With hearing-loss effects controlled, only hearing handicap differed between owners and non-owners of hearing aids, with owners reporting significantly more hearing handicap than non-owners. With regard to success, only pure-tone thresholds revealed significant differences. Successful users of hearing aids had poorer better-ear and worse-ear thresholds than unsuccessful users.

In addition to demographic and audiologic factors, attitudes toward hearing loss and expectations from hearing aids may contribute to hearing-aid success. Therefore, a second study was undertaken wherein self-report questionnaires were administered to hearing-aid candidates (N=164). Hearing-aid owners completed the Satisfaction with Amplification in Daily Life questionnaire (SADL), the Attitudes toward Loss of Hearing Questionnaire for hearing-aid users (ALHQ), and an 8-item questionnaire related to hearing aids, satisfaction, and usage. Subjects who did not own a hearing aid completed the ALHQ designed for non-owners. Among hearing-aid owners, most were experienced users who generally wore their hearing aids regularly with little difficulty, but only 24% were successful hearing-aid users, according to an overall satisfaction rating. For hearing-aid owners, satisfaction with hearing aids was not significantly different for males and females and did not correlate with age, better-ear or worse-ear hearing, hearing asymmetry, or self-report hearing handicap. Greater hearing-aid satisfaction was associated with better unaided word recognition in quiet and low-context sentence recognition in noise. Users of two hearing aids reported significantly more satisfaction than users of one hearing aid.

Using the ALHQ, hearing-aid owners reported significantly more social and emotional impact of their hearing loss than non-owners and male owners reported significantly less adjustment to their hearing loss than non-owners, suggesting that these attributes are characteristic of individuals who seek treatment for their communication problems. Consistent with this assumption, non-owners reported significantly more denial of hearing loss and a non-significant trend for greater stigma of hearing loss than owners. More denial and greater stigma of hearing loss were also consistent with significantly less hearing handicap reported by non-owners than owners in the first study.

[Work supported by NIH/NIDCD]

A28

How Does the "Acoustic Ecology Differ Among Hearing Aid Users, Cochlear Implant Recipients, and Persons with Normal Hearing

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In this study, the “acoustic ecology” of three groups of persons was studied, using the data-logging feature of several different commercially-available hearing aids. The three subject groups were as follows

1) Persons with normal hearing thresholds,
2) Persons with mild- to severe- sensorineural hearing loss, and
3) Cochlear implant recipients (monaural and binaural)

As hearing aid and cochlear implant technology has converged in recent years, the use of automated signal processing strategies has become more common, both for hearing aids and cochlear implant external processors. At issue is whether the “typical” listening environments encountered is similar across patients, in terms of the average or range of intensities, distribution of “quiet”, “noisy”, and “music” environments, and the use of directional or omni-directional microphones. Twenty-five subjects per group each wore a “datalogging” instrument for a period of one week, after which these data were compiled and compared between groups. The impact on hearing aid and cochlear implant design will be discussed.

A29

Evaluating a Listener-Driven System for Fitting Hearing Aid Algorithms

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Standard audiological tools can be inadequate for fitting hearing aid algorithms that have many interacting parameters, particularly when the perceptual consequence of the parameters is complex and the interaction with hearing loss unknown. Patients are unable to actively participate in the refinement of the algorithm parameter settings, and exploration of a high-dimensional parameter space is a lengthy and difficult process. Optimal fittings of such algorithms to a patient's individual loss are, therefore, unlikely.

We have implemented a listener-driven interactive system for navigating the high dimensional parameter space of hearing aid signal processing algorithms. The system allows listeners to compose a two-dimensional subjective space of parameter settings, and provides for smooth, real-time interpolation among the settings. The most common method for generating such a spatial representation is the multidimensional scaling (MDS) of pair-wise dissimilarity judgments. As an alternative to the MDS method we have found that directly arranging the stimuli in a subjectively meaningful spatial layout provides representations comparable in quality to MDS. In this subjective space, we have found that a large number of stimuli can be arranged much more rapidly than is possible using the MDS method with pair-wise dissimilarity judgments. The system allows intuitive exploration of high dimensional parameter spaces, and captures rich data structures from its users that can be used for understanding individual differences in hearing impairment as well as significant correlations between parameter settings and perceptual changes in the processed sound.

We present the results from an ongoing sequence of experiments to determine whether hearing-impaired listeners are able to use the subjective space navigation system to find preferred or optimal parameter settings for hearing aid signal processing algorithms. In separate, repeated trials, our subjects use the subjective space to configure parameters of three different varieties of signal processing algorithms. For each algorithm, we evaluate 1) the consistency with which subjects organize the subjective space to reflect perceptual dissimilarity between preset algorithm parameter configurations, and 2) the consistency with which subjects identify an optimal interpolated configuration of the algorithm parameters. Their performance is compared with the performance of normal hearing subjects on the same tasks, and we evaluate the efficacy and usability of the system as a tool for configuring or fitting different signal processing algorithms in hearing aids.

A30

Sensitivity to Interaural Time Differences with Bilateral Bimodal Stimulation

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The interaural time difference (ITD) is an important cue to localize sound sources. Changes in ITD of about 10us can be detected by normal hearing (NH) subjects in low frequency sinusoids. Recent studies have shown that users of bilateral cochlear implants (CIs) are also sensitive to ITDs, although much less than NH listeners. Best just noticeable differences (JNDs) reported for pulse trains of about 100pps are around 100-200us. For CI users with residual hearing in the contralateral ear (bilateral bimodal stimulation), combined electric-acoustic stimulation may be a feasible alternative to bilateral cochlear implantation. Sensitivity to ITD was measured in 8 users of a cochlear implant (CI) in the one ear and a hearing aid (HA) in the other severely impaired ear. The stimulus consisted of an electric pulse train of 100pps and an acoustic filtered click train. The electric pulse train was presented on either an apical, medial or basal electrode and the acoustic signal had cutoff frequencies of either 400-800Hz, 800-1600Hz, 1600- 200Hz or 3200-6400Hz. JNDs in ITD were measured for every combination of the electric and acoustic signals using a lateralization paradigm. Four subjects exhibited JNDs in ITD down to 57, 91, 91 and 155 us, the other subjects could not lateralize the stimuli consistently. Lateralization performance using ITDs was related with the average acoustic hearing thresholds at 1000 and 2000 Hz. For the electric and acoustic signals to be perceived synchronously, a delay has to be introduced into the electric pathway because of the traveling wave delay of the acoustic signal. From the same lateralization experiments, this delay was found to be 1.4ms, which is consistent with the results obtained in auditory brainstem response (ABR) and electric ABR (EABR) studies.

A31

Applications of the International Speech Test Signal (ISTS) On Dynamic Compression and Noise Reduction Algorithms

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Since standardized measurements procedures and test signals are not able to adequately characterize modern nonlinear digital hearing aids, an International Speech Test Signal (ISTS) was developed for worldwide application. The features of the ISTS are shown in another contribution to this meeting (”Development and Analysis of an International Speech Test Signal (ISTS)” by Holube, Fredelake and Vlaming). It is intended to standardize the ISTS together with a new measurement procedure based on a percentile analysis of the input and output of hearing instruments. This analysis method is presented in another contribution to this meeting (”Characterization of Speech Amplification for Modern Hearing Aids” by Vlaming). In this study the ISTS as well as the new measurement procedure have been applied to simulated hearing aid algorithms as well to commercial hearing instruments. The outcomes of the measurement procedure are the overall gain of the hearing instrument as well the gain for soft and loud speech tracts. From these outcomes, the effective compression ratio can be derived using different approaches. The results are compared to each other and to the static compression ratio determined with conventional measurement procedures. The hearing instrument output as measured for the ISTS can be compared to the hearing threshold of the respective hearing-impaired listener. In addition, the ISTS has also been applied to nonlinear noise reduction algorithms. The ISTS served as speech signal and the noise signal was generated by a superposition of 90 segments of the ISTS with randomly chosen time delays and was called International Female Noise (IFnoise). ISTS and IFnoise were added with different SNRs and processed by several noise reduction algorithms. Furthermore, the noise reduction algorithms processed the ISTS and the IF-noise separately with the same filtering parameters derived from the noisy signal. Therefore, it was possible to estimate the effect of the noise reduction algorithms on speech and noise separately. The output of the noise reduction algorithms was investigated by the percentile analysis and compared to some other objective measurement procedures, e.g. from Hagerman and Olofsson.

[This work was supported by AGIP/EFRE and Audiologie-Initiative Niedersachsen]

A32

A Perceptual-Learning Investigation of Auditory Amplitude-Modulation Detection: Testing the Existence of Frequency-Selective Mechanisms in the Temporal-Envelope Domain

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The importance of the slow amplitude modulations (AM) of speech is demonstrated by near-normal speech identification with preserved temporal-envelope but degraded temporal fine-structure and spectral cues. Evidence from psychophysical and electrophysiological studies suggests that AM processing is modulation-frequency (f_m) specific, either in terms of different coding mechanisms for slow and fast AM, or a bank of selective AM channels, each tuned to a different f_m.

In a previous study (Füllgrabe et al., IHCON 2006), we used a perceptual-learning paradigm to test the existence of either form of selectivity in the AM domain. It was reasoned that if selectivity exists, training on AM detection using a single f_m should improve post-training detection thresholds (relative to pre-training thresholds) for the trained but not for the untrained f_m; if no selectivity exists, learning should generalize to untrained f_m. Results showed a uniform decrease in thresholds for trained and untrained f_m that largely remained when retested after ten weeks. This observation was consistent with the notion that sensitivity was limited by internal noise at a post AM-channel stage and that the f_m-unspecific effect reflects a reduction in this noise. Possible training-induced selective changes in AM channels (e.g. narrowing of the bandwidth) might have occurred, but could not be demonstrated given the use of unmasked AM.

In the present study, the previous experiment was repeated using AM presented within a notched noise in the AM domain, making it more likely that sensitivity would be limited by noise in the AM channels, and allowing effects of improved selectivity to be revealed. Sensitivity to AM was measured using an adaptive 3-interval, 3-alternative forced-choice procedure and a 3-down 1-up stepping rule. Twenty listeners were trained for eight daily 1-hour sessions to detect a 5- or 97.2-Hz AM applied to a 4-kHz carrier. Improvements similar to those previously obtained were observed for trained f_m, but less learning occurred for untrained f_m. This pattern of thresholds is consistent with selectivity in the AM domain and persisted seven weeks after training.

Taken together, the data from both studies demonstrate long-lasting AM-detection learning in the adult auditory system. The fact that the specificity of this improvement depended on the type of training stimulus might be important for the design of rehabilitation strategies.

A33

Effects of Speech Bandwidth on Sound-Quality Preferences for Hearing-Impaired Listeners

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

Most cases of sensori-neural hearing loss are characterized by the poorest audiometric thresholds occurring at high frequencies. However, the frequency response of most commercially available digital hearing aids (HA) starts to roll off at about 5-6 kHz, as a result of the low sampling rate in older HA, tubing effects for behind-the-ear HA, and problems with acoustic feedback due to the amplification of high frequencies. Furthermore, it was reported that high-frequency amplification can lead to degraded speech recognition in hearing-impaired (HI) persons with moderate-to-severe hearing losses.

The aim of the present study was to assess the effect of extended bandwidth on sound-quality in listeners with mild-to-moderate sensori-neural hearing losses. Stimuli were 5-s speech excerpts taken from recordings of three female and three male talkers, lowpass filtered at 10, 7.5, or 5 kHz. An extended version of the CAMEQ (Moore et al., 1999, Br. J. Audiol. 33: 241-258) fitting method was developed to select appropriate gains at high frequencies. This new fitting method aimed to satisfy the following criteria: (i) the overall loudness of speech at medium to high levels should be similar to that evoked in a normal ear with unamplified stimuli; (ii) for frequencies from 0.5 to 5 kHz, the specific loudness should be roughly constant (equal loudness/ERB_N – the same criterion as for CAMEQ); (iii) for frequencies above 5 kHz, audibility should be partially or completely restored, while avoiding a specific loudness that is markedly greater than that evoked in a normal ear by unamplified stimuli.

Using a round-robin paradigm, listeners compared preferences for lowpass-filtered amplified speech tokens delivered via one ear piece of a Sennheiser HDA200 headphone. Instructions were given to focus on the clarity of the speech and not its pleasantness.

Four of 12 HI listeners tested so far showed greater preference for male and female speech with larger bandwidth. Conversely, speech with narrower bandwidth was preferred by three listeners, possibly reflecting judgments based on pleasantness. The remaining listeners did not report hearing any differences. Future studies need to determine if "positive" effects of an extended bandwidth observed in some HI listeners can also be obtained in objective speech-perception tasks, and if the pleasantness of amplified stimuli depends on acclimatization.

[Work supported by Marie-Curie Fellowship and MRC (UK)].

Posters for Session B should be put up by 8 A.M. Friday, August 15, and taken down after 10 P.M. Friday, August 15 or before 7 A.M. Saturday, August 16. Presenters should be at their posters from 9:45 – 11:00 A.M.; 4:30 - 5:00 P.M.

POSTER SESSION B

Friday 8:00 AM –10:00 PM

B1

The Effect of Room Volume on Speech Recognition in Enclosures with Similar Mean Reverberation Time

Jason Galster, Vanderbilt University

This project investigated speech recognition in rooms of different size with similar mean reverberation times. A comparative analysis of existing literature has provided evidence to support that speech recognition in small rooms may be poorer than in larger rooms when the two spaces have a similar amount of reverberation. This study evaluated speech recognition with sentences binaurally recorded using an acoustic manikin in three rooms of different volume and/or dimension. The three rooms included a small reverberation chamber (48 m3), a university lecture hall (479 m3), and a high school band practice room (474 m3). Speech recognition was tested using bilateral insert earphones in two groups with 13 participants in each group. The first group consisted of individuals with normal-hearing and the second group consisted of participants with mild-to-severe hearing impairment. All testing was completed at five signal-to-noise ratios (SNRs). Several measures, selected to quantify the acoustic characteristics of each room, were collected: mean free path, frequency-specific reverberation times and the Speech Transmission Index (STI).

This investigation determined that listeners in both groups showed both a significant decrease in speech recognition performance as SNR decreased as well as a significant effect of changing room size. The poorest speech recognition was measured in the smallest room. There was no interaction between SNR and room type for either of the two participant groups. The effect of both change in room size and SNR correlated with changes in measured Speech Transmission Index.

A rationale was proposed as the source of the room size-specific reverberation effects. This idea speculates that the period during which early reflections are beneficial to speech understanding may decrease as room size increases. This is consistent with measures of decreased mean free path in smaller rooms. In addition, the reverberant field of a small room will contain more reflections than a larger room when the two are matched for reverberation time. It is proposed that the increased number of overlapping reflections also contributes to decreases in speech recognition ability.

B2

The Noise Reduction Index as a Predictor of Benefit on a Speech-In-Noise Task

Robert Ghent, Michael Nilsson, Michelle Hicks, & Victor Bray, Sonic Innovations, Salt Lake City, UT

The Noise Reduction Index (NRI) has been shown to be a useful bench-top measure of the signal-to-noise ratio (SNR) change through a hearing aid. Whereas the NRI is based on the recorded speech and masking materials developed for the Hearing In Noise Test (HINT) and, whereas the HINT is used for behavioral testing in the same sound field conditions under which NRI measurements are obtained, a study was undertaken to evaluate the predictive value of the NRI on

HINT benefit for aided, hearing-impaired individuals.

Unaided and aided HINT scores were obtained in two-dimensionally diffuse noise on 20 hearing-impaired subjects and a difference score was taken to represent HINT benefit re: the unaided condition. Aided HINT scores were obtained on the subjects with 24-channel WDRC hearing aids programmed with a feature set designed to individually optimize speech understanding in noisy environments. Prescribed fittings were verified using probe microphone measures. NRI values were then obtained on the hearing aids, programmed with both the subjects' prescribed fittings as well as with a flat, linear fitting with 15 dB of gain. The feature set with which the NRIs were obtained was the same as that prescribed for individual subjects to use for optimal benefit in noisy environments. Use of a flat, linear fitting as an alternate to an individual's prescribed fitting was evaluated as a means for clinicians to predict benefit from a generic NRI 'specification' without having to obtain NRI values for individual fittings.

Results will be presented that address two research questions:

- What portion of individual HINT performance/benefit can be predicted by NRI values obtained with that individual's hearing aids?
- Are predictive relationships better explained by NRI values obtained with subjects' fittings or with a generic flat, linear fitting?

B3

Impact of Noise on Estimates of Compression in Normal Hearing Listeners

Melanie Gregan, Andrew Oxenham, and Peggy Nelson, University of Minnesota, USA

Poor performance of hearing-impaired (HI) listeners on certain psychoacoustic tasks may be related to loss of cochlear nonlinearity. Similarly degraded performance is seen for normal hearing (NH) listeners in certain tasks when a background noise is added that equates thresholds with those of HI listeners. It is not clear if this reduced performance in noise-masked NH listeners is attributable to reduced audibility or if the background noise creates a more linear cochlear response. There were two primary questions: 1) Does threshold elevation using a steady broadband noise result in more linear estimates of compression in NH listeners? 2) Does additivity of forward (AFM) masking provide an estimate of compression that is comparable to those derived from forward growth-of-masking (FGOM) functions?

FGOM functions and AFM were measured for six NH adults. For FGOM, masker levels at threshold were obtained for several fixed signal levels at a single brief masker-signal delay. The signal was a 4000-Hz tone and the masker was either an 1800-Hz or 4000-Hz tone. FGOM functions were obtained in quiet and in 3 levels of threshold equalizing noise (TEN). Slopes of the behavioral input/output (I/O) functions were used as an estimate of compression. For the AFM paradigm, two contiguous forward maskers (M1 and M2) were set in level so that they each produced similar masked thresholds for the brief 4-kHz signal when presented in isolation. Signal thresholds were then measured in the presence of combined maskers. The maskers, M1 and M2, were octave bands of noise centered around 4000 Hz, which were presented either in quiet or in TEN. The amount by which signal thresholds increase when both maskers are combined can be used to estimate compression, with a greater increase in threshold implying more compression.

Results from both paradigms indicate no strong effects of background noise on compression estimates. At low signal levels, both paradigms indicate substantial compression; however, at high levels, the AFM paradigm suggests a more linear response, with or without noise, whereas the FGOM estimates remain compressive.

Compression estimates using FGOM and AFM measures indicate that poor performance for noise-masked NH listeners is probably not due to more linear cochlear response growth, suggesting that broadband noise does not necessarily have a "linearizing" effect on cochlear response. Follow-up studies will assess the potential role of efferent activity in producing the discrepant compression estimates between FGOM and AFM at high levels. [Supported by NIH R01 DC 03909].

B4

Combined Binaural and Monaural Feedback Cancellation Strategies: Algorithms and Validated Measures

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The effect of a binaural coherence-based noise reduction scheme on the feedback stability margin and sound quality in hearing aids is analyzed. For comparison, a conventional adaptive feedback canceller (AFC) and the combination of the adaptive filter with the binaural coherence filter are tested. An objective measure of feedback stability, i.e., the added stable gain (ASG) was obtained for a number of algorithmic settings. The ASG is the difference of the maximum stable gain between the conditions with and without the respective feedback reduction algorithm. To validate this objective measure, it is compared to a subjective measure of feedback stability (added tolerable gain, ATG): A broadband gain was applied to the input signal and normal hearing subjects were asked to increase the gain in the feedback loop as far as possible, with the speech quality still being acceptable. To eliminate the subjective bias in estimating the ATG, the "unbiased added gain" (UAG) is introduced and discussed as a new method of subjective evaluation. It is not based on quality rating at a given gain, but reveals the maximum possible gain with a given quality threshold.

The results with the different algorithms indicate that a binaural coherence filter can substantially increase the feedback stability if it is combined with an AFC method. Moreover, the unbiased, new subjective measure agrees quite well with the objective measure of the ASG. This allows for a valid comparison across different feedback reduction schemes both in isolation and in combination: While the ASG of the coherence filter without combination with AFC is negligible, the superior performance of the combination indicates that a robust feedback suppression in hearing aids can be achieved if the benefit of de-correlation and the head-shadow effect in binaural hearing aids is used in an advantageous way. The ASG reaches 23 dB for the best combination at the expense of an average target signal attenuation of 15 dB at critical frequencies. The specific contribution of the coherence filter is that it adaptively limits the maximum gain before feedback becomes audible.

[This work was supported by BMBF 01EZ0212, EU FP6/004171 and Spanish ministry of science and education]

B5

A Transcutaneous Bone Conduction Implant System – A Future Alternative to the Percutaneous BAHA System?

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It is well-known that the percutaneous bone anchored hearing aid (BAHA) offer an important rehabilitation alternative for patients suffering from conductive or mixed hearing loss. Beside comfort aspects the main advantages of the BAHA over conventional bone conduction devices are related to that the transducer is directly attached via a skin penetrating titanium fixture to the skull bone.

Even if the bone anchored titanium fixture and the concomitant skin penetrating abutment have a reasonably low complication rate there are some drawbacks associated. Some implants are spontaneously lost and must be replaced. There are also sometimes skin complications that must be treated. Also, in some cases patients who are good candidates for a BAHA rejects the implant for psychological or cosmetic reasons.

Since long time it has been suggested that the advantages with direct bone excitation and keeping intact skin could be combined in a transcutaneous bone conduction implant (BCI) system where the implanted transducer is placed in the temporal bone and where the signal and power are transmitted with an induction loop system.

Several studies have been made to find out if one such a transcutaneous BCI system could be an alternative to a percutaneous BAHA or not. Investigations comparing a transcutaneous BCI system and a percutaneous BAHA have been made on a Skullsimulator, a dry skull and a cadaver head. The results indicate that a transcutaneous BCI system can be a future alternative to a percutaneous BAHA. A summary of these results will be presented.

B6

The Impact of Directivity Index Manipulations on HINT Performance

Michelle Hicks, Brenda Fedor, and Michael Nilsson, Sonic Innovations, Salt Lake City, UT

The purpose of this investigation was to determine whether the degree of directivity in a commercially available directional hearing aid, as represented by the three-dimensional Directivity Index (DI) (ANSI S3.35-2004), correlates to speech perception in noise, as measured with the Hearing in Noise Test (HINT). Although there is some evidence that this relationship exists (Ricketts, 2000; Ricketts et al. 2001), the different subject pools and various test conditions in previous experiments makes definitive conclusions difficult. Thus, this study was completed to systematically manipulate directivity to determine the effect on directional benefit.

Directional hearing aids are intended to improve the signal-to-noise ratio (SNR) for listeners in noisy situations. DI provides an estimate of the effective SNR for the situation where the signal of interest is in front of the listener with diffuse noise all around them. There is a theoretical "ideal" polar pattern and corresponding DI value, though the actual value obtained when built into a device and measured in the free field will likely be somewhat less (Bentler et al., 2004). Indeed, a recent investigation (Howell and Saunders, 2008) has demonstrated that there is significant variability in the degree of directivity from directional hearing aids as delivered by various manufacturers.

For this study, BTE hearing aids were programmed with a linear 15dB gain and a fixed hypercardioid polar pattern with all automatic, adaptive and noise reduction algorithms turned off. Three-dimensional polar plots were measured on KEMAR in the free field and DI calculations were completed. For reference, two-dimensional (2D) polar plots of the devices in a free field by themselves were also measured. By mismatching the two omni microphones in BTE devices in a systematic way, the degree of directivity was varied to achieve 2D free-field DIs of 3.0, 4.0, 4.5, 5.0, 5.5, and 6.0 dB. HINT performance was then measured in a diffuse noise environment with the speech presented at 0 degrees and four uncorrelated noise sources at 45, 135, 225, and 315 degrees. Subjects repeated the HINT testing several times using the hearing aids set to a variety of DI values. The relationship between DI and HINT scores was evaluated and the results are presented to provide more evidence about how effective the DI is at predicting real-world speech-in-noise benefit.

B7

The Importance of Temporal Fine Structure Information in Speech at Different Spectral Regions

Kathryn Hopkins and Brian C. J. Moore, Department of Experimental Psychology, University of Cambridge

Temporal fine structure (TFS) information in speech appears to be most important when listening in a fluctuating background. TFS could be important for accurate estimation of the fundamental frequency of a speaker's voice, which could improve intelligibility in noise by aiding segregation of the target and background into different auditory streams. TFS may also be important for identification of portions of the target speech in the dips of a fluctuating background, where the signal-to-background ratio is most favourable.

Hopkins *et al* (2008) assessed the benefit from TFS information for normal-hearing and hearing-impaired subjects when listening in a competing talker background. They found that intelligibility for normal-hearing subjects improved as TFS information was added to a vocoded signal for frequencies up to around 4000 Hz. Results for hearing-impaired subjects varied between individuals, with one showing a benefit similar to that for the normal-hearing subjects and others showing no benefit at all. As a group, no improvement in performance was seen when TFS information was added at frequencies above 1600 Hz.

Here the study by Hopkins *et al* was extended to more accurately identify the frequency region where TFS information is most important for normal-hearing subjects. The signal to background ratio that was needed for 71% correct performance was measured in a competing talker background. Target and background speech was combined and filtered into 30 1-ERB_N wide channels between 100 and 8000 Hz. Channels were separated into two frequency regions by a cutoff channel (CO). Performance was measured when the high-frequency region was tone vocoded and CO was increased and when the low-frequency region was tone vocoded and CO was decreased. Five values of CO were tested in each case. Consequently, performance was measured as TFS information was progressively added, starting at either the high or low end of the frequency spectrum.

As TFS information was added starting at low frequencies, performance improved, consistent with the results of Hopkins *et al* (2008). There was no significant improvement in performance as TFS was added to channels with centre frequencies above 2000 Hz. As TFS information was added starting at high frequencies, performance also improved, but this improvement was small until TFS information was added to the frequency region around 400-1000 Hz. We suggest that TFS information in this frequency region is the most important for improving speech intelligibility in a competing talker background.

Hopkins, K., Moore, B. C. J., and Stone, M. A. (2008). "The effects of moderate cochlear hearing loss on the ability to benefit from temporal fine structure information in speech.," J. Acoust. Soc. Am. **123**, 1140-1153.

B8

Effect of Hearing Aid Use on Cognitive Processing and Listening Effort in Everyday Settings

B Hornsby, Vanderbilt University Medical Center, Nashville, TN

Although poorer-than-normal speech understanding is common in many difficult listening situations, listeners with mild-to-moderate hearing loss often show essentially normal speech understanding in "easy" everyday listening conditions. Maintaining good speech understanding in these situations, however, may require allocation of more cognitive resources and more effortful listening for persons with hearing loss than for individuals without hearing loss (e.g. McCoy et al., 2005). This extra cognitive effort may lead to additional mental stress that can negatively affect quality of life and work experience for persons with hearing loss (Kramer et al., 2006).

In this study we examine the effects of hearing aid use on cognitive processing, and indirectly listening effort, in everyday situations. Cognitive processing will be tested multiple times during the day, to assess changes in cognitive demands and perceptual effort throughout the day. To assess cognitive competence, the MiniCog Rapid Assessment Battery (MRAB; Shephard and Kosslyn, 2005; Shephard et al., 2006) will be used. The MRAB was developed specifically to allow individuals to rapidly assess cognitive processing in adverse environments (space, Antarctica, Mount Everest...) using a handheld PDA. The MRAB consists of nine tests from the cognitive/neuroscience literature that assess various cognitive abilities (e.g. attention, working memory) via measures of accuracy and reaction time. In addition, prior to each MRAB session, participants will complete a 3 item questionnaire asking them to rate recent listening effort and demands.

Cognitive processing, assessed via the MRAB, will be monitored in 16 participants with mild-moderate hearing loss over a three week period. All participants will be existing part-time hearing aid users. Baseline data will be obtained during an initial 1 week training period. During the first week, participants will wear hearing aids full time and complete the MRAB three times/day (morning, afternoon and evening). During the second week participants will again complete the MRAB three times/day; however, participants will be counterbalanced into an unaided or aided condition (aids will be worn full time or not at all). During the third week participants will crossover into the opposite condition (unaided or aided) and again complete the MRAB three times/day.

An analysis of results will focus on the effects of time of day and hearing aid use on cognitive processing and, indirectly, listening effort. In addition, relationships between subjective ratings of listening demands and objective measures of cognitive processing will be explored.

This research is supported, in part, by a grant from Starkey, Inc.

B9

Construction of a Virtual Subject Response Database to Reduce Subject Testing

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This paper addresses the question whether the selection of responses from a database can replace real subject responses that are required for the development of an efficient interactive fitting procedure. This question arose because subject data is expensive in terms of both time consumption and subject burden, and because we needed subject data for the development of a novel interactive hearing-aid fitting procedure. This new fitting procedure (called HearClip, see De Vries et al., this conference) is based on user preference and uses model-based fitting. In order to minimize the number of subject experiments we constructed a database from which we can conveniently draw realistic subject responses. These simulated responses are based on data that was gathered in a round-robin experiment with a two-alternative forced-choice paradigm. In the experiment the noise floor (Gmin) of a noise reduction algorithm (single-channel spectral-subtraction) was varied from 0 to +14 dB in steps of 2 dB. We used speech in stationary noise (HINT corpus, all non-native listeners) at a signal-to-noise ratio of +5 dB. Normal-hearing subjects were asked which sound sample they preferred. In order to estimate measurement error, learning effects, and patient fatigue, we conducted the entire experiment six times. From this large amount of measurement data we constructed 'virtual subjects'. These virtual subjects give realistic responses that are based on the distribution of the actual responses and include learning effects, inter and intra subject variability, and conflicting answers. The paper discusses the results and evaluates the use of the database to minimize costly subject testing. [We gratefully acknowledge financial support from STW (Dutch Foundation for Science and Technology) for the HearClip project.]

B10

"Hearing At Home" - Developing Assistive Technology for the Everyday Life Of Hearing Impaired People At Home

Rainer Huber, HoerTech, Oldenburg, Germany

The Hearing at Home (HaH) project focuses on the needs of the hearing-impaired in home environments. It anticipates that formerly separated devices like personal computer, hi-fi systems, TV, digital camera, telephone, fax, intercom and services like internet access, Voice over IP, Personal Information Management, Pay TV and home automation will grow together and will be accessible via a TV-set that is connected to a PC or set top box. The TV might thus become the »central Home Information and Communication (HIC) platform« of the household. The HaH project will support the hearing-impaired by developing easy-to-configure "supportive audio signal processing (SASP)" for HIC platforms in combination with visual support on the TV screen.

This contribution gives an overview of the HaH project, focusing on those parts of the project that deal with SASP strategies using

the "Master Hearing Aid" as a realtime signal processing environment. For the application in the home environment, amplification schemes and noise reduction schemes known from hearing-aid processing need to be further developed and adapted to the specific needs. One particular complication is the need of an "intelligent" mixing of different overlapping sound sources to be attended by the hearing-impaired listener, e.g., voice of another person and TV sound. Another complication is that the type of the sound to be attended might change often and abruptly, e.g., the TV program itself (incl. commercial breaks) or the TV program interrupted by sound sources outside the HIC platform. This requires advanced classification schemes for detecting the sound environment. Possible solutions for the adaptation of the signal processing schemes to the home environment and first results will be discussed.
[This work is supported by EU FP6/IST-045089]

B11

Using Finite Element Modeling To Estimate the Influence of Pinna When Calculating Hearing Aid Relevant Transfer Functions

Mads J. H. Jensen and Morten P. Linkenkær,

Widex A/S

In many hearing aid applications, detailed knowledge of specific transfer functions is important. These include the acoustic feedback path for feedback cancellation applications, the characterization of the vent effect, and the directly transmitted sound path. This information is used for fitting purposes as well as for occlusion management strategies. Traditionally the transfer functions are determined either by using a two-port transmission-line like model (the plane-wave assumption which may be too crude in complex geometries, such as the pinna) or by direct measurements (which may be useful mainly for documentation purposes). Modeling all physical details and having the flexibility for optimization may, e.g., be achieved by a full numerical finite element (FEM) solution of the acoustic problem. Such a method may, however, be computationally heavy.

In the current study, we have used a model of a specific BTE hearing aid placed on a specific pinna with a specific mold. For the purpose of the study, we use the less computationally intensive transmission line model (TLM) for the acoustics of the tubing sections (tubing, vent, etc.) in combination with the accurate full FEM simulation of the acoustics around the pinna and hearing aid. By using this method, we may flexibly and accurately simulate any transfer function relating to the hearing aid. The TLM/FEM model allows for a systematic study of the influence of vent placement and vent size on the different transfer functions. Furthermore, the microphone placement in or behind the complex geometry of the pinna is investigated with regard to the feedback path. These simulations give valuable input to optimal positioning of the hearing aid microphones for minimizing the risk of feedback. The effects related to specific microphone location are most evident for frequencies over about 4 kHz where the pinna features are comparable to the wavelength. Moreover, the finite element model may be used to optimize the TLM such that it includes all the acoustic features of the pinna. Simulations are verified by actual measurements on a test system consisting of a rubber ear, coupler, and hearing aid. The advantage of using the FEM/TLM method is exemplified by comparing various transfer functions calculated by using the classical plane wave two-port model, measured data, and the FEM/TLM method.

B12

Properties of the Distinctive Features Differences Test for Hearing Aid Research

Lorienne M. Jenstad, PhD, Sarah Barnes, BA, University of British Columbia, Donald Hayes, PhD, Unitron Hearing Ltd.

We evaluated several properties (namely, magnitude of practice effects, and acoustic properties of individual speech segments) of the University of Western Ontario Distinctive Features Differences (UWO-DFD; Cheesman & Jamieson, 1996, adapted from the original DFD test, Feeney & Franks, 1982) test to establish protocols and procedures for evaluating hearing outcomes both behaviourally and acoustically.

The UWO-DFD test consists of digitized recordings of 4 talkers (2 male, 2 female), speaking nonsense syllables in the form a/C/il, where /C/ is the target consonant sound, selected from 21 possible consonants. The test is characterized as having a shallow performance/intensity function, making it sensitive to changes in listening conditions, and also having high test-retest reliability (Cheesman & Jamieson, 1996).

Despite the high test-retest reliability, practice effects have been noted on repeated administrations of the test. Our first purpose was a behavioural evaluation of the magnitude of practice effects on the DFD test to determine how many runs of the test would be required to achieve a stable score and whether the practice effect could be minimized by manipulating listening conditions, namely, the type and level of background noise, and the number of talkers randomized within a block of trials.

Fifty four listeners with normal hearing were recruited for the behavioural study. Listeners were randomly assigned to listening conditions, with different type and level of background noise taken from Auditec of St. Louis, and either a single talker per block of trials or all 4 talkers per block. Participants heard the words presented via headphones. All 21 responses were displayed on the computer monitor and participants were asked to identify which of the words they heard.

Results showed that practice effects were minimal for this task, but did vary significantly across the listening conditions. On average, participants required 4 runs to reach a stable score. Recommendations for will be made for study design using these stimuli in behavioural hearing aid evaluations.

Our second purpose was a full acoustic description of the UWO-DFD words. We have defined important time markers for the words of all four talkers to allow for an examination of the temporal and spectral characteristics of transitions and steady-state portions of each speech segment. This baseline description allows for the quantification of the changes that hearing aid processing makes to individual types of speech sounds. We will present our general approach to the segmentation process and some potential uses of these stimuli in acoustic hearing aid evaluations.

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B13

A Formant Enhancement Strategy Reducing Acoustic Masking Effect

Yuyong Jeon, Se Kee Kil, Sangmin Lee, Department of Electronic Engineering, Inha University, Korea;

Because the purpose of wearing a digital hearing aid is for hearing impaired people to communicate with others, speech enhancement algorithms in digital hearing aids have been developed to compensate hearing loss of hearing impaired people since digital hearing aids are invented. However most of digital hearing aid users still complain that it is difficult to understand speech wearing a digital hearing aid. This can be because the quality of speech through digital hearing aid is insufficient to understand the speech caused by feedback, residual noise and some other reasons. And another reason that makes sound quality poorer can be acoustic masking effect among formants by excessive formant enhancement.

In this study, we measured the basic and masking characteristics of hearing impaired listeners. The experiment is composed of 5 tests; pure tone test, speech reception threshold (SRT) test, word recognition score (WRS) test, pure tone masking (PTM) test and speech masking score (SMS) test. Pure tone test, SRT test and WRS test is basic test being measured in the hospital. SRT test is to find threshold that 2-syllable speech can be heard and WRS test is what how much words can be heard in the most comfortable level (MCL) within 50 1-syllable speeches. In the PTM test, masker is narrow-band (50Hz) noise, probe is pure tone and subjects are required to determine two sounds; masker and sum of masker and probe were same or not. Speeches recorded by a Korean male announcer in his twenty and their formant enhanced versions are used in the WRS test and SMS test respectively.

Because SMS result is lower comparing to WRS result, someone maybe think that this result is because of distortion by excessive formant enhancement. To evaluate the distortion of each speech objectively, log likelihood ratio (LLR) is introduced. LLR is a spectral distance measure of the mismatch between the formants of the original and formant enhanced speech, and the lower LLR indicates the better speech quality.

As a result of PTM, masker on the first formant-frequency masks other components more easily than other frequency maskers. It means that the acoustic masking effect in speech itself is occurred by the first formant frequency. And in the SMS test, the speech perception became lower by formant enhancement however speech perception is not proportional to LLR. It means that acoustic masking effect rather than distortion influences speech perception. Characteristics of masking effect are not similar among each person. So it is required to check the characteristics of masking effect before wearing a hearing aid and to apply these characteristics to formant enhancement algorithm.

To apply this acoustic masking effect to formant enhancement algorithm, gain of each formant is required to control. Because acoustic masking effect in speech itself is mainly occurred by the first formant, we proposed a formant enhancement algorithm reducing gain of first formant frequency. To evaluate this algorithm, the WRS test using formant enhanced speech is proposed and then its results are compared with WRS test using clean speech. As a result of this test, result of WRS test using formant enhanced speech is higher.

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B14

Effect of Extending the Bandwidth of Amplification to High Frequencies for Sound Quality

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While it is becoming possible for hearing aids to give a broader frequency range of amplification than possible in the past, there is little consistent objective evidence that a greater audible bandwidth gives perceptual benefit to hearing-impaired (HI) listeners. This study investigates whether extending the bandwidth of reception to high frequencies gives an improvement of sound quality to HI listeners, as it does for normal-hearing listeners.

We address this question by asking 10 moderate HI listeners to rate their preference in terms of sound quality of different upper frequency limits of amplification (4, 6, 8, 10 and 12 kHz) in paired-comparison trials. Subjects rate the quality of 3 music samples and 1 speech sample, with all samples selected to have significant high-frequency content. Stimuli are amplified linearly according to a high-frequency version of the CAMEQ prescription that compensates for the subject's hearing loss.

Inconsistent findings from past studies regarding the efficacy of increasing bandwidth may be due to insufficient audibility of high-frequency energy. The problem stems from the difficulty of verifying sound levels at the ear drum at high frequencies. The present study addresses verification of audibility by measuring sound levels in each subject with a probe microphone placed approximately 2-3 mm from the ear drum. The proximity of the probe tip to the ear drum helps overcome the across-subject variability in the levels of high-frequency components of sound due to individual differences in ear-canal geometry. The study also verifies audibility by measuring the ability of individual subjects to discriminate the different bandwidth conditions for every stimulus sample used in the assessment of sound quality.

We will discuss the results of the experiment and its implications for the bandwidth of amplification in moderate HI listeners.

B15

A Model of Speech Quality Judgments

James M. Kates Kathryn H. Arehart, GN ReSound, University of Colorado at Boulder

The quality of the reproduced speech is an important factor in the design and fitting of hearing aids. In this paper, models are developed for speech quality judgments made by normal-hearing and hearing-impaired listeners under a wide variety of linear and nonlinear processing conditions. The objective is to be able to use the estimated speech quality to predict processing effectiveness and to establish initial settings for hearing-aid fittings. Speech quality is reduced by noise, distortion, nonlinear signal processing, and linear spectral modifications. In a companion experiment, Arehart et al. (2008) obtained quality judgments from normal-hearing and hearing-impaired listeners for a wide variety of signal degradations including noise and nonlinear processing alone, linear filtering alone, and combinations of nonlinear and linear processing. The nonlinear processing includes additive noise, peak clipping and quantization distortion, dynamic-range compression, and noise suppression. The linear filtering includes bandwidth reduction, spectral tilt, and spectral peaks typical of hearing-aid tubing resonances. The process of modeling an arbitrary combination of signal degradations is decomposed into one model for noise and nonlinear data and a second model for the linear filtering data. The nonlinear and linear models are then combined into a composite model to fit the combined nonlinear and linear processing results. Several candidate modeling approaches are evaluated for the nonlinear processing data, including coherence, envelope time-frequency modulation, envelope modulation spectrum, and the output from a cochlear model including inner hair cell neural firing patterns. A different set of modeling approaches are evaluating for the linear filtering data, including differences in spectral shape, differences in the mel cepstral coefficients, and changes to the spectrum center of gravity and bandwidth. All of the modeling approaches incorporate changes to the auditory periphery caused by the hearing loss. The best model for the nonlinear and the best model for the linear data are then combined to model the combined data. The modeling results show correlation coefficients greater than 0.94 for modeling the noise and nonlinear processing effects alone, greater than 0.85 for modeling the linear filtering effects alone, and greater than 0.95 for modeling the combined effects of nonlinear and linear processing. The models are equally accurate for the normal-hearing and hearing-impaired listeners.

B16

Human Loudness Scaling: Arithmetic or Geometrical Loudness Intervals?

Mead Killion, Edgar Villchur, Mary Meskan, Brian Glasberg, Jeremy Marozeau, Mary Florentine

When subjects are given neutral instructions to trisect the loudness range between two 1 kHz reference tones presented at 40 and 80 dB SPL (nominally 1 and 16 sones), subjects choose levels of approximately 66 and 74 dB SPL, nominally 6 and 11 sones, giving intervals consistent with arithmetic loudness scaling. Further trisections between a) the original 40 SPL tone and the upper choice at approximately 74 dB SPL, and b) between 80 dB SPL and the lower choice at approximately 66 dB SPL produce similar scaling. In contrast, three Chicago Symphony Orchestra musicians were asked to produce tones corresponding to musical loudness notations between ppp (extremely quiet) and fff (extremely loud). These subjects increased the SPL nearly the same amount between each notation (as did one of the authors as singer), consistent with geometric loudness scaling. Over the musical notations between pppp to fff, the bass trombone player deviated an average of less than 0.6 dB SPL at any step from an average increase of 5.7 dB SPL (approximately 1.5x loudness) per step. The phon equivalent of each tone was calculated using Stevens Mark VII and the loudness model of Glassberg and Moore, and was measured directly using a group of subjects who adjusted the level of a 1 kHz tone to match the loudness of the musical tone. The improved reliability of the trisection method, which gave a standard deviation of approximately 1 dB in the trisection judgments, suggests the possibility of adjusting the compression characteristics of hearing aids on an individual basis, something that has been impractical with previous methods, that have reported standard deviations of 6-7 dB.

B17

Research of the Acoustic Feedback Phenomenon and Placement of the Microphone for Implantable Middle Ear Hearing Devices

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Recently, several types of the implantable middle ear hearing devices (IMHEDs) have been developed for hearing impaired person with a severe hearing loss. As the IMHEDs directly drive the ossicular chain by changing acoustic sound into a mechanical vibration, IMHEDs are able to produce a high quality sound.

The IMEHDs are divided into three parts which are an implantable microphone, a signal processing unit, and vibration transducer. Generally, implantable microphone has been implanted in the temporal bone because of the easy surgical procedures for implantation. However, since the microphone is implanted in the temporal bone, a microphone membrane can be damaged by external impact and the biological noise caused by the external contact can be produced. In order to overcome the problems, the placement of the implanted microphone should be changed. As an alternative, the microphone can be implanted in the auditory canal.

As the microphone is implanted in the auditory canal, the microphone input sound can be increased by the resonance characteristic of the auditory canal and the horn collecting sound effect of the pinna. However, since the microphone is implanted in the auditory canal, the howling can be produced by the acoustic feedback.

In this paper, an amount of the acoustic feedback is measured as the changing the placement of microphone in the auditory canal using the physical model which is similar to anatomical properties of the human ear. Then, the proper placement of the microphone is determined on the basis of the measured amount to minimize the howling effect by the feedback cancellation. To verify reduction of the acoustic feedback in the determined position, an amount of the acoustic feedback by the feedback cancellation is measured. It reveals that the howling effect by the acoustic feedback is decreased.

Therefore, it shows that the microphone input sound can be increased by the characteristic of the auditory canal as implanting the microphone in the auditory canal and the howling effect using feedback cancellation can be also decreased.

B18

The Effect of Linear Frequency Transposition on Speech Identification in Adults

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Individuals with steeply sloping high frequency hearing loss may not benefit from conventional amplification. The limitations can be technical, or, as has been recently acknowledged, if the hearing loss is a consequence of a complete depletion of hair cells, the acoustic stimulation of these “dead regions” does not improve performance or may even negatively affect speech understanding (Ching et al., 1997; Turner & Cummings, 1999; Moore, 2004). Several methods based on delivering the higher frequency region as a lower frequency substitute have been proposed as a means to restore the audibility of the high frequency sounds which are either unaidable or unreachable. The current study examines the efficacy of a commercially available frequency lowering algorithm based on linear frequency transposition.

Fourteen test subjects with steeply sloping hearing loss were fitted with an open fit hearing instrument. Their speech identification performance in quiet with and without frequency transposition processing was tested in three separate visits at 1) the initial fitting; 2) two weeks after the adapted to the master program and frequency transposition program; 3) one month after they were trained on voiceless consonant identification for two weeks.

Subjects showed an average improvement of 10-15% when frequency transposition was used in the consonant identification task during the last visit. The improvement was more pronounced at lower presentation levels (55dB vs. 68dB SPL). The greatest improvement in performance with frequency transposition was observed two weeks after the initial fitting, during which the subjects were instructed to listen through both the master program and the frequency transposition program in all of their listening environments. The directed training used in the current study had a minimal impact on overall scores, but some individuals benefited greatly from its use. The comprehensive phoneme level error analysis showed that identification of the phonemes with a broader spectrum improved first, even without training or experience with frequency transposition processing. Initially, identification of phonemes with predominantly high frequency cues was negatively affected . However, identification of these phonemes improved the most when training was provided.

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B19

Perception of Time Compressed Speech in New Hearing Aid Wearers

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Aging adults with sensorineural hearing loss have shown difficulties in understanding rapid (time-compressed) speech in previous studies (Gordon-Salant & FitzGibbons, 1993; Gordon-Salant & FitzGibbons, 2001). These difficulties have been attributed to reduction in abilities of elderly listeners to process duration of brief acoustic information in time-compressed speech (Dubno et. al., 1987; Gordon-Salant & FitzGibbons, 2001; Turner et. al., 1997). To date, no studies have investigated possible effects of amplification in elderly hearing aid wearers on their abilities to process time-compressed speech.

The purpose of the current study was to compare the Time Compressed Speech (TCS) scores of new elderly hearing aid wearers under unaided and aided conditions. Recordings of TCS stimuli with 0%, 30%, and 60% time compression were presented in sound field at 62 dB SPL under unaided and aided conditions. For the aided conditions, subjects were fit with hearing aids from two different manufacturers programmed to first-fit algorithms and TCS testing was conducted at initial fitting and during a one month follow-up evaluation to explore possible acclimatization effects. Preliminary results indicate significant changes in TCS scores across unaided and aided conditions during initial fitting and follow-up testing phases.

B20

Protocol for the Evaluation of Auditory Functions for RCMP Members Wearing Hearing Aids

Chantal Laroche, Christian Giguère and Véronique Vaillancourt, University of Ottawa, Dr. Marc-André Beaulieu and Dr. Jean-Pierre Legault, Royal Canadian Mounted Police

While the audiogram is a highly useful clinical diagnostic tool, its relationship to functional hearing ability is limited. Like many law-enforcing agencies, the Royal Canadian Mounted Police (RCMP) currently uses the audiogram to classify the hearing of its officers into 5 main categories (H1-H5). While these criteria have been successfully used in the RCMP for several years, the stimuli and tasks inherent in audiometric testing can be quite different than the listening activities that take place in the workplace. Such activities typically involve sound detection, recognition, localization and speech perception, occur in the presence of background noise, and are performed under binaural listening conditions.

To assist the RCMP in making more informed decisions regarding fitness to work in officers wearing hearing aids, a testing protocol has been proposed and successfully administered to RCMP members. The protocol includes unaided and aided soundfield measures of sound detection, speech perception and sound localization, in addition to standard audiologic evaluations. It is used to: 1) evaluate the auditory functions for individual RCMP members currently facing operational restrictions because they do not meet the hearing criteria set forth in the RCMP Hearing Policy and could therefore compromise the safety of others as well as their own, and 2) verify if hearing aids allow these members to carry out the necessary auditory functions required to safely perform their job.

Individual results of the functional hearing evaluation protocol help the medical team at RCMP in making more informed decisions about the operational suitability of each member. A secondary objective is to use the overall results across all tested members, together with the complete description of hearing aid parameters used, to form a database that will hopefully help identify best practices in hearing aid fitting for optimal functional hearing abilities in the RCMP work environment. To date, approximately 50 members have been tested with the protocol. Preliminary data suggest that, generally, hearing aids yield greater benefits for speech recognition in quiet than in noise. Hearing aids are found to hinder sound localization of front/back sources in the horizontal plane in some cases, without significantly affecting localization in the left/right plane. The detailed methodology, together with examples of depersonalized results and a preliminary analysis of the data will be presented. [This work was supported by RCMP].

B21

Syllabic Compression versus Sophisticated Automatic Volume Control (AVC)
The Winner Is…

Matthias Latzel, Kirsten Wagener, Matthias Vormann, Katrin Glass, Siemens Audiological Engineering Group, Erlangen, Hörzentrum, Oldenburg, University of Applied Science, Oldenburg

In the hearing aid business there is almost no questioning whether wide dynamic range compression (WDRC) is the right tool to compensate for the loss of normal loudness perception which often accompanies the sensorineural hearing loss. Consequently WDRC is well established in the listening device industry as one of the most utilized tool in modern hearing instruments. However, the choice of correct settings of the compression system's parameters is still an open question, and as such is being addressed in a large numbers of publications. The majority of these publications offer fitting rules for calculation of the target gain, compression ratio, number of channels and/or sometimes the compression knee points while staying clear from the issue of time constants of the compression systems. Since a WDRC system compensates only for the loss of normal loudness perception while the damage of temporal aspects of a sensorineural hearing loss is not addressed, we suggest that the right choice of the time constant of the compression system could be a first step towards investigation of the time domain.

This poster describes two studies where a state-of-the-art syllabic compression is compared to a rather slow compression with long time constants (lasting several seconds like an AVC). In contrast to the former study, the compression systems were integrated in physically similar housings to ensure equal electro-acoustic conditions and to avoid biasing judgments due to cosmetic aspects.

Overall twenty-eight subjects with sensorineural hearing loss and reduced dynamic range took part in both studies. Both subjective and objective speech tests were used in the study. Further more, tests comprising a special "dynamic" loudness scaling method and absolute and relative judgments of several processed sound samples were conducted. In all cases, the compression systems were evaluated for typical level ranges relevant in real life listening situations.

In the second study, additional tests were performed under real life conditions in which the subject used and judged the test hearing aids in their individual environments.

The investigations show no clear winner for one of the compression schemes under investigation. In some situations, the slow time constants provide a more comfortable loudness perception, especially in very loud environments. On the other hand, the syllabic compression system showed advantages for speech perception in situation with competing noise.

The results lead to the conclusion that the time constants for compression systems should not be chosen ad hoc during the fitting process but should be adjusted by the hearing system "online".

B22

Amplitude-Level Functions For Mixed Modulated ASSRs In Noise

Elizabeth Leigh-Paffenroth, Owen Murnane, and Richard Wilson, Mountain Home VAMC

Listeners with sensorineural hearing loss uniformly complain about their difficulty understanding speech in noise. Speech perception deficits can occur at different levels of the auditory system (Eggermont, 1994; Krishnan, 2002; Rhode, 1994; Steinhauer, 2003) it is not possible, however, to determine at which level(s) these deficits exist using behavioral measures of speech perception. Electrophysiologic correlates of speech perception in humans have been measured with auditory evoked potentials (AEPs) using stimuli that mimic the temporal and spectral complexities of speech (e.g., Dimitrijevic et al., 2001, 2004; Tremblay et al., 2002, 2004). The auditory steady state response (ASSR) is a type of AEP evoked by modulations in amplitude and/or frequency of pure tones and is typically used as an objective estimate of behavioral pure tone thresholds in difficult-to-test patients (Picton et al., 2003). Recent experiments have shown significant positive correlations between the number of suprathreshold ASSRs and word recognition scores in quiet and in noise for listeners with normal hearing and for listeners with sensorineural hearing loss (Dimitrijevic et al., 2004; Dimitrijevic et al., 2001).

The purpose of this study was (1) to measure auditory steady state responses (ASSRs) in quiet and in the presence of multitalker babble, (2) to measure word recognition in quiet and in the presence of multitalker babble, and (3) to compare ASSRs to word recognition for listeners with and without hearing loss. The construction of the electrophysiologic test was an attempt to mimic behavioral word recognition tasks currently used to assess speech perception performance. Mixed modulated (MM) ASSRs for speech-shaped stimuli were recorded in 24 listeners with normal hearing and in 24 listeners with sensorineural hearing loss for modulation rates of ~ 40 Hz and ~ 90 Hz. ASSRs were elicited by a complex signal constructed of 100% amplitude-modulated and 20% frequency-modulated pure tones with carrier frequencies of 500, 1500, 2500, and 4000 Hz. The amplitude of each frequency component was calibrated to match the long term speech spectrum average. MM ASSRs were recorded in quiet and in multitalker babble at nine signal-to-babble ratios (S/B) from -16 to 16 dB. The amplitude-level functions of the MM ASSRs were compared to performance on the words-in-noise protocol (Wilson, 2003) for each listener. The relations among ASSR amplitude, number of detected ASSRs, word recognition performance in quiet, and word recognition performance in noise will be discussed.

B23

Binaural Interference in Listeners with Sensorineural Hearing Loss

Elizabeth Leigh-Paffenroth, Ph.D., Christina M. Roup, Ph.D., and Colleen M. Noe, Ph.D., Mountain Home VAMC, The Ohio State University

The advantages of binaural hearing are well-established in the literature (Akeroyd, 2006). Persons with bilateral hearing loss, therefore, typically are fit with two hearing aids (Carter et al., 2001; Erdman & Sedge, 1981); the majority of the literature supports binaural amplification for individuals with bilateral sensorineural hearing loss (e.g., Brooks & Bulmer, 1981; Kobler et al., 2001). Despite the success of binaural hearing aid fittings, however, several studies indicate that 20-30% of individuals with binaural hearing loss choose to wear one hearing aid (e.g., Chung & Stephens, 1986; Kobler et al, 2001). One explanation for the rejection of binaural amplification is binaural interference, which occurs when the signal presented to one ear interferes with the perception of the signal presented to the better ear (Allen et al., 2000; Jerger et al., 1993; Silman, 1995). The purpose of the study was to determine the extent to which people who have difficulty understanding speech in one ear also have difficulty processing information binaurally.

Binaural processing was measured in 30 listeners (aged 18-80 years) with symmetrical high frequency sensorineural hearing loss. Monaural and binaural middle latency responses (MLRs) were compared to behavioral performance on the Dichotic Digits Test (DDT), the Word recognition In Noise (WIN) test, and masking level differences (MLDs). Preliminary results revealed significant relationships between the amplitude of the MLR and measures of behavioral performance. The DDT revealed expected right ear advantages for most listeners with a few listeners showing a left ear advantage. In addition, the MLD revealed listeners who performed poorly in the SπNo condition. Relationships among monaural and binaural auditory tasks and the potential impact on audiologic rehabilitation will be discussed. Individuals with symmetrical hearing loss, who find binaural amplification problematic, may have binaural interference that would lead to rejection of amplification. Discussion includes listeners who prefer monaural amplification and listeners who have asymmetrical word recognition in noise performance.

B24

Physiological Assessment of Nonlinear Hearing Aid Amplification Schemes

Benedict Leung, Ian Bruce, McMaster University

Nonlinear amplification schemes for hearing aids have been developed primarily to deal with the problem of loudness recruitment. The most commonly used form of nonlinear amplification is wide-dynamic-range compression (WDRC). Unfortunately, finding WDRC characteristics that satisfactorily deal with loudness recruitment while maintaining good speech intelligibility has proven difficult. An alternative nonlinear scheme, Advanced Dynamic Range Optimization (ADRO), has been shown in several studies to provide better speech intelligibility and listening comfort than fast-acting WDRC. ADRO uses a set of fuzzy-logic rules to make gain changes to optimize audibility, comfort, protection against loud sound, and noise attenuation. The "hearing protection" gain rule acts instantaneously, whereas the audibility and comfort rules adjust the gain slowly, such that ADRO provides linear amplification most of the time.

The goal of this study is to examine the physiological basis for the relative performance of linear amplification, WDRC, and ADRO. Sentences from the TIMIT Speech Database were processed by each algorithm. In the case of WDRC, both single-channel and multi-channel schemes with fast and slow dynamics were tested. Speech signals were presented at 52, 62, 74, and 82 dB SPL (sound pressure level) with various noise levels and types, to simulate real-life environments. The simulations first use an auditory-periphery model to generate a "neurogram" of the auditory nerve's representation of the test speech material. The spectral and temporal modulations in the neurogram are then analyzed by a model of cortical speech processing. The effects of the background noise, the presentation level, the hearing loss and the amplification scheme are evaluated by comparing the cortical model response for a given condition (the "test" response) to the cortical model response to the same TIMIT sentence presented in quiet at 65 dB SPL to the normal-hearing model (the "template" response). From the difference between the test and template responses, a spectro-temporal modulation index (STMI) value is calculated. High STMI values predict good speech intelligibility, while low values predict poor intelligibility. Preliminary results show that ADRO is better at restoring the neural representation of speech than the other algorithms tested, even when the WDRC algorithms utilize slow time constants. In the case of no background noise, all the algorithms perform similarly well. However, when background noise is added, STMI values for higher SPLs drop notably for all the algorithms except for ADRO, which sustains a stable value throughout the range of SPLs tested.

[The authors thank Dynamic Hearing for providing a software simulation of the ADRO algorithm.]

B25

Patient and Spousal Expectations of Hearing-Aid Utilization

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The success of a hearing-aid fitting is impacted by factors such as patient pre-use expectations regarding hearing-aid outcome. Since the majority of first-time hearing-aid users report seeking treatment because of their spouse, it seems logical to assume that the spouse's pre-use expectations about hearing aids also might have an impact on hearing-aid outcome. At this point in time, no information is available regarding the spouse's pre-use expectations of hearing aids. The purpose of the present investigation was to elicit patient and spousal information regarding their pre-use expectations about hearing aids.

Twenty-five male veteran patients with bilateral mild to severe sensorineural hearing loss and their non hearing-impaired female partners participated in this investigation. These subjects underwent an interview, where they were asked open-ended questions regarding their pre-use expectations of hearing aids. Specifically, patients and their spouses answered questions about their current problems related to the hearing loss and how they thought the hearing aids would affect their day-to-day life, their communication, and their relationship. They were also asked if they had any concerns about the hearing aids. The interview sessions were conducted with the patients and the spouses separated and were completed prior to the veteran getting hearing aids through the Audiology & Speech-Language Pathology Services at the Portland VA Medical Center. These interview sessions were audio-tape recorded and later transcribed into a Microsoft Word document. Responses collected during the interviews were analyzed with qualitative methods derived from grounded theory to identify themes associated with patient and spousal expectations of hearing aids. The themes that emerged from these interviews included difficulty with communication and with environmental-sound awareness, irritation and frustration, restriction of social life or isolation, and a negative impact on relationships. Many patients and spouses expected improvements in these domains with the use of hearing aids; however, the patients and spouses also noted significant concerns about the physical properties of the hearing aids and adjustment to the hearing devices. The information obtained during these interviews revealed that spouses had clear expectations about their partner's hearing-aid utilization. The spouse's perspective regarding the rehabilitation process should be explored further as it may have a substantial impact on the success of the treatment.

[This work was supported by the VA Rehabilitation Research & Development Service]

B26

Improved Two-Stage Binaural Speech Enhancement Based On Accurate Interference Estimation for Hearing Aids

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Speech enhancement is one of the most crucial functions, if not the most, in hearing aids, as hearing-impaired people have great difficulty in understanding speech in noisy environments. To improving the communication ability of hearing-impaired persons via voice, we previously presented a two-stage binaural speech enhancement (TS-BASE) approach to be used in hearing aids, which consists of interference estimation by pre-trained adaptive filters and speech enhancement using the Wiener filters (J. Li et. al., “A two-stage binaural speech enhancement approach for hearing aids with preserving binaural benefits in noisy environments,” in Proc. Forum Acousticum 2008).

Though its effectiveness has been confirmed in single- and multiple-noise-source conditions, our previously proposed TS-BASE approach still suffers from one main problem that the interference estimates by adaptive filters are different from the interference components embedded in the input signals. In this paper, we attempt to give a much more accurate interference estimation approach by minimizing the mean square error between the output signals at adaptive filters (that was regarded as interference estimate in the previous algorithm) and the input signals. This minimization procedure is implemented with the normalized least-mean square (NLMS) adaptive algorithm. The benefits of the improved TS-BASE algorithm from this accurate interference estimate are investigated through comprehensive experiments in various acoustic environments. It is expected that the improved TS-BASE algorithm can give much higher interference suppression performance. Moreover, like the previous TS-BASE algorithm, this improved TS-BASE algorithm yields the binaural output signals. At the binaural outputs, the binaural cues are preserved, which are able to provide the additional benefit in understanding speech resulting from the selective binaural hearing and provide the ability in localizing sound source, i.e., keeping the auditory scene of acoustic environments.

B27

Focused On Hearing Impaired (HI) Speech Perception and Speech Enhancing Algorithms for Hearing Aids

Feipeng Li, University of Illinois at Urbana-Champaign

My PhD thesis is focused on Hearing Impaired (HI) speech perception and speech enhancing algorithms for hearing aids. Our speech perception studies reveal that HI listeners have difficulty understanding noisy speech because of certain component-sounds (features) which they can not hear. As an example, we have been working with a HI listener who has a problem discriminating /ka/ and /ga/. These two sounds share a small mid-frequency region in the speech spectrum. NAR-L compensation improves the average percent correctness by 10%, but it has no effect on /k/ and /g/ scores. In order to explain why the subject can not hear certain sounds, audiometer and TEN test [Moore2000] are used to diagnose the HI listener's hearing threshold and cochlear dead regions.

Based on a large amount of speech perception data, it is hypothesized that initial consonant speech sounds are encoded by across-frequency temporal onsets. One speech sound may be more robust (more tolerant to noise) than another because it has a more intense acoustic onset. We hypothesize that our HI listener cannot detect certain onsets because the perceptual cues are missing, due to the hearing loss. The problem is one of identifying the features the subject cannot hear.

Systematic psychophysical methods have been developed to determine the features quantitatively. To measure the time-frequency importance function for consonant sounds, speech stimuli are high and lowpass filtered, or time-truncated, before being presented to Normal Hearing (NH) listeners. Databases of NH and HI speech perception under various SNR conditions are constructed to investigate the effect of noise on speech recognition [Phatak&Allen2007, Yoon&Allen@IHCON2006]. A visualization tool that simulates the auditory peripheral processing, called the AI-gram

[Bryce&Allen@ICSLP2006], is used for the observations of the speech events under various SNR conditions [Regnier&Allen2008]. Using software developed for the purpose of speech modification, we can convert most initial consonant sounds into another sound, based on our knowledge of speech features. Speech enhancement for HI listeners is our long-term goal.

*All referenced may be found at
http://hear.ai.uiuc.edu/wiki/Main/Publications

B28

Redundant Speech Information in the Temporal Envelopes of Neighbouring 1/3-Octave Bands

Adrian M. Lister, B.Eng., Lorienne M. Jenstad, PhD, University of British Columbia

We are quantifying the redundant speech information contained in the temporal envelopes of neighbouring third octave bands.

Perceptual weighting of temporal information in discrete 1/3-octave bands will be quantified by correlating acoustic changes in the speech signal (i.e., number of bands of temporal information and their relative spacing) with an individual's performance on a nonsense syllable task - the University of Western Ontario Distinctive Features Differences test (UWO-DFD; Cheesman & Jamieson, 1996, adapted from Feeney & Franks, 1982).

Stimuli were created by passing the 21 DFD nonsense words through a third-octave IIR filter-bank providing 21 bands. Envelope extraction of these bands was performed by low-pass filtering and full wave rectification. The resulting envelopes were used to modulate white noise. Finally, the modulated noise was frequency limited by filtering with the same band pass filter used in the original analysis band.
Ten listeners with normal hearing are being recruited for the behavioural study. Participants hear the stimuli presented via headphones at an overall level of 70 dB SPL. All 21 possible responses are displayed on a computer monitor and participants are asked to identify which of the words they heard.

Previous research has typically examined the spectral nature of the speech signal. Less attention has been given to the temporal domain. Previous work (e.g., Hedrick & Jesteadt, 1996; Apoux & Bacon, 2004; Shannon, 1995), has examined the importance of the temporal envelope on speech intelligibility scores in situations where the speech signal has been degraded spectrally, with the results indicating that normal and hearing-impaired listeners can use the temporal envelope in order to maintain high intelligibility scores. One limitation of these studies, however, has been that they've used wide frequency bands; e.g., octave or wider, when examining the contribution of the envelope to speech perception

We desire to set the basis for the development of a temporal weighting function - the amount of perceptual weight an individual places on 1/3-octave bands of the temporal envelope for speech.

Crouzet and Ainsworth (2001), however, suggest that temporal envelopes extracted from distant frequency regions are partially correlated, whereas adjacent bands exhibit strong across-channel envelope correlations. Because of these across-channel correlations it is difficult to isolate the unique information contained within each band.

We therefore need to quantify the redundant speech information between neighbouring third octave bands' temporal information prior to assessing perceptual weighting values. Preliminary data of 1/3-octave band temporal redundancy will be presented.

B29

Replication of Experiments on the Ability to Benefit from Temporal Fine-Structure Information in Speech among Persons with Moderate Cochlear Hearing Loss

Thomas Lunner, Kathryn Hopkins, Brian CJ Moore, Oticon Research Centre 'Eriksholm', Denmark, Department of Experimental Psychology, University of Cambridge, United Kingdom

Recent studies have shown that hearing-impaired test subjects are less able to benefit from temporal fine structure (TFS) information than normal-hearing subjects. Hopkins, Moore and Stone (2008) showed that, if part of the spectrum is noise- or tone vocoded above a 'cut off channel' CO) to remove TFS information, most hearing-impaired subjects benefited less than normal-hearing subjects from the additional TFS information that was available as CO increased. The reduced ability to take advantage of TFS information in speech may partially explain why subjects with cochlear hearing loss get less benefit from listening in a fluctuating background than normal-hearing subjects. TFS information may be important in identifying the temporal "dips" in such a background. However, there were individual differences in the ability to utilise TFS.

In another study Hopkins and Moore (2007) measured the ability of normal-hearing and hearing-impaired subjects to access TFS information in complex tones. The results suggest that normal-hearing subjects can use TFS information provided the temporal spacing between fine-structure peaks is not too small relative to the envelope period, but subjects with moderate cochlear hearing loss make little use of TFS information for unresolved harmonic components. These TFS results may have interesting clinical implications if the ability to utilise TFS varies from person to person. This may indicate that hearing aid signal processing should differ depending on sensitivity to TFS.

Here we present perceptual experiments to confirm the findings above. The same test procedure and tone-vocoding scheme were used as in the Hopkins, Moore and Stone (2008) study. However, to test the applicability of the findings in new contexts, the outcome measures were different with regard to language and speech-in-noise test and speech-in-noise assessment procedure. Furthermore, a simple TFS-test developed at the University of Cambridge, similar to the one used in Hopkins and Moore (2007), was used to predict TFS ability. Twenty hearing-impaired test subjects with moderate hearing loss of cochlear origin were recruited, as well as 10 normal-hearing test subjects. Results and implications will be discussed.

B30

Development of Working Memory Capacity, Phonological and Reading Skills in Children with Cochlear Implantation

Björn Lyxell, Birgitta Sahlén, Malin Wass, Tina Ibertsson, Birgitta Larsby, Mathias Hällgren & Elina Mäki-Torkko, Linköping and Lund universities, Sweden

We will present an overview of a set of cross-sectional and longitudinal studies conducted in our laboratory with the purpose to examine cognitive and communicative development in children with cochlear implants (CI). The results demonstrate that children with CIs perform at significantly lower levels on a majority of the cognitive tasks. The exceptions to this trend are performance levels on cognitive tasks with relatively low demands on phonological processing (e.g., the visual parts of working memory). The results also demonstrate a relationship between expressive speech skills and phonological skills. A fairly high proportion of the children can reach a level of reading comprehension that is comparable age-matched hearing children, despite the fact that they have relatively poor phonological decoding skills. General working memory capacity is further correlated with the type of questions asked in a referential communication task. Factors related to the implant (e.g., time with CI, uni- or bilateral CI) had only little impact on cognitive and communicative development. The results are discussed with respect to issues related to education and rehabilitation.

B31

Effect of Amplification on the Intelligibility of Speech in Hearing Impaired Children With and Without Dead Regions In The Cochlea

Alicja Malicka, Kevin J. Munro and Thomas Baer, University of Manchester, University of Cambridge

Adults with high-frequency (HF) sensorineural hearing impairment with and without dead regions (DRs) in the cochlea differ in benefit from amplification of speech presented in quiet [Vickers et al., J. Acoust. Soc. Am. 110, 1164-1175 (2001)] and in background noise [Baer et al., J. Acoust. Soc. Am. 112, 1133-1144 (2002)]. Subjects with HFDRs showed no improvement in speech intelligibility when spectral components of the speech above about 1.7 times the edge frequency of the DR were amplified according to a hearing-aid-prescription formula while performance of those without DRs showed improvement with addition of amplified frequency components up to 7.5 kHz. In the present study we tested a group of six children (8-12 years old) who were experienced hearing aid users with moderate-to-severe sensorineural hearing impairment. The presence of DRs was diagnosed using the TEN test and 'fast' psychophysical tuning curves. Four children show evidence of DRs (two unilateral and two bilateral). The vowel-consonant-vowel stimuli (65-dB SPL) were subjected to the frequency-gain characteristic prescribed by the DSL prescription formula then low-pass filtered with various cutoff frequencies and presented via headphones. The speech material was presented in quiet and in the presence of background noise. The results showed that in ears with or without DRs the performance improved with increasing cutoff frequency up to 7.5 kHz. The pattern of results was similar in both quiet and background noise.

B32

Analysis and Control of Statistical Fluctuations in Noise Reduction Systems

Rainer Martin and Colin Breithaupt, Institute of Communication Acoustics, Ruhr-Universität Bochum

The design of noise reduction algorithms based on short-time spectral analysis has to strike a careful balance between the desired temporal and spectral resolution and the variance of estimated spectral quantities. While the trade-off between temporal and spectral resolution has been subject of many investigation, the influence and the control of the estimation error variance is less well understood. It turns out, however, that the variance of estimated quantities limits the performance of noise reduction algorithms. Most notably, the fluctuations in the residual noise, also known as 'musical noise', are frequently counteracted by applying less noise reduction to the noisy signal with the aim of covering annoying fluctuations, or, by smoothing in time and/or in the frequency dimension which may introduce distortions to the speech signal.

In this contribution we present an analysis of the spectral outlier statistics by means of logarithmic histograms and an approach for noise reduction in hearing aids which reduces random fluctuations and leads to an improved quality of the processed signal, especially in speaker babble noise. The algorithm makes use of parametric signal and error models and a novel smoothing algorithm in the cepstro-temporal domain. While the signal model accounts for the super-Gaussian distribution of short-time spectral amplitudes of speech signals, the error model provides means for applying a compressive function on the spectral amplitudes of speech. It thus increases the noise reduction and controls the outlier statistics of the processed signal. The latter can be further improved by using the cepstro-temporal smoothing process which reduces random fluctuations in the spectral domain while preserving a high frequency resolution and the temporal dynamics of the speech components. We explain the basic principles of both the parametric spectral estimation and the cepstro-temporal smoothing procedures and demonstrate their performance with speech samples and various noise types.

B33

Modeling the Contributions of Harmonic Amplitude and Phase Contrast to Improvement in Vowel Recognition by Hearing-Impaired Listeners

Michelle Molis, Anna Diedesch, Marjorie Leek, Frederick Gallun, National Center for Rehabilitative Auditory Research, Portland VA Medical Center

Vowel recognition is strongly dependent on the frequencies of the lowest two or three spectral peaks or formants. Because of the abnormally broadened auditory filters typically observed in hearing-impaired listeners, the internal representation of amplitude contrast between formant and non-formant frequencies may be altered for these listeners, resulting in impaired vowel recognition. Previous research has shown that hearing-impaired listeners typically require up to three times more spectral contrast than normal-hearing listeners to differentiate between vowel-like sounds in which formant location is indicated by increased amplitude of pairs of harmonics at formant frequencies relative to background harmonics. Increasing amplitude contrast in "formant" regions can improve vowel identification for highly stylized synthetic vowels. Similarly, manipulating the phases of the formant harmonics also enables formant perception, thereby allowing accurate stimulus identification. Listeners are able to correctly identify vowel-like stimuli with a flat amplitude spectrum when formant location is indicated solely through manipulation of harmonic phase.

In this study, five normal-hearing and five hearing-impaired listeners identified a set of three vowel-like sounds that had formants coded by coherent, but sub-threshold, increments in formant amplitude and phase. Recognition of these sounds was improved for nearly all listeners for the combined contrast dimensions over recognition with either amplitude or phase alone. The Auditory Image Model will be used to model the effects of sensitivity loss, broader-than-normal auditory filters, and possible loss of phase locking on predictions of internally preserved formant structure for the set of vowel-like sounds. An estimate of threshold internal contrast will be generated and used to predict identification performance by hearing-impaired listeners. [Work supported by NIH].

Posters for Session C should be put up by 8 A.M. Saturday, August 16, and taken down after 10 P.M. Saturday, August 16 or before 7 A.M. Sunday, August 17. Presenters should be at their posters from 9:45 – 11:00 A.M.; 4:30 - 5:00 P.M.

POSTER SESSION C

Saturday 8:00AM to 10:00 PM

C1

Frequency-Domain Feedback Cancellation in Hearing Aids

Ramesh K. Muralimanohar, James M. Kates, University of Colorado at Boulder, GN ReSound

Feedback cancellation is an important component of hearing-aid signal processing. Most implementations of feedback cancellation use a time-domain adaptive filter to continuously model the feedback path. The output of the model is an estimate of the feedback that is subtracted from the input. An ideal model would give perfect cancellation of the feedback signal at the input and would provide stability for any amount of gain. In practice, the amount of additional gain is limited to about 10 to 15 dB, and the feedback cancellation algorithms tend to cancel sinusoidal inputs and are unable to respond rapidly to large changes in the feedback path. Frequency-domain feedback cancellation algorithms offer the potential advantage of independent adaptation in each of the Fourier transform frequency bands. The frequency domain adaptation should provide faster convergence for large changes in the feedback path and should produce fewer processing artifacts than a comparable time-domain algorithm. In this paper, two frequency-domain feedback cancellation algorithms are implemented using a real-time digital prototyping system based on the Texas Instruments TMS320C5510 DSP. The input to the real-time system is the microphone and the output will be used to drive the receiver of a behind-the-ear hearing aid mounted on a dummy head. The performance of the frequency-domain system will be compared to that of a time domain system for a variety of input signals including, white noise, speech, and pure tones, and for a variety of dynamically-changing feedback paths.

C2

Perceptual Correlates of the Long-Term SNR Change Caused By Fast-Acting Compression

Graham Naylor, Filip Munch Rønne and René Burmand Johannesson, Oticon Research Centre 'Eriksholm', Denmark

At IHCON 2006 we presented systematic measurements demonstrating that the long-term SNR at the output ('Output SNR') of an amplitude compression system generally differs from the SNR at the input ('Input SNR'), and that the difference can be several dB in either direction. The SNR change is affected by various parameters of the compression system and the overall levels and modulation properties of the Signal and Noise components, in ways which are fairly straightforward to explain. The question remains, whether these objective effects can be meaningfully linked to perceptual phenomena. A compression system may affect the long-term Output SNR, but it cannot affect the instantaneous SNR, so it is not certain that for example a higher long-term SNR at the output due to compression will yield higher speech intelligibility. It is conceivable that side-effects of compression, such as temporal distortion of envelopes, may outweigh an apparent SNR improvement.

Here we present perceptual experiments designed to provide a preliminary answer to the above question of whether long-term SNR changes after compression correlate with perceptual results. We focus on speech intelligibility as the perceptual quantity to study first. By manipulating compression parameters and the modulation characteristics of an interfering noise, as well as by setting Input SNR conditions individually for each listener, we create a set of experimental conditions in which Input SNR and Output SNR differ by controlled amounts. Our findings indicate that in the majority of cases, the direction of change in speech intelligibility between the mixture at the input and at the output of a compression system is the same as the direction of change in long-term SNR. This suggests that change in long-term SNR through the system is a meaningful indicator of perceptual benefit of that system. In addition, a given SNR change generated by a compressor seems to have a smaller perceptual effect than the same SNR change, when produced simply by changing the level of Signal or Noise in a linear system. This suggests that there are phenomena other than change of long-term SNR which also have perceptual effects.

C3

Relations between Hearing Loss and Cognitive Abilities As Well As Spatial Release from Speech-On-Speech Masking In Aided Hearing-Impaired Listeners

Tobias Neher & Thomas Behrens
Eriksholm Research Centre, Oticon A/S, Kongevejen 243, 3070 Snekkersten, Denmark

This poster supplements the presentation by Behrens et al. dealing with a field test into the spatial hearing abilities of 21 experienced bilateral hearing-aid users. In particular, it focuses on two potential predictors of performance in cocktail party-like situations involving multiple competing talkers: hearing loss and cognitive abilities. Performance in such listening situations was quantified with the help of a paradigm that allowed measuring spatial release from speech-on-speech masking along both the left-right and the front-back dimension. The effects of hearing loss were considered by determining average hearing threshold levels in frequency regions known to contribute differentially to left-right and front-back perception. The effects of cognitive abilities were considered by determining subject performance on tests of working memory and attentional capacities.

The analyses showed that performance along the left-right dimension was strongly affected by the degree of hearing loss in the frequency region where interaural time differences are known to dominate spatial auditory perception. Furthermore, the left-right condition seemed to engage the subjects' cognitive capacities much more than the other (less complex) conditions that were tested. These findings seem to suggest that access to interaural temporal fine structure information as well as the ability to "enhance" a target source by means of top-down processing are crucial for the functioning of hearing-aid users in complex multi-talker listening environments.

C4

Individual Variability for Listeners' Speech Recognition and Satisfaction in Noise

Peggy Nelson, University of Minnesota
Benjamin Hornsby, Vanderbilt University

Research reports have long noted that there are significant individual performance differences among listeners with sensorineural hearing loss, even for pairs of listeners with similar audiograms. We wish to describe to what extent individual differences in susceptibility to masking and masking release (using modulated maskers) affect listeners' success with amplification. In this study we examine individual variability in speech recognition in steady state and gated noises in both unaided and aided conditions. The articulation index (AI) was used to predict performance in steady and gated noise and to quantify performance deficits above those due to reduced audibility alone. Finally we examined relationships between laboratory measures of speech understanding in noise and hearing aid outcome measures.

HI listeners were fit bilaterally with Starkey Destiny 1200 behind-the-ear hearing aids, matched closely to NAL-NL1 targets. Rear-ear aided response measures were obtained using speech-shaped noise (matched to the long-term spectrum of the IEEE sentences) presented at 1 meter (0° azimuth), for purposes of estimating AI. Participants used the hearing aids for 30-45 days and returned for additional testing. They were tested using IEEE sentences in quiet, in steady state noise and in gated noise (10 Hz, 50% duty cycle) at +5 and +10 dB SNR. Their acceptable noise levels (ANL, Nabelek et al., 2004) and measures of satisfaction with the hearing aids in noise are obtained at the same time. Questionnaires include the Speech, Spatial and Qualities of Hearing Scale (SSQ, Gatehouse and Noble, 2004) and the Satisfaction with Amplification in Daily Life (SADL, Cox and Alexander, 1999).

Correlations between AI estimates and listeners' performance in steady and gated noise will be reported, as well as correlations between lab measures of speech recognition in noise with satisfaction measures as reported by participants. The extent to which AI and laboratory measures can predict self-reported outcomes will aid in predicting which hearing aid users may need additional intervention, beyond a typical amplification protocol.

This work is supported by Starkey Research. We gratefully acknowledge the assistance of Bill Woods in calculating the AI estimates.

C5

Individual Differences in Hearing Aid Benefit: Addressing the Problem

Michael Nilsson, Harry Levitt, Sharon Sandridge, and Lynn Alvord, Sonic Innovations, Advanced Hearing Concepts, Cleveland Clinic, Henry Ford Medical Center

Speech recognition in noise was evaluated for two multi-channel compression hearing aids. The experimental factors investigated were: Directionality (directional vs omnidirectional input), Noise Reduction (on versus off) and Type of Aid (Aid #1 had 9 frequency channels compared to Aid #2 which had 16 frequency channels). Speech intelligibility in noise was evaluated using the Hearing In Noise Test (HINT). User satisfaction was evaluated using the Abbreviated Profile of Hearing Aid Benefit (APHAB). Forty two subjects across two sites wore the hearing aids for a period of approximately 8 months (approximately 1 month for each of the 8 combinations of noise reduction, directionality and type of hearing aid). Hearing aid benefit was defined as the difference in reception threshold for sentences (RTS) between the aided and unaided conditions using the HINT test. Results found that benefit increased with noise reduction by just less than 1 dB for both hearing aids, and by approximately 3 dB with the use of a directional input. The benefits from noise reduction and directionality were independent and additive. The two sites showed the same pattern of results. There were, however, substantial individual differences. The range of benefit for the 42 subjects was 10.9 dB with one subject showing a benefit of more than10 dB for both hearing aids (directional input plus noise reduction) and another subject showing a benefit of less than 1 dB for both hearing aids with directional input and noise reduction. The subjects' hearing levels, SRTs, monosyllabic word discrimination, unaided speech recognition in quiet and in noise were analyzed in order to identify possible predictors of these large individual differences. A multiple linear regression analysis showed the following three variables to be useful predictors of relative benefit: average hearing level in the high frequencies, speech recognition in quiet (either monosyllabic word discrimination or the HINT test in quiet) and subject's age. If these three variables are taken into account, the error variance in predicting benefit is reduced by more than 30 percent. The analysis also showed that subjects with poorer hearing in the high frequencies (and/or poorer speech recognition in quiet) showed more relative benefit with amplification than subjects with comparatively good hearing. A significant correlation was found between the HINT scores and the assessment of difficulty in everyday communication using the APHAB questionnaire. A non-parametric analysis of the APHAB data shows similar results with respect to individual differences.

C6

Effects of Nonlinear Frequency Compression on Speech Production in Children with High Frequency Hearing Loss

Melissa Polonenko, Susan Scollie, Danielle Glista, Marlene Bagatto, Richard Seewald, Marilyn Kertoy, Andreas Seelisch, Child Amplification Laboratory, National Centre for Audiology, University of Western Ontario, London, Ontario.

Nonlinear frequency compression signal processing has been proposed as a strategy for improving high frequency audibility in hearing aid fittings. This technology lowers the high frequency energy in speech, presenting it at a lower frequency for the listener. Children with hearing loss may be expected to have access to new and/or altered cues for high frequency speech recognition with such a strategy. Therefore, they may have altered speech production after a period of using frequency compression. In order to test this hypothesis, speech production samples were recorded from 10 children with moderately-severe to severe high frequency hearing loss, both before and after a trial with a prototype nonlinear frequency compression hearing instrument. The speech production samples were evaluated to determine the effects of using the nonlinear frequency compression signal processor on speech production. Spectral analysis and subjective rating results to date will be presented. [This work is supported by NSERC].

C7

Estimating Sound Pressure Levels At the Tympanic Membrane over a Wide Frequency Range

Karrie Recker, Starkey Laboratories, Inc., Tao Zhang, Starkey Laboratories, Inc., Third: Janice LoPresti, Knowles

For various applications, it is useful to know the sound pressure level (SPL) at the tympanic membrane (TM). However, measuring the SPL close to the TM is not clinically feasible, due to safety and discomfort concerns. As a result, it is desirable for clinical usage to estimate the SPL at the TM using measurements away from the TM. For frequencies below 4 kHz, the difference between measurements away from the TM and measurements at the TM is small (Stinson & Lawton, 1989). However, for higher frequencies, the difference becomes larger and more complex. This occurs because the wave length at high frequencies is comparable to or smaller than the ear canal (EC) length. Furthermore, the sound field in an individual EC becomes more complex due to differences in ear canal geometry and eardrum impedance. All of these present significant challenges for estimating the SPL at the TM using measurements away from the TM.

In this study, we tested how well real-ear measurements away from the TM can be used to estimate the SPL at the TM over a wide frequency range. To do this, we used an in-ear monitor (IEM) to present a stimulus with frequency components up to 16 kHz. The response was measured at 0 to 16 mm from the TM in 2 mm increments using a commercial probe microphone (ER-7C). We made 3 sets of measurements on 10 different ear canals, removing and reinserting the IEM and the probe tube between sets. Each subject's data were normalized to the 0 mm response. The normalized data were used to estimate the actual distance from the TM using the dominant pressure minima. Using the data from all 10 subjects, we derived correction factors for various distances from the TM. Given a measurement at a specific distance from the TM, the corresponding correction factor was calculated and used to estimate the SPL at the TM. To validate this method, we made measurements on an additional10 subjects using the same procedure. The estimated SPL at the TM was compared with the actual SPL at the TM. Preliminary data indicate good agreement between the estimated and the actual data below 12 kHz. Above 12 kHz, the discrepancy between the estimated and the actual data becomes greater.

C8

Evaluation of a Frequency Transposition Algorithm in a Behind-The-Ear Hearing Aid

Joanna Robinson, Thomas Stainsby, Thomas Baer, and Brian Moore

Department of Experimental Psychology, University of Cambridge, UK

Transposition of high-frequency information to lower frequencies may help people with high-frequency hearing loss. If the hearing loss is associated with a high-frequency 'dead region' - involving a complete loss of inner hair cell and/or neural function above a certain frequency, f_e - then conventional amplification may not be sufficient. Our previous research (Robinson et al 2007), presented at the IHCON 2006, evaluated the benefit of an FFT-based transposition technique in a lab-based study. Results showed that transposition significantly improved 's' detection. Some improvements of consonant identification were also seen. However, these were reduced by the introduction of new confusions. We hypothesised that further training and exposure would be needed to gain the full benefit of our transposition technique.

Implementing the processing in a wearable hearing aid allows this and we are now running a field trial on six listeners with high-frequency dead regions. The processing in the aids is the same as in our previous study. Frequencies up to $1.7f_e$ are left unprocessed, preserving information about voice pitch. When high frequencies dominate the spectrum, high-frequency components within a certain target range are transposed to the range f_e to $1.7f_e$ and are presented 'on top of' any original frequency components falling within this range. Otherwise, no transposition occurs. This conditional transposition aims to prevent the interfering influence of high-frequency background noise. In a control condition, stimuli are low-pass filtered at $1.7f_e$.

In our trial, listeners were fitted with both the transposing and control program. Target gains were verified using real ear measures. All volunteers were experienced hearing aid users. The listeners were instructed to use both programs equally and to record their preference in varying situations. Objective speech testing was performed on the transposing and control program as well as the listeners own aids one, three and five weeks after fitting. Tests included a vowel-consonant-vowel, an ‘s’-detection task and a speech reception threshold measurement. Preliminary results indicate that listeners are definitely aware of the extra information provided by transposition but more results from objective speech measures are needed to indicate whether this new information can be utilised to improve speech perception. [This work was supported by the RNID]

References:

Robinson, J.D., Baer, T. & Moore, B.C.J. 2007. Int J Audiol, 46, 293-308.

C9

Relating the Absence of Binaural Pitch Percept to Retro-Cochlear Impairment

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

Binaural pitch stimuli, created by introducing an interaural phase difference over a narrow band of otherwise diotic white noise, produce an immediate tonal sensation with a pitch close to the centre of the phase-shifted band. In Santurette and Dau [Hear. Res. 223(1-2):29-47, 2007], it was shown that the salience of binaural pitch was affected by hearing impairment. Specifically, for subjects with a sensorineural impairment, binaural pitch perception was weaker than the normal-hearing average but the pitch sensation was immediately present. In contrast, no binaural pitch sensation at all was found for the (only) two subjects with damage at central stages. The aim of the present study is to clarify whether such a sharp distinction between levels of impairment can be made using binaural pitch stimuli. A pitch detection test was performed by three groups of subjects with: 1) normal hearing; 2) a cochlear impairment with no sign of retro-cochlear impairment; and 3) a diagnosed retro-cochlear impairment. Subjects were asked to judge the pitch direction of series of five notes of equal duration (300, 600 or 900 ms), ranging from 523 to 784 Hz, presented either in an ascending, descending, or constant sequence. The results from two stimulus configurations, namely Huggins’ pitch stimuli and pure tones presented in diotic white noise, were compared. In addition to the pitch detection experiment, measures of frequency selectivity, fine structure and envelope processing, binaural interaction, and cognitive abilities, were obtained in order to investigate the correlation between these outcomes and results from the binaural pitch test. As no spectral cues are provided by binaural pitch stimuli, their perception is expected to heavily depend on the acuity of fine structure coding and the accuracy of the binaural system in combining the input from both ears. Overall, the absence of any binaural pitch percept is expected to be found only among subjects from group 3, while deficits at cochlear level are expected not to be sufficient to eliminate the perception of binaural pitch. If so, a binaural pitch test would be an interesting indicator of retro-cochlear deficit and useful for characterising the auditory profile of individual hearing-impaired listeners.

C10

Expectations, Pre-Fitting Counseling and Hearing Aid Outcome

Gabrielle Saunders, Samantha Lewis and Anna Forsline. National Center for Rehabilitative Auditory Research (NCRAR), Portland VA Medical Center, Portland, Oregon.
Researchers and clinicians often discuss the potential impact that expectations about a hearing aid likely have on hearing-aid outcome. Data collected with tools such as the Expected Consequences of Hearing aid Ownership (ECHO) tend to show that higher expectations are associated with better over-

all outcome. On the other hand, others have postulated that excessively high expectations can result in disappointment and thus poor outcome. It has also been suggested that it is important to counsel patients prior to fitting a hearing aid if expectations are unrealistic. Data, however, are mixed as to the effectiveness of such counseling in terms of whether expectations are altered and outcome is improved.

In this study, two forms of pre-fitting expectations counseling were compared. One form involved verbal discussion of the benefits and limitations of hearing aids, specific to the listening situations of particular interest to each participant. The other form was supplemented with aided auditory demonstration of those listening situations via a four-speaker system. Expected outcomes pre- and post-counseling were measured in terms of anticipated residual activity limitations and participation restrictions, and anticipated psychosocial benefit, using an adapted version of the Abbreviated Profile of Hearing Aid Benefit (APHAB), the ECHO and an adapted version of the Psychosocial Impact of Assistive Devices Scale (PIADS). These expectations data were compared to outcomes data measured with the APHAB, Satisfaction with Amplification in Daily Life (SADL) and PIADS following 8-10 weeks of hearing aid use. Data from 58 new hearing aid users will be presented. The data show that pre-fitting counseling altered expectations, but that in general expectations remained higher than eventual outcome following 8-10 weeks of hearing aid use. There were no differences in terms of changes in expectations or outcome between the verbal-only and verbal + auditory demonstration counseling.

C11

Application of the Acceptable Noise Level to Single Microphone Noise Reduction

Anne Schlueter, Institute of Hearing Technology and Audiology at the University of Applied Sciences, Oldenburg, Germany
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The common way to develop single microphone noise reduction algorithms is to measure the improvement of understanding with speech intelligibility tests. Unfortunately, the respective algorithms work efficiently at positive signal to noise ratios (SNRs), while the speech reception thresholds in speech intelligibility tests are reached at negative SNRs. Our approach for the solution of this problem is using the Acceptable Noise Level Test (ANLT) to measure how much background noise listeners tolerate. The ANLT consists of two steps. Within the first step, the subjects are asked to adjust speech to their Most Comfortable Level (MCL). Within the second step, background noise is added to the speech signal and the subjects adapt the noise to a maximum level of acceptance, i. e., speech understanding may not strain the subject and noise may not annoy. The Acceptable Noise Level (ANL) is defined as the difference between the subject's MCL and the adjusted noise level. In our investigations, the ANLT was used to determine the benefit of three single microphone noise reduction algorithms. One of the algorithms used a-priori knowledge about the noise and an optimal gain function according to Wiener. The two other real-world algorithms applied noise estimation (Minima Controlled Recursive Averaging) and spectral subtraction as the gain rule. The benefit of the algorithms was calculated from the difference between measurements with and without noise reduction. Outcomes were compared with results of the Oldenburg Sentence Test (OLSA), the Just Follow Conversation Test (JFCT) and a paired comparison test. Measurements were performed with subjects with normal and impaired hearing. The results show that the ANLT is a suitable tool to investigate single microphone noise reduction algorithms, since most adjusted noise levels resulted in positive SNRs, which are more suitable for this application. Unfortunately, interindividual variations of the ANL-values are high. In contrast to the ANLT, the speech intelligibility tests are not a tool to detect the benefit of real-world noise reduction algorithms. The results for OLSA and JFCT do not indicate any advantages, since the used SNRs are to low for this kind of algorithms. In the paired comparison test subjects with hearing impairment clearly prefer the situations with noise reductions, while normal hearing subjects disapprove algorithms with artefacts. Hence, the paired comparison test is also an applicable measurement for the investigation of noise reduction algorithms.

C12

Modulation Detection Interference in Hearing-Impaired Listeners with Nonlinear Amplification

Yi Shen, and Jennifer J Lentz, Indiana University

The detection of the depth of amplitude modulation of a signal carrier frequency (target) can be disrupted by the presence of other modulated carriers (maskers). This phenomenon, known as modulation detection interference (MDI), has not been well-studied in listeners with hearing loss. However, listeners with mild-to-moderate hearing loss do receive similar amounts of MDI compared to normal-hearing listeners for sinusoidally amplitude modulated signals. In an effort to better understand MDI in listeners with hearing loss, this study expands upon previous work by measuring MDI in listeners with more severe, sloping hearing losses and using narrow bands of noise as modulators (as loss of cochlear compression is expected to alter shape of the modulated waveform).

Listeners detected the presence of modulation imposed on a 500-Hz target tone presented at 80 dB SPL. The modulator had a bandwidth of 10 Hz, with modulation frequencies ranging between 2 and 12 Hz. When a masker tone was present, its carrier frequency was 2140 Hz. The presentation level of the masker was set to be equally loud as the target tone, ensuring audibility of the target and the masker. The masker tone was either unmodulated or modulated by a narrowband modulator with the same modulation frequencies as the target modulator and a modulation depth ranging between 0 to 1.

Results show that modulation detection thresholds increase with increasing masker modulation depth. The MDI, defined as the difference between thresholds for the fully modulated (m=1) and unmodulated (m=0) conditions, is about 10 dB. Modulation detection thresholds are very similar for both listener groups, indicating that hearing loss does not impact modulation detection in the presence of an interferer for these narrowband noise modulators. Such data suggest that hearing-impaired listeners have a relatively intact capacity for sound segregation/grouping based on the temporal envelope.

Although modulation detection thresholds are very similar between the two groups of listeners, these listeners all have audiograms that would typically be fit with a hearing aid with a compression algorithm (especially at 2000 Hz). These compression algorithms can reduce the effective modulation depth of an acoustic input signal and thereby would impact (decrease) MDI. In this way, compression algorithms might reduce sound segregation/grouping abilities in listeners with hearing loss. These data will be discussed in terms of the influence of compression algorithms using acoustic analyses and additional psychoacoustical data. [This work was supported by Indiana University.]

C13

Non-Native Listeners' Use of Context in Perception of Noisy and Reverberant Speech

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Non-native listeners have difficulty processing speech degraded by noise and/or reverberation. The effect of noise on non-native listeners' speech perception has been well established, but little has been done to evaluate how non-native listeners perform in highly reverberant environments (Nábělek & Donahue, 1984; Takata & Nábělek, 1990). Reverberation plays a significant role in speech perception, as it exists in most realistic listening environments and exerts a different masking effect over speech, compared to noise (Helfer, 1994). Context often helps limit phonetic and lexical alternatives to the speech signal (Boothroyd & Nittrouer, 1988), but these multi-level contextual cues may not be readily accessible to non-native listeners when the signal is degraded by noise (Mayo, et al., 1997; von Hapsburg & Bahng, 2006).

The current study was designed to further the investigation on how native and non-native listeners utilize context to combat the negative effect of noise and reverberation on speech perception. The reverberant Speech-Perception-in-Noise (SPIN) test (Sandridge et al., 2005) was presented to 10 native monolingual (NM), 10 native bilingual (NB), and 20 sequential bilingual (i.e., non-native, NN) listeners with normal hearing. Bilingual listeners varied greatly in their language background. Four combinations of signal-to-noise (SNR) and reverberation (RT) values were included (SNR = +6 & 0 dB; RT = 1.2 & 3.6 s) to simulate the acoustics of a typical classroom and auditorium in real life. For each of the eight experimental conditions (2 levels of contextual cues × 2 levels of noise × 2 levels of reverberation), two SPIN lists were randomly selected and presented at 45 dB HL through headphones. Each list included 25 high-predictability (HP) and 25 low-predictability (LP) sentences. Listeners' written response to the last word of every sentence was scored.

Preliminary results showed that all listeners' performance worsened as noise and reverberation levels increased. NN listeners performed poorly with both HP and LP sentences, whereas NM and NB listeners performed poorly only with LP sentences. Difference in recognition with and without context remained the same for NM listeners across test conditions; however, it became smaller for NB and NN listeners in the most degraded condition (0 dB SNR, 3.6 s RT). Difference between the least and most acoustic degraded conditions was the largest with NB listeners, but was comparable between NM and NN listeners. These findings revealed a complex pattern in how context benefits listeners in challenging listening situations, depending on listeners' language background.

C14

Independent Acclimatization for Localization and Speech Perception in New Hearing Aid Users

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It is well known that experience increases the benefit hearing aid users obtain from their hearing aids, but specific components of the acclimatization process and their relations and interactions leave many questions to be answered. What types of hearing aids will produce greater localization acclimatization? What types of hearing aids will produce greater speech perception acclimatization? Will improvements in localization facilitate improvements in speech perception? In the present study, independent measures of acclimatization for localization and for speech perception were carried out in the same groups of new hearing aid users to determine how these two components of acclimatization co-vary.

Adult participants with bilateral, symmetrical, mild to moderately-severe SNHL were fit with binaural ITC hearing aids. Participants were divided into four groups, using two different hearing aid platforms each programmed with either wide-dynamic-range multichannel compression (WDRMCC) or linear amplification (LA), employing the NAL-R target. The primary distinction between the two hearing aid platforms was the difference in processing time delay. One device used a fast Fourier transform algorithm (FFT) with a 10 ms time delay, while the other used a non-FFT algorithm with a 1 ms time delay. Laboratory measurements of noise-band localization and of consonant discrimination were done prior to and during the first 32 weeks of hearing aid use.

The present report focuses primarily on the comparisons of changes in localization and consonant-discrimination performance and how they varied with hearing aid platforms and signal processing programs. The primary factor determining localization acclimatization was the hearing aid platform, with the short time delay device producing significant improvements in noise-band localization and the long time delay device showing no consistent changes over time. There was neither an effect of amplification program (WRDMCC versus LA) nor an interaction of amplification program and hearing aid platform on localization acclimatization. In contrast to the localization results, the primary factor determining speech perception acclimatization was the amplification program, with WDRMCC producing significant improvements in consonant discrimination and LA showing little or none. Furthermore, there was neither an effect of hearing aid platform nor an interaction of hearing aid platform and amplification program on speech perception acclimatization. These results suggest that the combination of WDRMCC and the short time delay platform would facilitate speech perception in spatially segregated noise. [This work is supported by VA RR&D and NIDRR. Hearing aids provided by the manufacturers, Sonic Innovations and GN ReSound.]

C15

Loudness Scaling By Young and Elderly Normal-Hearing Listeners

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In a recent publication (Smeds, Keidser et al. 2006) loudness scaling data was presented. In the laboratory, normal-hearing and hearing-impaired listeners rated loudness for eleven audio-visually presented recorded real-life situations. The hearing-impaired listeners had the sound amplified according to NAL-NL1 (based on the long-term average sound level for each listening situation). It was found that the NAL-NL1 prescription provided the hearing-impaired listeners with less-than-normal calculated overall loudness. Despite the results from the loudness calculations, the hearing-impaired listeners rated the loudness of the listening situations higher than the normal-hearing listeners, especially for high-level situations. One proposed explanation of the results was that the loudness model used for the calculations (Moore and Glasberg 1997) was not appropriate. Another possible explanation was that the two groups of listeners differed not only in hearing ability, but also in age. The normal-hearing listeners were considerably younger than the hearing-impaired listeners. Perhaps there exists an age difference in how loudness is perceived or rated.

The current project investigated how young and elderly normal-hearing listeners rated loudness. The study also aimed at examining if the two groups of listeners rated loudness for artificial and realistic test stimuli in the same way.

The inclusion criteria for the study were that the participants should have pure-tone thresholds better than 20 dB HL between 0.25 and 6 kHz; the young participants should be between 18 and 25 years and the elderly participants should be older than 60 years. 20 younger (median age 21) and 20 elderly (median age 64) listeners participated. Artificial test stimuli (octave bands of noise with centre frequencies 0.5, 1, and 2k Hz) ranging from 45 to 85 dB SPL and realistic test sounds ranging from 50 to 85 dB SPL were presented. The participants rated the loudness of the sounds using category scaling in seven categories. The results showed that there was no statistically significant difference between the ratings from the two groups of listeners, neither for the artificial stimuli, nor for the realistic stimuli. Both groups rated the realistic stimuli slightly higher than the artificial stimuli with the same sound pressure level.

The results of the current study cannot explain the findings in the previously described study by Smeds et al. It would therefore be interesting to make an evaluation of the loudness model used in that study.

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C16

Evaluation of the International Outcomes Inventory for Hearing Aids in Veterans with Multi-Channel, Multi-Memory Hearing Aids

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The International Outcomes International Outcomes Inventory for Hearing Aids (IOI-HA) was developed as supplemental outcome measure to examine hearing aid outcomes in seven global domains (Cox et al., 2000). One item represents each outcome domain that includes the following: (1) daily use, (2) benefit, (3) satisfaction, (4) residual activity limitations, (5) residual participation restriction, (6) impact on others, and (7) quality of life. A different, 5-point response scale is used for each item, with the left-most response representing the least optimal outcome and the right-most response representing the most optimal outcome. The values are summed for a total score. Higher scores represent better hearing aid outcomes. The IOI-HA has been shown to have adequate psychometric properties (Cox & Alexander, 2002) and was normed on an adult population with single-memory and single-channel hearing aids (Cox, Alexander, & Beyer, 2003).

The overarching goal of this study was to evaluate the IOI-HA in patients who were fitted with multi-channel, multi-memory hearing aids. The specific purposes of this study were as follows: (1) to establish the psychometric proprieties (i.e., factor structure and internal consistency) of the IOI-HA in this population, (2) to evaluate differences in IOI-HA scores based on a number of demographic and audiologic variables (e.g., hearing aid experience, degree of hearing loss, hearing handicap scores based on the Hearing Handicap Inventory for the Elderly-Screening, etc.) for which group norms may be developed, (3) to determine test-retest reliability of the IOI-HA, and (4) to determine critical difference scores. A sample of veterans who were issued digital hearing aids between November 2003 and March 2005 at the VA Medical Center, Mountain Home, Tennessee was mailed the IOI-HA six weeks after their fitting. Useable IOI-HA questionnaires were obtained from 564 (65.8% response rate) veterans (mean age of 68.8 years, SD = 9.7). To evaluate test-retest reliability and to obtain critical difference scores, two additional copies of the IOI-HA were mailed to 250 veterans, two weeks apart. A total of 130 veterans completed both copies of the IOI-HA (52% response rate). The principal component analysis with varimax rotation resulted in a two-factor solution similar to the factor structure found by Cox and Alexander (2002) initially. The internal consistency (Chronbach's α = .071) and test-retest reliability was good (λ = 0.94) for the total scale. Group norms and critical difference scores currently are being evaluated.

C17

A Large-Scale Substantiation of Own-Voice Issues in Hearing-Aid Users, Part II: Reducing Occlusion Problems Is Still Important

Niels Søgaard Jensen, Søren Laugesen, Patrick Maas, Marie Louise Kamp González Cárdenas, and Sidsel Mørch Rysager, Eriksholm, Oticon Research Centre, Oticon España S.A., and ViSP, Resource Centre for Special Needs Education.

In a companion presentation (part I), Laugesen et al. report on a questionnaire study (utilizing the Own Voice Qualities (OVQ) questionnaire) where the main hypothesis under test was that hearing-aid users have other issues and concerns related to their own voice besides the well-known problems caused by occlusion. This hypothesis was strongly confirmed by the questionnaire data.

In the same study, a secondary hypothesis was that hearing-aid users who are exposed to occlusion (due to an unfortunate combination of hearing loss and vent size) will experience more own-voice issues than hearing-aid users, who are not exposed to occlusion. Accordingly, the 187 participating hearing-aid users were recruited so one third could be included in a group expected to suffer from occlusion problems (due to small low-frequency hearing losses and small hearing-aid vents) while the remaining two thirds could be included in a group not expected to suffer from occlusion problems (due to either large low-frequency hearing losses or large hearing-aid vents).

Surprisingly, the questionnaire data did not support the secondary hypothesis. The group expected to suffer from occlusion did not in fact report about more own-voice issues than the other group. Rather than questioning the evidence that open hearing-aid fittings provide major improvements on occlusion-related issues, the data indicate that 'self-selection' has played a significant role in the recruiting of test subjects, since all test subjects evaluated the own-voice perception with their own hearing aids. It is therefore quite likely that the test subjects who actually had decided to buy and use small-vent hearing aids are people who are simply not bothered by occlusion.

These results led to a follow-up study where 43 test subjects with small low-frequency hearing losses compared open fittings with small-vent fittings (using the same type of hearing aid) in a balanced cross-over design. Each type of fitting was used for a period of one month before the OVQ questionnaire was filled in. The data showed that significantly more own-voice issues were reported with small-vent fittings than with open fittings. This finding supports both the secondary hypothesis (i.e., reducing occlusion problem reduces own-voice issues) and the explanation for the observations made in the first study. Data from both studies will be presented and discussed.

C18

Effect of Amplification on Consonant Modulation Spectra

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Current auditory models suggest that speech information is conveyed by a composite of modulations at multiple rates, superimposed on a carrier signal, and that this modulation spectrum can be used to characterize available acoustic information. Our previous work[1] demonstrated that unamplified consonants with similar modulation spectra were likely to be confused with one another. This study addressed two questions: (1) how do hearing aids alter modulation spectra? (2) what effect do these alterations have on consonant confusions?

Fifteen adults with bilateral mild-to-moderate sensorineural loss were fit with a behind-the-ear hearing aid. Consonant recognition was measured for a set of 22 consonant-vowel syllables under two amplification conditions: multichannel fast-acting wide-dynamic range compression (WDRC) and linear; each at 3 input levels (50, 65, 80 dB SPL). Attack and release times were 5 and 100 ms, respectively. Frequency-gain response was individually adjusted using NAL-NL1 targets and verified with probe microphone measures. Final data consisted of a confusion matrix for each subject in each condition (2 amplification x 3 input levels).
To capture individual amplification effects, each amplified syllable was recorded at the tympanic membrane of each subject using a probe microphone and digitally stored for analysis. Spectral Correlation Index[1] (SCI) values were calculated for each condition. The SCI[1] is obtained by deriving modulation spectra (six octave-spaced carrier frequencies [250-8000 Hz] by six octave-spaced amplitude modulation frequencies [1-32 Hz]) over the duration of an individual phoneme. Similarity across phonemes in a stimulus set is then obtained by correlating the six modulation spectra (one for each octave) for each possible pair of phonemes in a stimulus set.

As expected, fast-acting multichannel WDRC amplification produced the greatest SCI change, but significant changes were noted even with linear amplification. There was considerable variability across individuals that did not relate in a simple way to amount of loss, audiometric slope or frequency-gain response. The amount of alteration also varied by consonant with the greatest change for affricates. For listeners with mild-to-moderate loss, who presumably had good frequency selectivity, results demonstrated that the modulation characteristics can be altered to a significant extent without degrading consonant recognition. The acceptable range of alteration will be discussed and could serve as a useful index to determine acceptable amplification parameters.

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Gallun F., Souza P. Exploring the role of the modulation spectrum in phoneme recognition. Ear and Hearing, in press.

C19

Stream Segregation Due To Apparent Spatial Location: Effects of Interaural Level and Time Differences (ILDs & ITDs)

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The effects of Interaural Time and Level Difffererences (ITDs and ILDs) on sequential streaming were investigated over three experiments using five normally hearing subjects. In experiment 1, the threshold ITD for which listeners perceived off-centre lateralisation of a sequence of interleaved 'A' and 'B' complex tones was determined. Both complexes were 60 ms in duration, had a 100-Hz fundamental frequency, components in cosine phase, and were filtered into passbands of 353-707, 500-1000, 1250-2500, 1768-3536, or 2500-5000 Hz. Experiment 2 measured an equivalent ILD threshold. ITD thresholds were found to increase with increasing passband centre frequency, while ILD thresholds decreased. In experiment 3, the effects of ITD and ILD on obligatory stream segregation for the 353-707-Hz and 1768-3536-Hz passbands were investigated using ITD and ILD values that were 300% of each individual's thresholds. The conditions were (1) no ITD or ILD, (2) ITD alone, (3) ILD alone, (4) congruent ITD and ILD, and (5) opposing ITD and ILD. Based on the assumption that timing information is more difficult to judge across than within streams, the degree of stream segregation was measured by finding the detection threshold for a delay introduced to an otherwise isochronous sequence of interleaved 'A' tones and 'B' tones, using an adaptive two-interval forced-choice task (Roberts et al., 2002. J. Acoust. Soc. Am. 112: 2074-2085). This measure is termed the 'anisochrony threshold'. Values tended to be higher (indicating more stream segregation) for conditions 2-5 than for condition 1, although the effect of condition was significant only for the higher passband. However, the thresholds in conditions 2-4 were lower than those which have been observed in other experiments where stream segregation was manipulated by factors such as differences in phase or magnitude spectrum. This suggests that, while differences in apparent spatial location produced by ITD and ILD cues can produce obligatory stream segregation, they are not especially effective in doing this. The effects of ITDs and ILDs neither added nor cancelled, suggesting that they are perceptually independent.

Acknowledgements

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C20

Effects of Fast-Acting Dynamic-Range Compression on Stream Segregation in Normal-Hearing Listeners

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Dynamic range compression is used extensively in hearing prostheses to map the range of real-world signal levels into the residual dynamic range of the hearing impaired. Benefit from compression depends on a range of factors such as listening experience and mental agility (Gatehouse, Naylor and Elberling, Int J. Audiol. 45:153-71, 2006). Stone and Moore (J. Acoust Soc Am 116: 2311-23, 2004) using noise-vocoder simulations of severe hearing impairment identified a mechanism by which the varying gain applied by a compressor adversely affects speech intelligibility. Independent sound sources become 'cross-modulated' within a compression channel by the application of a common gain signal, and this tends to promote perceptual fusion of the sources.

In the present study, young, normal-hearing university students were presented with pairs of sentences from the 'CRM' corpus (Bolia et al., J. Acoust Soc Am 107:1065-66, 2000) recorded using male speakers of British or Irish English. The paired sentences had no keywords in common and also were of similar duration, and therefore rhythm. Processing was with a fast-acting 4-channel compression, each channel using a compression ratio of 1.0 (linear), 1.82, or 9. Subjects were required to identify the keywords by clicking out their response on a computer screen. Intelligibility tests often produce null results when scoring only measures of accuracy. Baer, Moore, and Gatehouse (J. Rehab Res Devel, 30:49-72, 1993) used reaction time as a measure additional to signal-to-background ratio, and showed that it was more sensitive to changes in processing conditions. Here, on half of the trials, a visual distracter appeared at a random position on the screen and required cancellation before completion of the scorecard. In addition to measuring the accuracy of the completed scorecard, the times it took subjects to fill it as well as to cancel the visual distracter were recorded. Following a single training session, data collection occurred in two sessions, all held on separate days.

Strong learning effects were observed for score and reaction times: initial significant effects of processing disappeared between the sessions. Increasing the degree of compression majorly slowed the filling of the scorecard in the presence of the visual distracter, but did not affect the subsequent score. In summary, normal-hearing subjects were adversely affected by compression, primarily when cognitive demands were high. However, this effect disappeared with increasing exposure to the task.

C21

The Effects of Interchanging Hearing Aid and FM Systems across Manufacturers

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Children with hearing loss use frequency modulated (FM) systems in their classrooms to improve speech recognition in noise. It is possible that a school district may need to interchange equipment across manufacturers because of upgrades and repairs. The performance of FM equipment from two manufacturers, Oticon and Phonak, was verified electroacoustically following the AAA Clinical Practice Guidelines: Remote Microphone Hearing Assistance Technology for Children and Youth Birth-21 years (2008). Two digital hearing aids were programmed for a 40 dB HL, flat sensorineural hearing loss. Four equipment configurations were assessed using a Phonak Campus S transmitter and MLxS receiver and an Oticon Amigo T21 transmitter and Amigo R2 receiver. The FM Receivers were programmed for three FM Advantage settings, +10, +14 and +18 dB. The difference between the outputs of the hearing aid alone and the hearing aid plus FM varied up to 7 dB across configurations despite a constant FM Advantage setting in the FM Receiver. Effects of FM settings including Pre-emphasis/De-Emphasis and Direct Programmable Audio Input (DPAI) were found to result in considerable variability in output. These results suggest that FM equipment may be interchanged successfully across manufacturers when the hearing aid settings are appropriate; however, electroacoustic verification is necessary to ensure appropriate settings for optimal speech recognition.

C22

The Influence of Audibility on Asynchronous Double Vowels with the Same and Different Fundamental Frequencies

Susie Valentine, Starkey Hearing Research Center

The hypothesis that reduced audibility decreases benefits received from onset asynchrony in double-vowel identification was investigated in normal-hearing (NH) and hearing-impaired (HI) listeners. Normal-hearing (NH) and hearing-impaired (HI) listeners were presented with two concurrent, synthetic, steady-state vowels and were asked to identify each vowel in the order in which the vowels were heard. One vowel had a duration of 250 ms (target vowel), while the duration of the other vowel was 275, 350 and 450 ms (distracter vowel). Because the double vowels had simultaneous offsets, the duration differences between the two vowels created an onset asynchrony of 25, 100 or 200 ms. The two vowels could either have the same fundamental frequency (f0) or different fundamental frequencies that were separated by two semitones (possible fundamental frequencies 126 and 141 Hz). To determine whether reduced audibility in high-frequency regions would decrease the benefits received from onset asynchrony, double-vowel identification was measured for these synthesized vowels but also was measured for the vowels low-pass filtered at 900 Hz.

Results indicated that an increase in the onset asynchrony between the two vowels improved double-vowel identification for both groups of listeners in both identification of the distracter vowel and the target vowel regardless of the fundamental frequency combination. Less benefit from increases in onset asynchrony was received by the HI listeners, with a larger decrease in benefit observed for the target vowel only when the two vowels have the same fundamental frequency. Filtering of the double-vowel stimuli led to a large decrease in identification scores and a reduction in the benefits received from onset asynchrony for both groups of listeners regardless of whether the vowels had the same or different fundamental frequencies.

Data obtained from the unfiltered conditions suggest that hearing-impaired listeners receive less of a benefit from onset asynchrony differences than normal-hearing listeners, but only when a single sound segregation cue (onset asynchrony) is present. When a second sound segregation cue (f0 differences) is included, benefit from onset asynchrony is restored for the hearing-impaired listeners. Data obtained from the filtered conditions indicate that reduced audibility of high-frequency components contributes to the lower onset asynchrony benefit observed in data obtained from hearing-impaired listeners. Therefore, loss of cochlear function as a result of sensorineural hearing loss degrades the use of onset asynchrony as a sound segregation cue; however the introduction of fundamental frequency differences aids the use of onset asynchrony for hearing-impaired listeners.

C23

Improvements in Speech Perception and Sound Localization in Hearing Aids Using Binaural Multichannel Wiener Filtering

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Multi-microphone noise reduction algorithms are commonly implemented in modern hearing aids to improve speech intelligibility in noisy environments. The development of these algorithms has mostly focused on monaural systems. The human auditory system is a binaural system which compares and combines the signals received by both ears to perceive and localize a single sound source. Providing two monaural, independently operating, noise reduction systems (a bilateral configuration) to the hearing aid user may disrupt binaural information, needed to localize sound sources correctly.

In this study, three multi-microphone noise reduction algorithms were evaluated with respect to their influence on speech intelligibility and on the ability to localize sound sources. This was done using theoretical, objective and perceptual evaluations in different spatial scenarios. Two recently developed noise reduction techniques for binaural hearing aids were evaluated, namely the binaural multichannel Wiener filter (MWF) and the binaural multichannel Wiener filter with partial noise estimate (MWF-N). The binaural MWF theoretically preserves the binaural cues of the speech component. To preserve the binaural cues of both the speech and the noise components, the MWF-N was developed. This algorithm, in theory, sacrifices some noise reduction to preserve the binaural cues of the noise component. During the different evaluations, a bilateral adaptive directional microphone (ADM) was used as a reference system since it is widely used in commercial hearing aids.

The main conclusions are:

a) The ADM only preserves localization in the forward direction. In these directions limited or no speech-in-noise enhancement is obtained.

b) The MWF preserves localization of the target speech component but can distort localization of the noise component. Objective and perceptual evaluations showed that these distortions are often smaller than those predicted by theory. Moreover, they are dependent on signal-to-noise ratio and masking effects. By adding more contralateral microphone signals to the binaural MWF, the noise reduction performance significantly improved.

c) The MWF-N improved the ability to localize the noise component when compared with the MWF. Objective performance measures showed that this came at the cost of noise reduction. However, perceptual evaluations did not show this tendency. When speech and noise components are spatially well separated, the MWF-N even outperformed the MWF in terms of speech intelligibility. This can be explained by an increased release from masking when preserving the binaural cues of both the speech and noise components.

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C24

Is There A Relationship Between Pure Tone Thresholds And Speech Recognition In Noise Ability?

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The literature appears to present conflicting reports on the relationship between the audiogram and speech recognition in noise ability. A strong relationship would indicate that speech recognition in noise ability may be inferred from the audiogram. A poor relationship would indicate the need for a direct measure of hearing in noise ability. Some investigators have argued that it is desirable to use a speech recognition in noise test where the data are strongly correlated to the audiogram (Wilson et al., 2007). Others have argued that speech recognition in noise and pure tone thresholds represent two uncorrelated components of audition (Plomp and Mimpen, 1979).

This meta analysis will examine how subject sample configuration, masker noise type and "world-view" of the authors may affect the study outcomes. A review of the literature and results from HINT studies at the House Ear Institute will be used to offer an explanation for the varied conclusions across the literature. The significance for various areas of audiology will be presented along with recommendations for clinical use of speech recognition in noise test.

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C25

Characterization of Speech Amplification for Modern Hearing Instruments

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Current ANSI and IEC standards do not provide adequate methods for characterizing the speech amplification of modern hearing instruments with non-linear processing such as multi-channel dynamic compression, noise reduction, feedback suppression and new processing schemes to be expected in the future. The reason for this is that special test signals must be used that have a limited relation to real-life speech (e.g. sinusoids, stationary noises), and secondly that the hearing instrument must be set in a special test setting that will switch off part of its non-linear signal processing. In addition, the results are presented as static quantities (gain, output) for specific input levels whereas no, or limited, information is obtained concerning the dynamic gain or output characteristics for speech. For this reason, the European Hearing Instrument Manufacturer Association (EHIMA) has set up the ISMADHA working group that has developed a new proposal for a standard. This proposed new standard includes a new measurement method for characterizing speech amplification, a new speech like test signal (International Speech Test Signal or ISTS, see parallel contribution by I. Holube et al.), as well a new set of standard reference audiograms. The method has been evaluated in two rounds of testing in several laboratories. Recently the proposal was submitted to the hearing aid workgroups of ANSI and IEC.

The present contribution gives an overview of the essential aspects of the new measurement method. It will describe the application of percentile analysis for determining the output and gain characteristics of speech elements at 30, 65 and 99 percentile levels corresponding to the soft, normal and loud parts of speech. The outcomes of the new method will be discussed for a number of settings of a non-linear hearing instrument as evaluated in round-robin testing executed at several laboratories.

C26

Internet Hearing Screening Tests In Europe

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One of the prime objectives of the European HearCom project (www.hearcom.org) is to develop a number of self-diagnostic, screening, hearing tests for use via Internet or telephone. Building on the initiative of the Dutch National Hearing Test, the Triple-Digit Screening test was developed for several additional languages. Next to Dutch, these languages are: English, German, Swedish, Polish and French. For the near future, Greek and Turkish versions are in preparation. The Triple-Digit Screening test measures 50%-correct speech-reception threshold using lists of digit triplets presented in variable background-noise levels. The resultant threshold signal-to-noise ratios show very good correlation to speech in noise tests using sentences and to the audiogram. The tests are self administered by users keying in the perceived digits on the telephone or computer keyboard.

Recently HearCom has introduced a second and novel Internet screening test to evaluate hearing localization skills. For this, the minimum audible angle is measured by requiring participants to indicate the directions of sounds, moving from left to right or vice versa. The sounds are produced at varying angles from two normal PC speakers using a cross-talk cancellation technique. The advantage of this test is that it is language independent, and it is expected to be able to indicate hearing problems that are, in part, independent of speech perception in noise.

The present poster presents both Internet screening methods, their characteristics, and recent validation results. In particular, results will be presented from an evaluation and validation study in which individual results from these tests are compared to other diagnostic measurements, presently performed in three clinics in the Netherlands and Germany.

C27

Speech Intelligibility Improvements in Noise with Ideal Binary Time-Frequency Masking

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Recent research in both computational auditory scene analysis and blind source separation makes use of time-frequency (T-F) masking techniques. Ideal binary time-frequency masking retains mixture energy in T-F units where local signal-to-noise ratio exceeds a certain threshold, and rejects mixture energy in other time-frequency units. Two experiments were designed to evaluate the effects of ideal binary masking on speech intelligibility of both normal-hearing (NH) and hearing-impaired (HI) listeners in different kinds of background interference: speech-shaped noise and cafeteria noise. The results from Experiment 1 show that ideal binary masking leads to substantial reductions of speech reception threshold for both NH and HI listeners, and the reduction is greater in cafeteria noise than in speech-shaped noise. Furthermore, listeners with hearing loss benefit more than listeners with normal hearing, particularly for cafeteria noise. Remarkably, after ideal masking, intelligibility performances of NH and HI listeners in noisy backgrounds are statistically indistinguishable. The results from Experiment 2 suggest that ideal binary masking in the low-frequency range, i.e. less than 1.35 kHz, yields larger intelligibility improvements than in the high-frequency range, i.e. greater than 1.35 kHz; the differences are especially conspicuous for listeners with hearing loss. The findings from the two experiments have major implications for understanding speech perception in noise and hearing aid design.

C28

Real-World Benefit of an Adaptive Null-Steering Algorithm

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Hearing aids with adaptive null-steering and/or fixed directional processing are currently available on the market. Although the noise reduction of null-steering can be tens of dB greater than that of fixed processing under some laboratory conditions, it is unclear if this benefit can be found in real-world conditions. Woods and Trine [Acoustics Research Letters Online 5(4), October 2004] showed that the theoretical benefit of adaptive null-steering over the average performance of the best fixed system was only 2 dB in the most favorable of expected real-world conditions. Desloge and Zurek (2004) demonstrated an even smaller advantage with null-steering setups using real-world recordings made in a home, a parking garage, and outdoors. Despite numerous other reports in the literature on adaptive null-steering, none has yet established the range of quantitative benefit of adaptive null-steering over fixed directional processing in a comprehensive survey of real-world conditions. This presentation aims to provide that information.

A behind-the-ear processor was programmed to provide two directional systems in parallel: an adaptive null-steering in 15 frequency bands, and a wideband fixed-null system. The minimum short-time power of the output of each processing type was tracked over time, and the difference in minimum power across types was taken as an estimate of benefit of one type over the other. The difference in minima and several other statistics were stored in a multi-dimensional histogram in the processor memory once every 30 seconds. Subjects wore the processor over multiple 3-hour periods while going about everyday activities, providing a dosimeter-like survey of the difference in minima in settings such as outdoors, office, home, auto, etc. Results on when and by how much adaptive null-steering provides benefit over fixed directional processing will be shown as function of type of environment and other acoustic variables.

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Nonlinear Feedback Cancellation in Hearing Aids with Mutual Infomax Algorithm

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Feedback cancellation algorithms have been implemented in digital hearing aids to provide listening comfort and to enhance the acoustic conditions. Some studies have suggested that the benefit of the least mean squares (LMS) algorithm is used to reduce acoustic feedback at high gain. This decorrelates the output signal with the reference feedback signal and the input signal based on second-order statistics. So their performance is often limited because of the correlation between the input and feedback signal. However there may exist many other components in input signal which depend on the feedback signal in higher-order statistics. In the present study, we proposed a nonlinear feedback cancellation algorithm based on mutual infomax algorithm in independent component analysis (ICA) for digital hearing aids. The method can remove feedback components based on statistical independence, which involves statistics of high orders. For that purpose, higher-order statistical information using mutual infomax algorithm is applied to reduce feedback signal by making the output signal of the proposed feedback canceller as much as possible independent of the output signal.

Computer simulations were performed using recorded real speech and speech shaped noise which was generated from zero-mean white Gaussian noise passed through a 16 pole AR filter. Korean sentences were recorded as the real speech with a 16kHz sampling rate. And the car and factory noise were obtained in NOISEX-92 CD-ROMs. It is known that speech signals approximately follow a Laplacian distribution. Therefore, sign function was used as the score function of adaptive filter coefficient. The coefficients of the adaptive filter were continuously updated based on the input signal. The performance of our method was compare with that of a conventional normalized least mean squares (NLMS) algorithm from the viewpoint of misalignment and SNR value. Simulation results showed that the mutual infomax algorithm using high-order statistics provides better feedback cancelling performance with Laplacian distribution signal with various noise situations than conventional NLMS methods. In future study we will need to confirm proposed method performance in the real environment involving hearing loss patients. [This work is the result of research activities of Advanced Biometric Research Center (ABRC) supported by KOSEF. (R11-2001-094-05004-02)]

C30

Comparisons in Consonant Confusions and Loss Profiles With and Without Linear Frequency Gains in Hearing Impairment under Noisy Environment

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The previous study (Yoon et al., IHCON 2006) showed that the difference in SNR, required for the hearing-impaired (HI) to reach the same performance of normal listeners (NH) on consonant recognition was the greatest for syllables /βα/, /ϖα/, /Δα/, and /Tα/. It is hypothesized for such poor performance, that the articulation index is less than 1 over speech frequency bands, that is, the speech spectrum is not audible because of higher thresholds. In this study, we tested this hypothesis by measuring consonant loss profile (CPL) and perceptual confusion patterns (CP) in noise with stimuli spectrally compensated (spectral gain) for 6 hearing impaired listeners. Thus, the purpose of the present study was to determine the effect of audibility on 16 consonant-vowel (CV) nonsense syllables and on perceptual confusions in noise (-12, -6, 0, 6, 12, Q dB SNR) per HI listener. The results demonstrated that the benefit of providing audible speech to HI was both consonant and listener specific. Increasing the audibility of speech was beneficial to three listeners, but not so for the other three listeners. The CLPs also revealed that the listeners who generally showed negative effects of gain, did so consistently for targets /βα/, /φα/, /γα/, and /κα/, but each also benefitted on certain consonants such as /τα/ and /Zα/. The CPs demonstrated two major confusion patterns. First, the primary competitors were the same between gain and no-gain conditions, but the error rates were different. Second, the primary competitors were different or additional competitors were added to the confusion with gain, but the error rates were relatively constant. Thus, audibility is one of the primary factors influencing speech recognition of HI, but reduced audibility alone cannot explain the difficulty of HI to understand speech in noise. Similarly, amplification alone cannot restore the speech recognition ability of most HI to the level of NH.

C31

The Effects of Different Environmental Noise Reduction Configurations on Speech Understanding and Listener Preferences

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It is common for environmental noise reduction (ENR) algorithms in hearing aids to adjust the gain based on the modulation depth and/or the noise level, with the relationship between the gain and acoustic parameters differing among aids. Several studies have investigated the effects of turning on or off the ENR algorithm of a hearing aid in combination with a directional microphone. These studies typically evaluated speech understanding and subjective preferences with recorded stimuli. Comparisons of the efficacy of the different ways of fitting ENR in these studies are affected by differences in the processing performed in the ENR algorithms, test conditions, hearing-aid hardware, WDRC fittings, and possibly other algorithms such as adaptive directional microphones. We are not aware of studies that compare the objective or subjective effects of different ENR algorithms with otherwise identical signal processing and hearing-aid hardware, and investigate the interaction of ENR with low-level expansion. Few studies report subjective data collected in real-life listening conditions.

The aims of the present study were to investigate the effects of different approaches to fitting ENR on: 1) objective measurement of speech understanding; 2) subjective preferences for speech-in-noise and noise-only conditions encountered in everyday life; 3) speech understanding in conditions where low-level expansion is active. The same DSP hearing aid was used for all conditions, and was programmed to run WDRC, an adaptive directional microphone, feedback management, and ENR. The ENR algorithm used the widespread approach of estimating the SNR from the modulation depth in each channel. Four different configurations of the ENR algorithm were evaluated. For two configurations the maximum possible SNR-based gain reduction was set to be the same in all channels, and for the other two it was shaped across channels to minimize the effect on the SII. For both configurations of SNR-based gain reduction, the maximum gain reduction was set to be either invariant with the noise level, or configured to increase at high noise levels and decrease at lower noise levels (below the expansion threshold of the WDRC algorithm). The effects of different ENR configurations and no ENR on speech understanding were evaluated with SRTs in quiet, speech-shaped noise, and multiple-talker babble. The subjects also performed paired comparisons between different configurations and no ENR in real-life low- and high-noise conditions, with and without speech present. This study is currently in progress, and we will report the final results.

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