IHCON 2006

International Hearing Aid Research Conference 2006

August 16 - 20, 2006

GRANLIBAKKEN CONFERENCE CENTER
LAKE TAHOE, CALIFORNIA

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House Ear Institute

National Institute on Deafness and
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Meng Yuan	The Chinese University, Hong Kong
Yun Zheng	West China Hospital of Sichuan University, China

Daily Schedule

WEDNESDAY, AUGUST 16

5:00 PM	Welcome Social *Sponsored by Grandlibakken to recognize House Ear Institute's 60th Anniversary*
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 17-19

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 20

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 (buses leave for airport with box lunches for passengers)

PROGRAM SUMMARY

WEDNESDAY, AUGUST 16

WELCOME AND KEYNOTE ADDRESS

7:30PM - 8:45PM

Welcome Remarks: Sig Soli
Stuart Gatehouse

KEYNOTE ADDRESS

Patrick M. Zurek	The evidence for supra-threshold deficits accompanying cochlear hearing loss

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THURSDAY, AUGUST 17

SESSION ONE

8:00AM –9:45AM

PSYCHOACOUSTICS AND SPEECH

Moderator: Judy Dubno

Andrew J. Oxenham	Perceptual consequences of normal and abnormal cochlear function: Implications for hearing aids
Debi Vickers	Band-importance functions for normal-hearing and hearing-impaired listeners
Aparajita Bhattacharya	Companding to improve hearing aids speech recognition in noise

POSTER SESSION A 9:45AM – 11:00AM

SESSION TWO

11:10AM –12:20PM

MODELS

Moderator: Michael Stone

Torsten Dau	Auditory processing models and their potential for enhancing the quality of hearing technology
Koenraad S. Rhebergen	A model approach to predict the speech intelligibility in fluctuating noise for signals with a normal dynamic range and for signals with different forms of compression of the dynamic range

SESSION THREE

5:15PM - 7:00PM

FITTING AND BENEFIT

Moderator: Ole Dyrlund

Ben Hornsby	Factors affecting the benefit of asymmetric directional fittings
Bert de Vries	Bayesian machine learning for personalization of hearing aid algorithms
Deniz Baskent	Using genetic algorithms with subjective input for fitting auditory devices

FRIDAY, AUGUST 18

SESSION FOUR

8:00AM –9:45AM

EMERGING TECHNOLOGIES

Moderator: Laurel Christensen

Michael F. Dorman	The recognition of speech, voice and music using combined acoustic and electric hearing
Sigfrid D. Soli	Initial results from an implantable "Round Window Hearing Aid"
Christopher J. Long	Binaural unmasking with bilateral cochlear implants

POSTER SESSION B 9:45AM – 11:00AM

SESSION FIVE

11:10AM –12:20PM

REHABILITATION AND COMMUNICATION

Moderator: Mary Florentine

Louise Hickson	Optimizing communication for hearing aid users: A randomized control trial of a group intervention program
Robert W. Sweetow	Beyond amplification: Listening and communication enhancement

SESSION SIX

5:15PM – 7:00PM

THERE'S MORE TO LISTENING THAN GOES ON IN YOUR EARS

Moderator: Jim Kates

Richard L. Freyman	Informational masking – what is it and how does it matter in sensorineural hearing loss?
Ervin R. Hafter	The role of shared attention in perceptual processing and its potential influence on the study and use of hearing-aids
Gurjit Singh	Cognitive and auditory factors underlying spatial attention in older adults

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SATURDAY, AUGUST 19

SESSION SEVEN

8:00AM – 9:45AM

LEARNING AND TRAINING

Moderator: Pam Souza

Beverly A. Wright	Human discrimination learning on basic auditory tasks
Sepp Chalupper	Learning optimal gain in real world environments
Larry E. Humes	The effects of word-based training on aided speech recognition and identification

POSTER SESSION C 9:45AM – 11:00AM

SESSION EIGHT

11:10AM –12:20PM

COMPRESSION

Moderator: Wouter Dreschler

Michael A. Stone	Quantifying the effects of fast-acting dynamic range compression on the envelope of speech
Graham Naylor	Fast-acting compressors change the effective signal-to-noise ratio - both upwards and downwards!

SESSION NINE

5:15PM – 7:00PM

SIGNAL PROCESSING

Moderator: Larry Humes

Justin A. Zakis	The acoustic and perceptual effects of noise-suppression algorithms
Simon Doclo	Perceptual and theoretical evaluation of the Interaural Wiener Filter (IWF) algorithm with respect to speech reception thresholds
Vijay Parsa	Speech quality evaluation of feedback cancellation technologies in hearing aids

SUNDAY, AUGUST 20

SESSION TEN

8:00AM – 9:10AM

SOUND QUALITY

Moderator: Brent Edwards

Todd A. Ricketts	The effect of bandwidth on sound quality in listeners with normal to moderate hearing loss
Stefan Fredelake	Comparison of objective and subjective measures for sound quality and speech intelligibility in nonlinear hearing instruments

BREAK 9:10 AM –9:30 AM

SESSION ELEVEN

9:30 AM – 10:40 AM

OWN VOICE & WHAT HAVE WE LEARNED?

Moderator: Brent Edwards

Søren Laugesen	Effects of amplification variations on vocal level and own-voice sound quality of hearing-aid users
Harvey Dillon	IHCON 2006: Themes, consensus, and divergent thinking

ADJOURNMENT

Oral Program
Wednesday, August 16

KEYNOTE ADDRESS

7.45PM THE EVIDENCE FOR SUPRA-THRESHOLD DEFICITS ACCOMPANYING COCHLEAR HEARING LOSS

Patrick M. Zurek, Sensimetrics Corporation

An important goal for the field of hearing science is to characterize the losses in auditory acuity, beyond elevated absolute thresholds, that result from cochlear hearing impairment. Progress towards this goal could have immediate practical significance for the design of signal processing algorithms for hearing aids. However, after hundreds of studies comparing hearing-impaired and normal-hearing listeners' performance on a variety of auditory detection, discrimination, and speech-reception tasks, conclusions regarding supra-threshold deficits are still difficult to reach. This difficulty is partly due to the variability in the results of hearing-impaired listeners and partly due to the unknown influence of several confounding variables. The possible role of stimulus audibility and/or overall level in studies of hearing-impaired listeners has been recognized for some time. With increasing evidence of declining auditory performance among older listeners, the importance of age as a confounding variable must also be considered. This talk will summarize a critical review of the literature on supra-threshold deficits and their relations to audiometric data.

Thursday, August 17

SESSION ONE

PSYCHOACOUSTICS & SPEECH

Moderator: Judy Dubno

8.00AM PERCEPTUAL CONSEQUENCES OF NORMAL AND ABNORMAL COCHLEAR FUNCTION: IMPLICATIONS FOR HEARING AIDS

Andrew J. Oxenham, Massachusetts Institute of Technology

Physiological studies over the past two decades have consolidated our understanding of the cochlea as a highly nonlinear transducer of sound. Audio engineers work hard to eliminate nonlinearities from sound transduction systems, such as microphones and loudspeakers, because they produce unwanted distortions. In the case of the ear, however, it turns out that the cochlea's nonlinearities are in large part responsible for the tremendous dynamic range and exquisite frequency selectivity of human hearing, both of which are vital to our ability to function in the acoustic environment. Cochlear damage often results in a loss of these nonlinearities, and this loss can account for many of the difficulties faced by people with cochlear hearing loss. This talk will explore ways in which properties of the cochlea, including its nonlinearities, can be measured behaviorally in humans, and will review the perceptual consequences of a loss of cochlear nonlinearity in people with cochlear hearing impairment, as well as ways in which this loss might be mitigated.

8.35AM BAND-IMPORTANCE FUNCTIONS FOR NORMAL-HEARING AND HEARING-IMPAIRED LISTENERS

Debi Vickers, Tom Baer, Christian Füllgrabe and Brian Moore, University of Cambridge, UK

For listeners with cochlear dead regions extending from the basal end of the cochlea to the place associated with an edge frequency (fe), amplification of high-frequencies is beneficial only for frequencies up to about 1.7 times the edge frequency (fe) of the dead region (Vickers et al., 2001; Baer et al., 2002) when fe falls below 2kHz. Amplifying frequencies above 1.7fe produces either no benefit or a degradation in speech recognition. The lack of benefit may be due to "overload" of the frequency channel tuned just below fe; this is the channel through which all frequencies falling in the dead region are detected and analysed. With such a limited frequency range available to the listener with a dead region, it might be beneficial to replace some frequencies with information from frequencies well above fe; this might give *extra* benefit, because the information is more independent of that below fe (Müsch and Buus, 2001a,b; Steeneken and Houtgast, 1999).

Four normal-hearing listeners and three listeners with mild-to-moderate losses were used to investigate band-importance functions for combinations of high-frequency bands and lowpass filtered speech. The lowpass cutoffs were 500, 750 and 1000 Hz, simulating dead regions with corresponding cutoffs (fe). Bandpass filtered speech (bandwidth fixed in ERB_N to be equivalent to the range fe-1.7fe.) with variable centre frequency was added to the lowpass band. VCVs were used to measure speech intelligibility.

Intelligibility scores for both the normal-hearing and hearing-impaired groups initially increased as the centre frequency of the added band increased. With further increase, scores either reached an asymptote or began to roll over. When the added band was immediately adjacent to the lowpass reference band, it provided a smaller benefit than any of the other added bands – consistent with the idea that speech information in adjacent bands is correlated (redundant). Also, the benefit of the band adjacent to the lowpass band was smaller for the hearing-impaired than for the normally hearing subjects; indeed, for the former, there was often no benefit at all. Interestingly this result was different to that observed in listeners with dead regions, who make better use of information in the range fe-1.7fe. This is consistent with cortical reorganisation for those with dead regions [Work was supported by the MRC and RNID and the Fyssen Foundation].

Baer T, Moore BCJ, and Kluk K. (2002) J. Acoust Soc Am, 112: 1133-1144
Müsch H, and Buus S. (2001) J. Acoust Soc Am, 109: 2896-2909
Müsch H, and Buus S. (2001) J. Acoust Soc Am, 109: 2910-2920
Steeneken, HJM and Houtgast, T. (1999). *Speech Comm*, 28, 109-123
Vickers DA, Moore BCJ, and Baer T. (2001) J. Acoust Soc Am, 110: 1164-1175

9.10AM COMPANDING TO IMPROVE HEARING AIDS' SPEECH RECOGNITION IN NOISE

Aparajita Bhattacharya and Fan-Gang Zeng, University of California, Irvine

Poor speech perception in noise is a major concern for hearing aid users. Sensorineural hearing loss broadens the auditory filters, thereby degrading the spectral contrast. Amplitude compression across the different channels in a hearing aid also reduces the spectral contrast, further deteriorating the speech perception in noise. We are focusing on spectral enhancement techniques to improve understanding of speech in noise. Recently, Turicchia and Sarpeshkar proposed the 'Companding' strategy (compression followed by expansion), to simulate two-tone suppression, which leads to simultaneous spectral contrast enhancement and multi-channel syllabic compression. Here we implemented the companding architecture to improve hearing aid speech performance in noise.

The incoming signal was divided into 50 channels by a bank of relatively broad band-pass filters. Then, the signal within each channel was subjected to amplitude compression. The compressed signal was then passed through a relatively narrow band-pass filter before being expanded. Finally, the outputs from all the channels were summed to obtain the processed signal. We studied the effects of companding on the acoustic features in both time and frequency domains. To evaluate the performance of the companding strategy, phoneme and sentence recognition experiments were conducted in hearing aid users. The phoneme materials included 12 /hvd/ vowels and 20 /aCa/ consonants spoken by a male and a female speaker. The target sentence material consisted of 250 HINT sentences spoken by a male speaker. Both phonemes and the sentences were presented in quiet and in a steady-state speech-shaped noise at different signal-to-noise ratios.

Preliminary results showed that the implemented strategy significantly improved vowel perception but not consonant perception of the hearing aid users. In addition, the subjects seem to derive small benefits for sentence recognition. The observed differences in the performance with the different types of stimuli will be addressed in view of the spectral and the temporal changes caused by companding. We are also exploring a new strategy that performs both spectral and temporal enhancement and has the potential to improve speech perception of hearing aid users in noise.

Thursday, August 17

SESSION TWO

MODELS

Moderator: Michael Stone

11.10AM AUDITORY PROCESSING MODELS AND THEIR POTENTIAL FOR ENHANCING THE QUALITY OF HEARING TECHNOLOGY

Torsten Dau, Technical University of Denmark, Denmark

The hearing-impaired often experience great difficulty understanding spoken language in the presence of background noise or reverberation. Recent advances in hearing-aid technology have addressed this problem by improving the signal-to-noise ratio of sound delivered to the listener. However, the benefit provided varies greatly. Some listeners do very well with such processing while others continue to experience great difficulty even when their audibility deficit has been compensated for. Clearly, far more than audibility needs to be taken into account for enhancing the signal delivered. The specific nature of an individual's hearing impairment needs to be characterized comprehensively in order to achieve optimum listening benefit in noisy backgrounds.

The first part of this presentation discusses specific psychoacoustic and electrical, evoked-potential correlates of speech intelligibility among listeners with high-frequency hearing loss. Audibility, temporal resolution, frequency selectivity, binaural pitch perception and auditory brainstem evoked potential tests were correlated for each individual with his/her speech reception threshold in noise. In our study, audibility turns out to be a poor predictor of speech intelligibility. Temporal resolution and frequency selectivity are better predictors.

The presentation's second part discusses models of spatio-temporal signal processing. Auditory processing models simulate the transformation of the acoustic signal into its internal auditory representation based on physiological and psychophysical studies. Across-channel correlation models provide a useful perspective with which to examine the temporal processing of speech and other complex signals. Cross-correlation models take advantage of the combination of phase-locked responses and systematic frequency-dependent delays along the cochlea associated with the travelling wave. Processing mechanisms that involve spatio-temporal encoding schemes are likely to be adversely affected by hearing impairment; this is because the phase properties of neural responses are heavily influenced by frequency selectivity. Models for detection and discrimination are of particular interest in our models, as the difficulties of hearing-impaired listeners are most pronounced in noisy environments. Such models may help in the design of future-generation hearing aid technology as well as the development of more accurate diagnostic procedures for treating the hearing impaired.

11.45AM A MODEL APPROACH TO PREDICT THE SPEECH INTELLIGIBILITY IN FLUCTUATING NOISE FOR SIGNALS WITH A NORMAL DYNAMIC RANGE AND FOR SIGNALS WITH DIFFERENT FORMS OF COMPRESSION OF THE DYNAMIC RANGE

Koenraad S. Rhebergen, Niek J. Versfeld, and Wouter A. Dreschler, Academic Medical Center, The Netherlands

The Speech Intelligibility Index (SII) is frequently used to predict the speech intelligibility for speech in a given interfering noise. However, the SII model in its present form is only reliable for speech in stationary noise. Since the SII departs from speech and noise spectra, it does not take into account any fluctuations in the masking noise. Consequently, the model will yield similar SII values, regardless of the degree of fluctuation. In contrast, from many SRT measurements it is clear that normal-hearing listeners can benefit from the fluctuations in the noise.

The present paper gives a description of the Extended SII (ESII) model, which is a novel approach to model Speech Reception Thresholds (SRT) for sentences, both for stationary and for fluctuating noise. The basic principle of this approach is that both speech and noise signal are partitioned into small time frames. The ESII accounts for the dynamics of forward masking by means of a noise-, and hearing-level dependent Forward Masking Function (FMF). Additionally, the ESII can account for compressed speech by means of Intensity Importance Function, that is dependent on the dynamic range of the speech signal.

The contribution of the dynamic range of the speech signal is examined with compressed speech for SRT in stationary and interrupted noise for normal-hearing listeners. In addition, a Wide Dynamic Range Compression scheme was used, as commonly used in hearing aids, to measure the speech intelligibility for compression applied to a mixture of speech in stationary noise or speech in fluctuating noise. The results show for both compression schemes best speech intelligibility with a compression ratio of 2. With the aid of the measured SRT, it will be shown that the ESII model can give a good account for normal and compressed speech.

Thursday, August 17

SESSION THREE

FITTING AND BENEFIT

Moderator: Ole Dyrlund

5.15PM FACTORS AFFECTING THE BENEFIT OF ASYMMETRIC DIRECTIONAL FITTINGS

Ben Hornsby and Todd Ricketts, Vanderbilt Bill Wilkerson Center

Past research has shown that the benefits of binaural listening are sometimes reduced when the signal-to-noise ratio (SNR) between ears differs. For example, when uncorrelated noise (NuS0) is used, increasing the SNR at one ear can turn a binaural MLD condition into an essentially monaural (NmSm) one resulting in an approximately 3 dB increase in threshold. A similar situation may occur when switching from a symmetric directional (directional aids in both ears) to asymmetric (omnidirectional in one ear and directional in the other) fitting. The SNR at the omnidirectional ear is worsened and has the potential to reduce the benefits of binaural processing. We suggest this factor may play a role in recent, and somewhat conflicting, research evaluating the benefits and limitations of asymmetric versus symmetric directional fittings. The actual change in the SNR due to an asymmetric microphone fitting may also be influenced by the spatial configuration of the noise sources and reverberation. By moving the concentration of noise energy from the rear to the front hemisphere, the relative difference in SNRs between omni and directional microphones is reduced.

The purpose of this experiment is to evaluate the effect of worsening the SNR at one ear, via an asymmetric directional fitting, on the benefits of binaural processing (as seen in differences in speech understanding compared to a symmetric directional fitting). Changes in SNR between ears in the asymmetric and symmetric fittings are systematically varied by changing the number and spatial location of the noise loudspeakers. Given that reverberation essentially increases the spatial locations of the noise performance was assessed in both anechoic and reverberant (RT~700ms) environments. The aided speech understanding of sixteen individuals with mild-moderately severe SNHL, in bilateral omnidirectional, bilateral directional and asymmetric (omnidirectional in one ear and directional in the other) modes are being assessed in four (4) different noise source configurations designed to systematically vary the SNR difference between ears in the asymmetric fitting. An analysis of the results will focus on the effects of number and spatial location of noise sources and reverberation on the benefits and limitations of asymmetric directional fittings.

5.50PM BAYESIAN MACHINE LEARNING FOR PERSONALIZATION OF HEARING AID ALGORITHMS

Bert de Vries, GN ReSound Research, The Netherlands and Technical University Eindhoven, The Netherlands; Alexander Ypma and Tjeerd Dijkstra, GN ReSound Research, The Netherlands; Tom Heskes, Radboud University Nijmegen, The Netherlands

Modern hearing aid algorithms contain many tunable parameters. The optimal setting of these parameters depends on the patient's preference function, which is often (partially) unknown. This raises two questions:

1. First, how should we optimize the parameters given partial information about a patient's preferences?
2. And secondly, what questions (e.g. through listening tests) do we ask to efficiently elicit new preference information?

In this paper, we present a coherent probabilistic framework to answer these questions in detail. In particular, following [ChajewskaKollerParr2000] we will derive incremental preference elicitation as a special case of Bayesian machine learning with a specific goal function. We will spell out what it amounts to in the setting of parameter tuning for hearing aid algorithms. Applications include automated fitting of hearing aids, e.g. through a web-based user interface, and on-line personalization of a hearing aid algorithm through learning from a (volume) control wheel.

6.25PM USING GENETIC ALGORITHMS WITH SUBJECTIVE INPUT FOR FITTING AUDITORY DEVICES

Deniz Başkent, Cheryl Eiler, Brent Edwards, Starkey Hearing Research Center

Optimization programs, such as genetic algorithms (GA), have been commonly used for finding the best solutions to complex engineering problems. Conventionally, a search space is defined that includes all possible solutions and the peak of the surface represents the best solution. There is usually a well-defined error measure and the solutions are iteratively modified by minimizing the error.

Optimization programs have also been suggested for finding the best fitting parameters of auditory devices, especially for new and more complex features where the best settings are not known yet. In these applications, however, there is no well-defined error; the only input to the program is the subjective response of the user. Therefore, it is not easy to evaluate the reliability of listener responses or the accuracy of the solutions produced by the program.

In the present study, two experiments were conducted to investigate the feasibility of the GA in fitting auditory devices using only subjective input from the users. The experiments were designed to generate complex listening problems where the best settings were known from previous studies. As a result, the final settings produced by the GA could objectively be evaluated.

In the first experiment, a known search space was created by intentionally distorting speech using a noiseband vocoder. The peak in the search surface was the least distorted condition. The subjects entered their preferences using paired comparisons. The results showed that many subjects were able to provide consistent responses in paired comparisons and the GA was able to produce acceptable solutions. Due to its inherent randomness, however, repeatability was poor. Therefore, when the GA was run multiple times, the probability of finding a good solution increased significantly.

In the second experiment, the GA was used to optimize a new noise reduction algorithm. The best solutions had been determined in a previous study by exhaustive listening. A few subjects preferred significantly different parameters compared to other subjects. However, when all the solutions produced by the GA were pooled and averaged across all subjects, the final solution was similar to the best solutions determined previously with the exhaustive listening.

Overall, the results show that most listeners are able to use the GA to produce a good solution to a listening problem. However, the program might be more valuable for finding the optimal parameters of a new algorithm in research settings rather than finding the best fit in clinical settings.

Friday, August 18

SESSION FOUR

EMERGING TECHNOLOGIES

Moderator: Laurel Christensen

8.00AM THE RECOGNITION OF SPEECH, VOICE AND MUSIC USING COMBINED ACOUSTIC AND ELECTRIC HEARING

Michael F. Dorman, Rene Gifford, Anthony Spahr and Sharon McKarns, Arizona State University

Clinical studies conducted in Europe and in the U.S. indicate that cochlear implant electrodes can be inserted 10-20 mm into the cochlea without destroying residual low-frequency hearing. We have studied the speech, voice and music recognition abilities of three groups of patients who hear with the aid of combined acoustic and electric hearing (EAS): (1) patients with a 20 mm electrode insertion;

(2) patients with a 10 mm electrode insertion and (3) patients with a full electrode insertion who use a hearing aid on the non-implanted ear. Patients in all groups have been found to integrate the information provided by the acoustic and electric stimulation. Performance on tests of speech understanding in noise is especially improved when information from low-frequency acoustic hearing is added to information provided by electric stimulation. Results of a comparison between EAS patients and patients fit with conventional cochlear implants indicate that EAS patients do not perform better than the very best conventional implant patients. However, proportionally there are many more EAS patients who achieve very high scores than conventional implant patients. [Research supported by R01 DC 00654-15].

8.35AM INITIAL RESULTS FROM AN IMPLANTABLE "ROUND WINDOW HEARING AID"

Sigfrid D. Soli, House Ear Institute; Vittorio Colletti, Marco Carner, and L. Colletti, University of Verona, Italy

Initial clinical results will be presented for subjects implanted with a vibratory transducer on the round window. The transducer is a modified version of the Floating Mass Transducer (FMT) manufactured by Med-El. The present criteria for the FMT limiting its application to patients with normal middle ear function have been extended to include patients with ossicular chain defects. Seven patients with severe mixed hearing loss were implanted with the transducer positioned on the round window. All had undergone previous middle ear surgeries with unsuccessful results. Round window implantation bypasses the normal conductive path and provides amplified input to the cochlea. Post-operative aided thresholds of 30 dB HL or less were achieved for most subjects, as compared with unaided thresholds ranging from 60-80 dB HL. Pre-operative speech reception thresholds at 50% intelligibility averaged 85 dB HL, while post-operative thresholds were 50 dB HL. Most subjects reached 100% intelligibility at conversational levels during post-operative testing, while only one subject reached 100% intelligibility pre-operatively. These results suggest that round window implantation may offer a viable treatment option for severe mixed hearing losses that have undergone unsuccessful middle ear surgeries.

Further research with patients exhibiting a wider range of pre-operative thresholds and etiologies is currently underway. In patients with less severe conductive losses and greater sensorineural losses who receive the round window implant, vibratory energy is delivered to the cochlea simultaneously at suprathreshold levels at the round window via the implant and at the oval window via the normal conductive pathway. When these two inputs are delivered with nearly equivalent energy, they potentially can produce interactions with unpredictable effects on basilar membrane motion. For example, when the round window input has the appropriate phase relationship with the normal oval window input, it may serve to amplify basilar membrane motion much as an air conduction hearing aid would be expected to do. However, if the round window input has the inappropriate phase relationship with the oval window input, attenuation of basilar membrane motion may occur. Further research on the nature of these interactions and their practical consequences for hearing will be reported

9.10AM BINAURAL UNMASKING WITH BILATERAL COCHLEAR IMPLANTS

Christopher J. Long, Robert P. Carlyon, MRC Cognition and Brain Sciences Unit, UK; Ruth Y. Litovsky, University of Wisconsin-Madison; Daniel H. Downs, MRC Cognition and Brain Sciences Unit, UK

Nearly 100,000 deaf patients worldwide have had their hearing restored by a cochlear implant (CI) fitted to one ear. However, although many patients understand speech well in quiet, even the most successful experience difficulty in noisy situations. In contrast, normal-hearing (NH) listeners achieve improved speech understanding in noise using two ears. Approximately, 6,000 people have been fitted with bilateral cochlear implants and these devices can potentially aid speech understanding in noise by two types of effect. "Better-ear" effects arise primarily from the enhanced signal-to-noise ratio (SNR) at one ear, and have been reported in a number of studies. In contrast, advantages derived from a fusion of the information in the waveforms at the two ears, although well-established in acoustic hearing, have been more elusive with cochlear implants. Here, we show that this fusion can aid signal detection, and report a Binaural Masking Level Difference (BMLD) for electric hearing.

Four cochlear implant users listened to stimuli containing signals in noise. The input noise was identical on the left and right sides while the signal was either identical across sides, or shifted by pi radians or by 600µsec on one side. Signal-to-noise ratios (SNRs) from -25dB to 20dB were used. Stimuli were half-wave rectified, low-pass filtered, and used to modulate a 1000-pps pulse train; this is analogous to the “transposed” acoustic stimuli used by van de Par and Kohlraush (1997).

All subjects showed a substantial BMLD. In NoSo versus NoSpi, at multiple SNRs, subjects showed consistent advantage in detection in the NoSpi condition. The derived psychometric function showed an average NoS0 threshold of +3dB and an NoSpi threshold of –6dB (a significant 9dB BMLD). With their normal-hearing subjects, van de Par and Kohlraush showed thresholds of 0dB and –16dB, respectively (a 16dB BMLD) in the comparable condition. With NoS600µsec, the cochlear implant subjects demonstrate a threshold intermediate between that of NoS0 and NoSpi.

Tests of implant users and normal-hearing subjects are ongoing to elucidate the mechanisms underlying these effects and the contribution of interaural time and interaural level difference cues. Based on these results, it seems that speech processors which present envelope information alone can provide sufficient information to allow binaural unmasking to enhance detection. We are currently investigating whether this advantage will generalize to supra-threshold tasks such as speech understanding in noise.

[This work was supported by the Royal National Institute for Deaf People and Deafness Research UK.]

Friday, August 18

SESSION FIVE

REHABILITATION & COMMUNICATION

Moderator: Mary Florentine

11.10AM OPTIMIZING COMMUNICATION FOR HEARING AID USERS: A RANDOMIZED CONTROL TRIAL OF A GROUP INTERVENTION PROGRAM

Louise Hickson, The University of Queensland, Australia

Although research evidence indicates the benefits of hearing aid fitting for the majority of those adults with hearing impairment who are fitted, there remain a substantial proportion of clients who discontinue aid use in the long term or who experience ongoing communication difficulties even with amplification. In this paper, the outcomes of a program, Active Communication Education (ACE), designed to address the communication needs of such individuals will be presented. ACE is a group communication education program that runs for two hours per week for five weeks. Outcomes were measured for a total of 178 participants, 96 (54%) of whom had previously been fitted with hearing aids. Participants were randomly allocated to one of two groups. One group (n = 78) undertook a placebo social program for the first five weeks, followed by the ACE program. They were assessed prior to the social program, immediately after it, and then again immediately post-ACE. The other group undertook the ACE program only and were assess pre and post-ACE. In addition, 167 participants were reassessed 6 months after completing ACE. A range of measures were used pre and post-intervention: the Hearing Handicap Questionnaire, the Self-Assessment of Communication, the Quantified Denver Scale of Communicative Function, Ryff Psychological Well-being Scale, and the Short Form-36. In addition, the following measures were used post-intervention: a modified version of the Client Oriented Scale of Improvement and the International Outcome Inventory – Alternative Interventions. The results contribute to the emerging evidence base regarding non-instrumentational intervention and support, and the implications of the findings for the audiological rehabilitation of adults fitted with hearing aids will be discussed.

11.45AM BEYOND AMPLIFICATION: LISTENING AND COMMUNICATION ENHANCEMENT

Robert W. Sweetow, University of California, San Francisco

Hearing-impaired individuals have neural plastic changes along with relearning of sound patterns. Some individuals utilize compensatory strategies that may result in successful hearing aid use. Others, however, are not so fortunate. Modern hearing aids can provide audibility, but may not rectify spectral and temporal resolution, susceptibility to noise, or degradation of cognitive skills associated with aging. Auditory training has long been advocated to enhance communication but has never been time or cost-effective. LACE (Listening and Auditory Communication Enhancement), is a cost effective, home-based, interactive adaptive computer program designed to engage the adult hearing impaired listener in the hearing aid fitting process, provide listening strategies, build confidence, and address cognitive changes characteristic of the aging process. Concepts underlying the development of this therapy will be described. The software will be demonstrated and multi-site validation data will be presented.

Friday, August 18

SESSION SIX

THERE'S MORE TO LISTENING THAN GOES ON IN YOUR EARS

Moderator: Jim Kates

5.15PM INFORMATIONAL MASKING – WHAT IS IT AND HOW DOES IT MATTER IN SENSORINEURAL HEARING LOSS?

Richard L. Freyman, University of Massachusetts

Informational masking in speech recognition describes the condition in which attended (target) speech is difficult to perceptually extract from a complex mixture of a few other competing voices. The definition appears to be quite broad, encompassing features of masking, and of release from masking, that cannot be captured by traditional energetic masking models. Although the majority of work on this topic has been conducted with young normal-hearing listeners, several features of the data may be relevant to hearing-impaired populations and to hearing aids and aural rehabilitation. In particular, binaural hearing and sound localization appear to be extremely useful for overcoming informational masking even in highly reverberant environments. Other factors include benefits from lipreading, bandwidth distinctions between target and masker, and decreases in the uncertainty of the target. Discussion will focus on which of these factors might be most realistically and effectively applied to hearing aid and rehabilitation strategies. [Work Supported by NIH DC01625].

5.50PM THE ROLE OF SHARED ATTENTION IN PERCEPTUAL PROCESSING AND ITS POTENTIAL INFLUENCE ON THE STUDY AND USE OF HEARING-AIDS

Ervin R. Hafter, University of California, Berkeley

Much of the research on attention in sensory detection or discrimination has focused on how performance is affected by imposition of a second, potentially competitive task. With few exceptions, the conclusion has been that processing of the primary task is reduced by the presence of the second task. Given, then, that hearing aids are often worn in complex environments where the listener must attend to other sensory inputs, one must question the efficacy of designing and testing new auditory prostheses without concern for the role of shared attention. This talk will include a brief overview of attention as well as a discussion of attempts in our laboratory to study the effects of attention as they interact with such signal-processing algorithms as noise-reduction.

6.25PM COGNITIVE AND AUDITORY FACTORS UNDERLYING SPATIAL ATTENTION IN OLDER ADULTS

Gurjit Singh, Kathy Pichora-Fuller and Bruce Schneider, University of Toronto, Canada

Many older adults have trouble understanding conversations in everyday noisy listening situations. The difficulties of older adults could arise from declines in cognitive processing (e.g., attention) or auditory processing (e.g., binaural comparisons). Two paradigms have been used recently to explore attention in binaural listening situations. In one paradigm, a target utterance is presented from one spatial location and competing utterances are presented from two different locations. Depending on the condition, the listener may take advantage of cues specifying either the identity or the location of the target (e.g., Kidd et al., 2006, *JASA, 118*, 3804-15). In a similar paradigm, a target utterance is presented from one perceived spatial location and a competing utterance is presented from either the same or different perceived location. The perceived location is achieved using the precedence effect (e.g., Freyman et al., 1999, *JASA, 106*, 3578-88). Common to both paradigms, word recognition is measured in conditions varying in attentional demands. The strength of the first paradigm is that one can assess the relative contributions to word recognition arising from information about target identity and location. The strength of the second paradigm is that acoustical differences arising from actual spatial separation are largely eliminated as a factor contributing to word recognition. In this study, the strengths of each paradigm are combined. In Experiment 1 a target and two competitor utterances are presented using actual spatial separation, and in Experiment 2 they are presented using perceived spatial separation. Young and old listeners with good audiograms in the speech range were tested in both experiments. By comparing age groups in Experiment 1, we determine the relative contributions arising from information about target identity and location to age-related differences in word recognition performance.

By comparing the results of Experiments 1 and 2, we determine the extent to which benefit from separation is governed by acoustical or attentional factors. Whereas Experiment 1 sheds light on the cognitive underpinnings of auditory spatial attention, Experiment 2 further specifies to what extent auditory factors may influence age-related differences in understanding speech in complex environments.

Saturday, August 19

SESSION SEVEN

LEARNING AND TRAINING

Moderator: Pam Souza

8.00AM HUMAN DISCRIMINATION LEARNING ON BASIC AUDITORY TASKS

Beverly A. Wright, Northwestern University

Human listeners can learn to discriminate between sounds that are initially indistinguishable. To better understand the nature of this learning, we have been using behavioral techniques to examine training-induced improvements on basic auditory discrimination tasks. In this talk, I will describe how multiple-hour training differentially affects the discrimination of sound frequency, intensity, location, and duration, how learning on a given discrimination condition generalizes, or fails to generalize, to untrained discrimination conditions, and how different training regimes can either enhance or degrade learning and generalization. I will discuss how these data contribute to our understanding of the mechanisms underlying performance on particular trained tasks, provide insights into the neurobiology of learning and memory, and inform the development of therapeutic training schemes. [Supported by NIH.]

8.35AM LEARNING OPTIMAL GAIN IN REAL WORLD ENVIRONMENTS

Sepp Chalupper and Heike Heuermann, Siemens Audiological Engineering Group, Germany

While prescriptive formulas provide a valid starting point for hearing aid fitting, they do not account for individual loudness preferences. Moreover, preferred gain in real world listening situations cannot be determined in a clinical setting. These shortcomings result in patient complaints about inappropriate loudness settings. An effective way to address these issues is with a trainable hearing aid. Such a hearing aid is able to actually learn and automatically apply what the wearer teaches it. The principle of trainable hearing aids has been investigated scientifically in the past few years by several researchers with chest-worn prototype devices. Now, hearing aids with a "Learning Volume Control (LVC)" are commercially available. These hearing aids are able to learn the preferred volume setting in various listening situations. Thus, it is possible to investigate this concept with more subjects, as not all patients are willing to wear bulky devices with cables in everyday life. Research questions include: How long does it take to achieve an optimal gain setting? Are patients able to "teach" different gain settings for different situations? Is it better to couple volume and program adjustments in a bilateral fitting or should preferred gain be learned independently for both ears? Does a learning VC support acclimatization or do subjects simply reduce gain? What percentage of patients with comfortable loudness in the clinic require different gain in real world situations? Do experienced users of hearing aids with analog VCs appreciate this new volume control behaviour?

Three different studies with hearing impaired listeners have been conducted to investigate these questions. The results indicate that after one to two weeks, volume can be optimized by a learning VC for different situations such that only minor (< 3dB) adjustments are required. The program specific gain settings as applied by manufacturer's FirstFit are in good agreement with the average learned volume. However, the individually preferred volume varies significantly (from -6 dB to +9 dB). Also, large differences between program specific gains were observed across individuals. Without bilateral synchronization of hearing aids, loudness differences between left and right ear were reported by subjects. Only 25% of subjects with comfortable loudness in the clinic changed the volume in their everyday environment by less than 3 dB. This shows that even for patients who do not have any complaints about loudness in the clinic, an optimization of gain in real world environments may be beneficial.

9.10AM THE EFFECTS OF WORD-BASED TRAINING ON AIDED SPEECH RECOGNITION AND IDENTIFICATION

Larry E. Humes, Matthew H. Burk and Lauren E. Strauser, Indiana University

Older adults with impaired hearing often delay arrival to the hearing clinic and pursuit of amplification for as many as 12-15 years from when they first experience difficulties in speech communication. For those who do ultimately pursue amplification, well-fit devices can lead to instant restoration of previously inaudible speech sounds. Is it reasonable to assume, however, that their measured speech recognition with amplification, soon after fit, is optimal or at asymptote? Given the long period of time during which the high-frequency portions of speech have been rendered inaudible, can older adults immediately re-learn to use this restored auditory information or is time and training required to optimize performance?

We have been addressing these general questions in a series of experiments completed over the past several months. In these experiments, we have been exploring a word-based approach to improving aided speech-recognition performance in noise in older adults, typically using sets of 50-75 words in training with trial-to-trial orthographic feedback. A series of experiments has been conducted to examine the impact of the training on the trained words, both spoken by talkers used in training (familiar talkers) and other talkers not used in training (unfamiliar talkers), on sets of 50-75 novel words not heard during training (with both familiar and unfamiliar talkers), and on sentences (comprised of both trained and untrained words). Results thus far have shown promising improvements over relatively short periods of training with mean improvements of 10-15% from pre-test to post-test for trained and untrained words, in open-set and closed-set response formats, strong talker generalization, and some generalization to sentences, although the effects for sentences are smaller and more variable across listeners. Additional experiments, currently underway, are making use of a larger set of trained words (150 – 600 words), all with high frequency of occurrence in spoken English, and explore the use of "frequent phrases" as training materials as well. In addition to the presentation of an overview of the preceding series of experiments, the group and individual data for these ongoing experiments will be presented. (Work supported, in part, by NIA.)

Saturday, August 19

SESSION EIGHT

COMPRESSION

Moderator: Wouter Dreschler

11.10AM QUANTIFYING THE EFFECTS OF FAST-ACTING DYNAMIC RANGE COMPRESSION ON THE ENVELOPE OF SPEECH

Michael A. Stone and Brian C. J. Moore, University of Cambridge, UK

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. Performance measures such as attack and release times (ANSI, 1996), and effective compression ratio (Braida *et al*, 1982) define the technical performance of a compressor. However, the resulting parameters have little value for predicting efficacy although Kates and Arehart (2005) have reported a method for predicting the resulting performance with real-world signals such as speech.

Stone and Moore (2004) identified one mechanism by which speech intelligibility may be degraded when independent sound sources become 'comodulated' within a compression channel by the application of a common gain signal from a compressor; this tends to promote perceptual fusion of the sources. With multi-channel compression, cues for across-frequency grouping of components from each source are also degraded. Finally, compression can modify the shape of the temporal envelope of the signal, and not just its modulation depth. We propose three measures to describe the within- and across-channel modifications to the envelope produced by compression systems when a mixture of signals is present:

(1) across-source modulation coherence (ASMC) is a measure of comodulation between sources caused by changes in compressor gain.
(2) within-source modulation coherence (WSMC) is a measure of the preservation of across-frequency envelope cues within a single source between system input and output.
(3) fidelity of envelope shape (FES) is a measure of the accuracy with which the envelope shape of a source in different frequency channels is preserved between system input and output, independent of the degree of compression intended.

Results from speech intelligibility tests using single- and multi-channel compression suggest that ASMC and WSMC are the best predictors of performance, with FES playing a lesser role.

References
ANSI (1996). ANSI S3.22-1996. Specification of hearing aid characteristics.
Braida LD, Durlach NI, De Gennaro, Peterson PM, Bustamante D (1982) Vanderbilt h. aid report
Kates JM, Arehart KH (2005) J Acoust Soc Am 117:2224-2237.
Stone MA, Moore BCJ (2004) J Acoust Soc Am 116: 2311-23.

11.45AM FAST-ACTING COMPRESSORS CHANGE THE EFFECTIVE SIGNAL-TO-NOISE RATIO - BOTH UPWARDS AND DOWNWARDS!

Graham Naylor, René Burmand Johannesson and Thomas Lunner, Oticon Research Centre, Denmark

Traditionally, amplitude compression in hearing aids has been specified in terms of an input-output curve. It is now generally recognised that real-life input signals will from moment to moment experience gain values which deviate widely from the specification.

This presentation goes a step further, to consider how the relationship between mixed non-steady signals is affected by fast-acting compression. This is important for two reasons:
- Most signals which hearing aids are exposed to are neither steady-state nor present in isolation.
- Speech-in-noise tests which establish an SNR for criterion performance, rather than a performance at fixed SNR, may place the system under test in widely differing modes of operation.

First we will define some concepts concerning SNR for mixtures of non-steady signals, including the distinction between conventional long-term SNR and momentary SNR.

Basic mechanisms in a compressor which act to alter the relationship between Signal and Noise will be explained and illustrated, along with the concept of 'Output SNR' . Output SNR is the ratio of the average long-term levels of signal and noise respectively, measured at the output of the compressor. We use the demixing method suggested by Hagerman & Olofsson (2004) to derive Output SNR for real hearing-aid compressors.

A very significant property of fast-acting compression systems is that the Output SNR of a signal and noise mixture will generally be different from the Input SNR. The difference can amount to several dB, and can be in either direction. The size and direction of the difference is dependent on the modulation characteristics of the Signal and Noise, the Input SNR, and on compression settings.

This result has critical implications:

- The choice of input signal characteristics and mixture SNR can strongly affect the conclusions of any study involving fast-acting compression.
- Speech intelligibility tests, especially adaptive ones, may give misleading results, depending on the SNR region they operate in. Dependencies here include baseline performance of individual listeners, which means that individual listeners comparing alternative amplification systems may experience contrasts having ‘opposite sign’.

We will also describe some pitfalls to avoid when evaluating SNR in single-channel and multi-channel systems.

Hagerman B, Olofsson Å. 2004. A method to measure the effect of noise reduction algorithms using simultaneous speech and noise. ACUSTICA - acta acustica 90(2), 356-361.

SESSION NINE

SIGNAL PROCESSING

Moderator: Larry Humes

5.15PM THE ACOUSTIC AND PERCEPTUAL EFFECTS OF NOISE-SUPPRESSION ALGORITHMS

Justin A. Zakis and Christi Wise, Dynamic Hearing Pty Ltd., Australia

Noise generated by hearing-aid circuits can be audible to aid users, and may lead to the rejection of hearing aids. Internal noise may become a greater problem as aid usage among people with a mild hearing loss increases with open-ear fittings. Two signal-processing algorithms that expansively suppressed internal circuit noise and low-level environmental noise were objectively and subjectively evaluated. The single-channel algorithm suppressed sounds below a configurable A-weighted expansion threshold (45 dBA recommended to maintain speech intelligibility). The aim was to relate the degree of noise suppression to the loudness of the noise. For the multiple-channel algorithm, the expansion threshold was different for each channel, and the thresholds were shaped to follow the long-term average spectrum of speech presented at 55 dB SPL. Thus, the multiple-channel algorithm was designed to maximize the suppression of noise without affecting the intelligibility of speech spoken at casual levels. With the recommended settings in static conditions, the single-channel algorithm provided a greater reduction in the level of internal noise at the output of the aid, and was perceived as quieter by most normal-hearing participants. However, in dynamic conditions 'pumping' noises were more noticeable with the single-channel algorithm than the multiple-channel algorithm. For impaired-hearing listeners fitted with the ADRO® amplification strategy, speech perception scores were 99.3% correct for words in sentences presented at 55 dB SPL in quiet, with or without either noise-suppression algorithm. Mean sentence reception thresholds in quiet without noise suppression and with the single- and multiple-channel algorithms were 39.4, 40.7, and 41.8 dB SPL, respectively. The increase in the mean sentence reception threshold was significant for the multiple-channel algorithm, but not for the single-channel algorithm. Thus, both algorithms suppressed noise without affecting the intelligibility of speech presented at 55 dB SPL, while the single-channel algorithm provided marginally greater noise suppression in static conditions, and the multiple-channel algorithm avoided 'pumping' noises in dynamic conditions.

5.50PM PERCEPTUAL AND THEORETICAL EVALUATION OF THE INTERAURAL WIENER FILTER (IWF) ALGORITHM WITH RESPECT TO SPEECH RECEPTION THRESHOLDS

T.J. Klasen, S. Doclo, and M. Moonen, SISTA-SCD, KULeuven, Belgium;
T. Van den Bogaert, and J. Wouters, ExpORL, KULeuven, Belgium

With increasing processing and communication power, hearing aids are evolving into true binaural processors. This evolution opens new possibilities for better noise reduction techniques, through more microphones, and for preserving interaural cues, specifically interaural time differences (ITDs) and interaural level differences (ILDs), which are important for auditory scene analysis and thus for speech perception in noise.

A binaural multi-channel Wiener filtering algorithm that preserves the interaural transfer functions of the speech and noise components was presented in [1]. By extending the underlying cost function to incorporate terms for the interaural transfer functions (ITFs) of the speech and noise components, weights can be used to control the emphasis on the preservation of the ITFs in addition to the emphasis on noise reduction. Adapting these parameters allows one to preserve the ITFs of the speech and noise components, and therefore ITD and ILD cues, while enhancing the signal-to-noise ratio. These results have been verified through simulations.

The focus of this submission is to investigate this algorithm perceptually. Speech reception threshold (SRT) tests with normal hearing subjects under headphones will be carried out with different values of control parameters and in different noise scenarios. The base case for our comparison will be the binaural multi-channel Wiener filtering algorithm without the additional ITF terms, which was presented in [2]. The results of these tests will show a gain in SRT with respect to ITF preservation and noise reduction.

[1] Klasen T.J., Doclo S., Van den Bogaert T., Moonen M., Wouters J., ``Binaural multi-channel Wiener filtering for hearing aids: Preserving interaural time and level differences", accepted for publication in *IEEE International Conference on Acoustics, Speech and Signal Processing (ICASSP)*.

[2] Klasen T.J., Van den Bogaert T., Moonen M., Wouters J., ``Preservation of interaural time delay for binaural hearing aids through multi-channel Wiener filtering based noise reduction", in *Proc. of the 2005 IEEE International Conference on Acoustics, Speech and Signal Processing (ICASSP)*, Philadelphia, Pennsylvania, Mar. 2005, pp. Vol. III 29-32.

6.25PM SPEECH QUALITY EVALUATION OF FEEDBACK CANCELLATION TECHNOLOGIES IN HEARING AIDS

Michael Wirtzfeld and Vijay Parsa, National Centre for Audiology, Canada

Most current generation hearing instruments incorporate digital signal processing techniques for active phase cancellation of the feedback signal. Previous studies have looked at the performance of the feedback cancellation algorithms in terms of Maximum Stable Gain (MSG), Added Stable Gain (ASG), Power Concentration Ratio (PCR) etc. In this project (a work in progress), we are investigating the impact of the feedback cancellation algorithms on speech quality during sub-oscillatory and oscillatory feedback stages. High-end BTE hearing aids from Bernafon, Oticon, Phonak, Siemens, and Sonic Innovations are used in this study. The hearing aids are programmed for a flat hearing loss and are placed on the Bruel & Kjaer head and torso simulator (HATS), which is also equipped with a telephone handset positioner. A generic cellular phone is placed in the handset positioner and its position relative to the ear is varied to simulate feedback path changes. Speech recordings are obtained at different feedback stages and with the feedback canceller in the hearing aid turned on/off under two conditions : (a) static condition where male and female speech samples are played back through a loudspeaker in front of the HATS and recorded through the ear microphone; and (b) active condition where speech samples are played through the cellular phone and recorded through the ear microphone. In both these conditions, speech quality is measured using validated instrumental measures such as the Perceptual Evaluation of Speech Quality (PESQ) and Bayesian model based speech quality estimation. The results from this study will help quantify the relative performance of the hearing aid feedback cancellation technologies in preserving the speech quality and will aid in the development of novel feedback cancellation strategies. [Work supported by the Oticon Foundation and the Natural Sciences and Engineering Research Council of Canada].

Sunday, August 20

SESSION TEN

SOUND QUALITY

Moderator: Brent Edwards

8.00AM THE EFFECT OF BANDWIDTH ON SOUND QUALITY IN LISTENERS WITH NORMAL TO MODERATE HEARING LOSS

Todd A. Ricketts, Vanderbilt Bill Wilkerson Center for Otolaryngology and Communication Sciences

Surveys have shown that poor sound quality continues to be a common complaint of hearing aid wearers. One of the most important factors related to sound quality is audible bandwidth. Considerable work suggests that listeners with normal hearing prefer the sound quality of signals with wide audible bandwidth. While work to date in adults with impaired hearing supports extension of low frequencies, extension of the high frequencies beyond 4-6 kHz has not been supported. To date studies examining the effect of high frequency bandwidth extension on sound quality have generally treated those with hearing loss as a homogenous group, however it seems given the preferences of listeners with normal hearing that some listeners with less hearing loss may also prefer more high frequency extension.

The purpose of this study was to determine if preference for high frequency bandwidth extension for signals processed using two different multi-channel compression schemes could be predicted based on individual hearing thresholds of listeners with normal to moderate hearing loss. The cut off frequencies were selected to represent a range of bandwidths that are potentially achievable in modern hearing aids (5.5 kHz, 9 kHz and 12 kHz). Three different monaurally presented sound segments (music and a movie clip) served as the test stimuli. For each of the three stimuli, round robin paired comparisons of sound quality were made on 8 repetitions of all possible comparisons (2 compressors X 3 bandwidths). Frequency shaping was provided for the hearing-impaired subjects. Preliminary results revealed a significant preference for the 9 kHz or 5.5 kHz cut off frequencies that was highly predictable based on threshold information. Interestingly, bandwidth preference was not predictable based on PTA hearing thresholds, but most notably, 12 kHz thresholds. No significant preference differences were noted between the 9 kHz and 12 kHz cut off frequencies. [This work was supported by the Dan Maddox Hearing Aid foundation and Great Nordic Research Group].

8.35AM COMPARISON OF OBJECTIVE AND SUBJECTIVE MEASURES FOR SOUND QUALITY AND SPEECH INTELLIGIBILITY IN NONLINEAR HEARING INSTRUMENTS

Stefan Fredelake, Inga Holube and Martin Hansen, University of Applied Sciences Oldenburg/Ostfriesland/Wilhelmshaven, Germany

Modern nonlinear hearing instruments including dynamic compression and noise reduction schemes are impossible to be described by standardized measurement procedures using test signals like sinusoids or stationary noises. Therefore, several objective measures like e.g. the effective compression ratio (CR_{eff}), the phase-locked modulation transfer function (MTF_{PL}) (Holube et al., IHCON 2004), the S/N-improvement from noise reduction algorithms described by Hagerman and Olofsson (Acta Acustica 2004), the SDR described by Olofsson and Hansen (IHCON 2002) and the PEMO described by Huber (PhD thesis 2003) were used to analyze different hearing instrument signal processing schemes. Except for the PEMO, the objective measures were calculated as the weighted average across frequency and – if applicable – across modulation frequency, thus resulting in a single number being compared to subjective ratings.

A number of 27 different simulated compression settings with varying number of compression channels (2, 8 and 19), compression ratios (2:1, 4:1 and 8:1) and release time constants (10ms, 100ms and 1000ms) were implemented and used for processing speech in quiet. In addition, a noise reduction scheme for hearing instruments was simulated for speech in noise. The input S/N and the settings of the noise reduction algorithm were changed systematically. The output of both algorithms was presented to 12 normal hearing and 11 hearing-impaired subjects. Paired comparison methods were used to derive subjective sound quality, speech intelligibility and all-over preference ratings. Speech intelligibility was also assessed by using the Oldenburg sentence test for speech audiometry.

The results of the objective measures of hearing instrument performance were compared to the subjective ratings. A monotonic relation was observed between the MTF_{PL} for both signal processing algorithms and the subjective preference rank scale. The relation between the MTF_{PL} and the preference rank scale for noise reduction showed a dependency on the input S/N. In addition, MTF_{PL}, CR_{eff} and S/N-improvement according to Hagerman and Olofsson are carrying equivalent information. Consequently, the subjective perception of sound quality is predictable for both used algorithms by any one of the objective measures. The predictive power of the different objective measures for describing the subjective measures of sound quality and speech intelligibility in nonlinear hearing instruments will be presented and discussed.

[This work was supported by AGIP, HörTech and Acousticon]

Sunday, August 20

SESSION ELEVEN

OWN VOICE AND WHAT HAVE WE LEARNED?

Moderator: Brent Edwards

9.30AM EFFECTS OF AMPLIFICATION VARIATIONS ON VOCAL LEVEL AND OWN-VOICE SOUND QUALITY OF HEARING-AID USERS

Søren Laugesen, Claus Nielsen, Patrick Maas, and Niels Søgaard Jensen, Oticon Research Centre, Denmark

Evidence collected over the last five years indicates that hearing-aid users have several own-voice issues – even when their eventual occlusion problems have been essentially solved by open fittings. Examples of issues are: Sound quality of own voice, level control, the ability to speak and hear at the same time, and the ability to use whispering. Presently, our focus of research is on level control, that is, the ability to produce adequate vocal level for various everyday occasions. This ability depends on three feedback mechanisms: auditory feedback, proprioception, and visual/verbal feedback from others. Of these, auditory feedback is affected by hearing loss and hearing-aid treatment, whereas the others remain intact. Thus, it is of interest how hearing-aid treatment affects auditory feedback and, in turn, the hearing-aid user's ability to control vocal level.

Basically, auditory feedback is affected in two ways by the hearing aid. First, by the hearing aid's amplification. Secondly, because the occlusion effect provides "amplification" of own voice, which may exceed the hearing aid's amplification, the self-perceived ratio between own speech and background noise may also be affected. These effects have been studied in two separate pilot experiments.

In this presentation, the emphasis will be on the results from the experiment focusing on amplification. Here, the task of the test subject was to address a listener across different distances and with different gain settings of an experimental hearing aid. The test subject was either instructed to do as usual (unsupervised), or to speak at the level, which the listener found adequate (supervised). Vocal output was recorded and the test subject rated either the self-perceived own-voice level or sound quality.

In the supervised condition, self-reported level and quality of own voice varied with hearing-aid gain according to expectations. In the unsupervised condition, however, a surprising divergence in the strategy employed by test subjects emerged. Some test subjects apparently depended entirely on their auditory feedback mechanism, with the result that vocal output varied dramatically amongst gain settings, while self-reported level ratings were “adequate” in all trials. The other test subjects relied more on proprioception and (largely) ignored auditory feedback. Thus, they (largely) produced adequate vocal output with all amplification settings, but showed considerable variation in their level ratings.

Implications of these findings for clinical practice will be discussed.

10.05AM IHCON 2006: THEMES, CONSENSUS, AND DIVERGENT THINKING

Harvey Dillon, National Acoustic Laboratories, Australia

This paper, which will be the last paper of the conference, will review and appraise the oral and poster presentations of IHCON 2006. The paper will draw out themes linking different presentations, identify areas that have been widely studied and contrast these with areas that appear to have been little studied over the preceding two years. Research findings that appear to be reaching a consensus will be highlighted. Divergent findings will also be noted and the reasons for them speculated upon. Implications for future research directions will be drawn out. Where research has reached a suitable consensus, implications for practice and for policy will be suggested. It is difficult to be more precise about this paper in advance of the findings that it will review. The paper has the potential to contribute significantly to the conference, because of the multiple and complex connections, and potential synergies, between the diverse high-quality papers on the wide range of disciplines that are traditionally brought to IHCON.

Poster Program

Posters for Session A should be put up by 8 A.M. Thursday, August 17, and taken down after 10 P.M. Thursday, August 17, or before 7 A.M. Friday, August 18. Presenters should be at their posters from 9:45 – 11:00 A.M.; 4:30 - 5:00 P.M.

POSTER SESSION A

Thursday 8:00AM –10:00PM

A1

Vibrant Soundbridge® clinical investigations: Expanding indications for use

Jane M. Opie, Geoffrey Ball, Markus Huetter, Peter Grasso, Samia Labassi, Nadia Giarbini, Peter Lampacher, VIBRANT MED-EL Hearing Technologies GmbH, Austria

Purpose/Material: The Vibrant Soundbridge is a partially implantable "direct-drive" hearing system for the treatment of hearing loss and is currently indicated for use in adults who have mild-to-severe sensorineural hearing loss. Recently, the device has been applied to persons with mixed and conductive hearing losses to provide amplification to residual sensorineural hearing. In order to appropriately place the device in disordered and malformed ears, the manner and/or location of placement of the device is altered, and, in some cases, the device is used in conjunction with commercially available, passive middle ear prostheses. The objective of these studies is to evaluate expanding indications for use of the Vibrant Soundbridge to include persons with conductive and mixed hearing losses.

Methods: Subjects were implanted with the Vibrant Soundbridge implantable hearing aid, using either the Vibroplasty or Round Window Vibroplasty surgical technique. A single-subjects, repeated measures design is used to evaluate the safety and efficacy of the Vibrant Soundbridge in persons with conductive and mixed hearing losses.

Results: An overview of the study and its procedures, as well as preliminary results, will be presented.

A2

The objective benefit of a "temporal fitting enhancing" procedure: A long term study

Matthias Latzel, Siemens Audiological Engineering Group, Germany; Juergen Kiessling, Sabine Margolf-Hackl and Jan Denkert, ENT Hospital of University of Giessen, Germany

The calculating of the target gain for fitting of hearing aids is basically focused on the optimization of the spontaneous acceptance and/or the sound quality. This is mostly in conflict with the core audiological requirement: the improvement of speech intelligibility. To compensate this mismatch the acoustician uses a kind of "temporal fitting enhancing" procedure which readjusts the "First-Fit" setting step by step during a certain time period (e.g. 6 month) so that the audiological requirements are finally achieved. The modifications of the hearing aid settings during the period of "temporal fitting enhancement" are based mostly on the individual experiences of the acoustician - a general validated procedure does not exist. Therefore, it is possible that the hearing aid setting is not audiologically optimized.

To investigate the efficiency of a "temporal fitting enhancing" procedure which is furthermore influenced by the acclimatization effect a long term study (period of 2 years) was initiated by the German Center of Competence HörTech.

16 subjects (first time users) with sensorineural hearing loss participated in the study. They were binaurally fitted with a four channel digital hearing instrument. The subjects were subdivided within 2 groups: 8 subjects performed a prescribed "temporal fitting enhancing" procedure comprising a stepwise increase of the compression and/or channel gain. The other 8 subjects performed the "temporal fitting enhancing" procedure individually by fine-tuning their hearing aids according to their individual needs and expectations using a structured interactive fitting method.

During the periodical visits, a couple of audiological test procedures were conducted to examine the status of the hearing loss and the progress of the hearing aid fitting: pure-tone audiogram, loudness scaling, speech test in quiet, (spatial) speech test in noise, questionnaires and coupler measurements.

The results of the study demonstrate that a hearing aid fitting after performing a "First-Fit" and a fine-tuning session is mostly not finished. The maximal audiological benefit especially an improved speech perception is achieved much later. However, the results do not show clearly whether this observed effect is caused by the systematic change of the gain settings of the hearing instrument, by the acclimatization effect, or by a combination of both.

The results show an advantage for the individual "temporal fitting enhancing" procedure, which leads to the conclusion that for the "temporal fitting enhancing" procedure an individual method should be preferred to the prescribed approach.

A3

Gain limits of mini-BTE hearing aids

Peggy Nelson and Melanie Gregan, University of Minnesota; Rupa Balachandran, California State University; Dianne Van Tasell, University of Arizona

Within the past few years most hearing aid manufacturers have introduced small, cosmetically appealing hearing aids designed to fit behind, or over, the ear. Some of these devices are coupled acoustically to the ear canal via a small-diameter tube that is retained by a device designed to produce as little ear canal occlusion as possible (an "open" fitting). Other devices are coupled electronically to the ear canal via a wire connection to a receiver module designed to fit into the canal. Although all the "mini-BTEs" achieve a similar cosmetic result, they can be expected to differ significantly in terms of achievable gain before feedback because of the way the hearing aid is coupled to the ear. The devices investigated were several mini-BTE hearing aids that use thin-tube acoustic coupling to the open ear canal, and a third mini-BTE hearing aid with the receiver sealed deeply into the ear canal. Real-ear occlusion response (REOR) was measured for all devices, as well as maximum stable gain for flat and for sloping hearing loss configurations. Preliminary results suggest that the hearing aid with deep-fitting sealed receiver could produce 20-40 dB additional real-ear aided gain (REAG) below 4000 Hz when compared with the thin-tube open-fit mini-BTEs. Large intersubject variability with the deep-fitting receiver device is related to the tightness of the acoustic seal. Relationships between REOR and maximum gain will be reported. Work supported by SeboTek Hearing Systems.

A4

Speech-like test signals for hearing instruments

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Electroacoustical measurement methods as defined in standards like, e.g., IEC 60118 are mainly applicable or targeted to linear or single-channel compression hearing instruments. The performance of modern hearing instruments however, containing multi-channel, signal-adaptive processing of the incoming sound, is not or only very poorly reflected by the measurement results of the current standards. Also it is a severe drawback that artificial test signals are used during the measurement process like sinusoids and stationary noises. These methods are applicable for quality control purposes but are not very suitable to describe the performance of hearing instruments in daily life when listening to speech. Signal processing algorithms like, e.g., input/output functions with multiple segments and kneepoints, variable time constants, noise reduction or signal-classification dependent gain, result in deviating frequency-dependent output levels for different input signals. In order to characterize the performance of those hearing instruments, several measurement equipment manufacturers have implemented artificial or natural speech signals to be applied during testing. Unfortunately, results achieved with different test signals cannot be compared to each other. The European Hearing Instrument Manufacturer Association (EHIMA) has taken the initiative to propose a newly to be made speech-like test signal and measurement procedure for standardization that should mimic properties of multi-lingual speech. Our contribution was firstly to make a detailed analysis of existing test signals as used in test boxes and by the telecommunication industry in comparison to real speech for different languages with respect to their spectra, modulation spectra, level distributions and comodulations across frequency bands. Secondly, based on the outcomes of this analysis, a speech-like test signal was developed bearing closest similarity to speech but being non-intelligible. The properties of this new test signal are demonstrated and discussed.

[This work was supported by EHIMA]

A5

Combinations of monaural and binaural feedback control algorithms increase added stable gain

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The acoustic gain provided by hearing aids is limited by the feedback signal from the receiver to the microphones. Digital feedback control algorithms have significantly increased stability at high gains. However, a further increase of gain and stability is still desirable.

In this study, several combinations of monaural feedback cancellation algorithms known from literature and binaural algorithms (e.g. binaural de-reverberation schemes, Wittkop and Hohmann, 2003) have been implemented on a low-delay real-time system (HoerTech Master Hearing Aid, Grimm et al., 2006). The short delay of the processing system and the possibility to use BTE hearing aid dummies made realistic tests of the maximum stable gain in the hearing aid possible. In addition, perceptual quality of music and speech signals as a function of the gain was estimated using a perceptual quality model (PEMO-Q, Huber, 2003).

The results show that combinations of monaural feedback cancellation algorithms with binaural de-reverberation schemes can significantly increase the stable gain, while the audio quality of the processed signal is tolerable. Because the binaural algorithms do not adapt to the feedback path, they are found to be more robust against changes in the feedback path than the adaptive filter approaches pursued in current hearing aids. The increase in stable gain observed here could be of practical relevance.

Work supported by HörTech (BMBF 01EZ0212, subproject 'Algorithms'), and by grants from the European Union FP6, Project 004171 HEARCOM.

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A6

Development of a metric for quantifying the temporal envelope of speech

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We are working on a metric for quantifying hearing-aid processing-related alterations of the relevant temporal cues in the speech signal. Our previous work (e.g., Jenstad & Souza, JSLHR, 2005; Jenstad & Souza, In Review) has shown a negative correlation between temporal envelope alteration and speech intelligibility for listeners with hearing loss; that is, as alteration of the temporal envelope increased, speech intelligibility decreased by a small but significant amount. This relationship was found using a broad method of quantifying the temporal envelope; namely, the Envelope Difference Index (EDI; Fortune, Woodruff, & Preves, Ear Hear, 1994). The EDI measurement, as originally proposed, is limited in that it combines all frequencies for the entire duration of the speech signal into a single index quantifying temporal envelope alteration. However, recent evidence shows that, depending on the listening conditions, listeners place greater reliance on the temporal envelope in some frequency bands than others (Apoux & Bacon, JASA, 2004). In addition, the amount of information in adjacent frequency bands may be redundant, leading to a need for a combined weighting method that accounts for redundancy across bands (Steeneken & Houtgast, Sp Comm, 2002). Finally, it is also well known that the temporal information is of greater importance for recognition of some phonemes than others (such as stops vs. vowels) and for some segments of the speech signal than for others (such as transitions vs. steady-state segments). Because of all these factors, further refinement of the EDI is needed to make it a frequency-specific and segment-specific tool.

We propose that further refinement of the EDI tool can improve its utility in predicting speech recognition for amplification processing schemes. Using previously-collected data of recordings of hearing-aid processed speech and speech intelligibility results for 25 listeners with hearing loss, we will present acoustic measures of the recordings using modified and refined versions of the EDI measurement. The EDI parameters under investigation include measurement of the temporal envelope in octave and 1/3-octave bands using different weighting functions to combine the envelopes across the bands. In addition, we are measuring specific phonemes and speech segments that are known to be affected by changing the temporal envelope rather than entire speech signals. The study goal is to determine which parameters can explain the greatest amount of variance in the behavioural speech intelligibility data. Our long-term goal for this work is to provide guidelines for the maximum amount of tolerable alteration to the temporal envelope, independent of the type of hearing aid processing applied to the speech signal.

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A7

Effect of speech presentation level on acceptance of noise in listeners with normal and impaired hearing

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Acceptable noise level (ANL) is a measure of willingness to listen to speech in the presence of background noise. ANLs are typically measured at one speech presentation level, the listener's most comfortable listening level (MCL) (called conventional ANL). ANL research indicates that conventional ANL is not affected by hearing sensitivity. ANLs have also been measured at fixed speech presentation levels in listeners with normal hearing; this research indicates that ANLs are dependent on speech presentation level in the normal hearing population. Specifically, as speech presentation increases, ANLs also increase in listeners with normal hearing. Furthermore, the effects of speech presentation level have been studied by measuring the slope of the ANL function at fixed speech presentation levels (i.e., ANL growth). Results of these ANLs studies have shown that listeners with large ANLs have larger ANL growth than listeners with small ANLs for the normal hearing population. Furthermore, it is well known that the auditory system in listeners with sensorineural hearing loss behaves differently than the normal ear. For example, listeners with sensorineural hearing loss have a reduced dynamic range and loudness recruitment, which causes abnormal loudness level growth in these listeners. It could, therefore, be hypothesized that ANL growth will be significantly affected in ears with sensorineural hearing loss. ANL growth in individuals with normal hearing and hearing loss have, however, not been compared. Therefore, the purpose of the present study was to determine the effects of speech presentation level on acceptance of noise in listeners with normal and impaired hearing.

ANLs were measured at speech presentation levels of 40, 45, 50, 55, 60, 65, 70, and 75 dB HL for 24 listeners with normal hearing and 46 listeners with impaired hearing. The listeners were matched for conventional ANL (i.e., ANL at MCL). The effect of speech presentation level on ANL was evaluated in two ways: (i) averaging ANL values at all speech presentation levels (called global ANL) and (ii) calculating the slope of the

ANL function across the fixed speech presentation levels (called ANL growth). Results demonstrated that neither global ANLs nor ANL growth were related to hearing sensitivity. Additionally, none of the ANL measures were related to pure tone average for listeners with impaired hearing. These results indicate that the effects of presentation level on acceptance of noise are not related to hearing sensitivity.

A8

Measuring and predicting quality ratings of fast-acting single-microphone noise reduction

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Fast-acting single microphone noise reduction (SMNR) is known to yield sound-quality improvements in some speech-in-noise conditions. It is unclear whether or not there is any interaction between such improvements and the degree of hearing loss, and the degree to which the interaction might depend upon noise conditions or noise reduction settings. The current study is an initial attempt at answering these questions.

Normal-hearing subjects and subjects with high-frequency sloping hearing losses of mild, moderate, or moderate-severe degree listened to speech in noise processed with several different algorithms and algorithm settings. Algorithms under test included an advanced algorithm from the SMNR literature and a simplified version of this algorithm (SA). Listeners provided quality ratings in an A/B (or "paired comparison") paradigm, indicating which processing yielded higher perceived quality and by how much. This was done for a sentence in noise (either car interior or cafeteria) at 0, 5, and 15 dB SNRs.

The current implementation of the SA algorithm was ranked better than unprocessed, and often better than its more advanced version. Initial results, fit with a Bradley-Terry-Luce choice model, indicate that there is no statistically significant difference in rank order of the settings of the SA algorithm, across noise condition and across normal, mild-loss, and moderate-loss listener, after it has been through several rounds of refinement with respect to quality judgments. Moderate-severe listeners, while generating rankings during refinements that were similar to those of the other groups, often indicated an inability to hear differences across algorithms. These results indicate that the setting of the SA may be held constant across conditions and loss type and still maintain high relative quality. Attempts to model the choice data using computational analyses of the test signals will be presented.

A9

Spatial benefit of hearing aids

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Individuals with hearing loss generally benefit less than listeners with normal hearing when speech and noise sources are separated in space. One explanation for this deficit is that speech information that could be made audible due to an improved signal-to-noise ratio at higher frequencies remains inaudible due to high-frequency hearing loss. Providing amplification should improve spatial benefit by restoring high-frequency cues. However, given recent evidence (Hogan & Turner, 1998; Ching et al., 1998) suggesting that speech recognition of hearing-impaired listeners may not improve with increased high-frequency audibility, aided spatial-separation benefit may be less than expected. Further, because interaural level and time difference cues also contribute to the benefit of spatial separation, differences between a listener's two hearing aids could reduce aided spatial benefit by altering these cues. Here, the benefit attributable to spatial separation was measured as a function of low-pass cutoff frequency of speech and babble with and without binaural amplification. Listeners were older adults with cochlear hearing loss. Speech levels corresponding to 50% correct recognition of sentences from the Hearing in Noise Test were measured in a 65-dB SPL multi-talker babble with two loudspeaker configurations: (1) sentences and babble at 0^{o} azimuth (in front of the listener) and (2) sentences at 0^{o} azimuth and babble at 90^{o} azimuth (at the listener's side). In addition, subjects' willingness to accept background noise with and without amplification was measured in the same two loudspeaker configurations using a procedure developed by Nabelek et al. (1991). Speech recognition in noise and acceptable noise level significantly improved when aided and when speech and babble were spatially separated. Furthermore, as low-pass-filter cutoff frequency of speech and noise increased from 2.0 to 4.0 kHz, spatial benefit of hearing aids (i.e., aided vs. unaided) increased significantly, but did not differ significantly as cutoff frequency increased further to 6.0 kHz. [Work supported by NIH/NIDCD]

A10

Intervention for restricted dynamic range and reduced sound tolerance: Clinical trial update

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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 for reduced 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 on electrophysiological measures. We will present interim results and selected examples of positive treatment effects. Supported by NIH R01DC04678.

A11

The computerized revised token test: Assessing the impact of hearing loss on auditory language processing in older adults

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The Revised Token Test (McNeil & Prescott, 1978, RTT) is a well-established test of auditory language processing disorders for use with persons with aphasia and other auditory processing deficits. A computerized version of the test, the Computerized Revised Token Test (C-RTT), was developed recently in our laboratory to better control stimuli presentation, and eliminates variability associated with the complex scoring requirements of the RTT. In the current study the impact of participant hearing loss, amplification, and acoustic signal intensity on performance of non-brain injured older adults on the C-RTT was assessed.

The participants consisted of two groups of older adults; one group with moderate-severe bilateral sensorineural hearing loss, the other group with normal hearing. The participants (ages 59-79) were community dwelling adults without history or evidence of speech-language impairment or brain injury. Their speech perception skills were consistent with their hearing status, and based on a battery of electrophysiological tests their central auditory pathways were considered largely intact. In order to eliminate auditory deprivation as a factor, all of the hearing-impaired participants were previous hearing aid users.

The fifty-five item version of the C-RTT was administered to all of the participants in soundfield across a range of intensity levels in order to construct performance-intensity functions. The hearing-impaired participants were administered the C-RTT with and without their own personal hearing aids. The normal hearing participants were administered the C-RTT in quiet and with a simulated hearing loss. The simulated hearing loss was produced with frequency-shaped noise and confirmed with elevated hearing thresholds consistent with the hearing-impaired group. The participants' overall and subtest responses were scored automatically and online. In addition, efficiency scores were derived, taking response time into account.

Despite more shallow PI functions, the participants with hearing loss produced maximum overall scores that were similar to those of the normal hearing subjects, although some subtest differences were observed. Overall scores for the participants with hearing loss were slightly lower at the lower presentation levels when they were wearing their hearing aids than when they were unaided, but with increased signal intensity there was little difference between the aided and unaided conditions. The normal hearing subjects evidenced a steeper PI function in the quiet condition than in the hearing loss simulation condition but their maximum overall scores were similar across their two listening conditions. However, their efficiency scores proved to be poorer in the quiet condition than in the hearing-loss-simulation condition. In contrast the listeners with hearing loss showed similar efficiency scores across listening conditions. One explanation for the efficiency score differences observed with the normal hearing participants is that processing language in noise required more focused attention than in the quiet, which may have been associated with quicker responses.

The results of this study are largely consistent with previous work on hearing loss in older adults, indicating that audibility plays a substantive role in auditory processing of language in older adults.

A12

Impact of frequency compression on listeners with high frequency hearing loss

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Conventional amplification cannot make all frequencies of speech audible to listeners with moderate to severe hearing losses in the high frequencies. It is possible that these listeners do not hear high frequency phonemes such as /s/, /sh/, and /f/, even when fitted with modern hearing instruments. Inaudibility of these phonemes is more likely for low-level speech inputs, and may be an issue of greater importance for children for developmental reasons.

Recently, a new technology has been developed that allows hearing instruments to compress a high frequency into a narrower bandwidth. The low frequency channel remains unmodified in the frequency domain, and multichannel compression/amplification are also applied. For speech understanding, this type of processing may be beneficial for hearing and understanding high frequency sounds. The effects of this technology have not yet been evaluated in children.

The current study aims to offer further insight into the prospect of using frequency compression technology to improve speech recognition scores for children, and how children's benefit compares with adults' benefit with the same technology. Participants in this study use hearing instruments programmed in one of two ways: conventional and frequency compression. Frequency compression settings are individually determined. A four-week acclimatization period takes place before laboratory outcome measures are completed. Speech recognition, loudness rating, and self-report questionnaires are used as outcome measures in both the conventional and frequency compression settings. The final stage of the study consists of programming both conventional and frequency compression in different programs in the hearing instrument (experimenter and participant blinded). Outcome measures are completed at the end of four weeks. Data collection for the study is ongoing, therefore progress to date will be presented.

A13

Assessing the effectiveness of feedback cancellers in hearing aids

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Feedback cancellation systems (FBC's) have matured within the hearing aid industry over the past ten years to the point where every major manufacturer has an implementation in at least one of their products. At IHCON 2004, Freed and Soli reported on performance benchmarks for products from a variety of manufactures and they showed dramatic differences across products. This article presents more extensive benchmarking paradigm that objectively quantifies FBC performance by measuring Maximum and Added Stable Gain (MSG & ASG), entrainment (artifacts in response to periodic signals), and robustness to acoustic path changes. These metrics are used to compare the performance of six feedback cancellation systems on the market today.

The MSG of the device in KEMAR's ear has been estimated as function of the frequency from measured acoustic impulse responses of the hearing aid system in different gain settings. Results show that the ASG often depends on the lowest frequency at which the FBC operates. FBC's that operate only at higher frequencies will offer very limited ASG.

Sensitivity to entrainment has quantified by recording the output of the device while real-world signals (music, speech, machine noise) play. Subsequently the entrainment level has been calculated using the loudness and sharpness of the difference output between FBC-off and FBC–on. Results show that three of the six hearing aids evaluated demonstrated objectionable entrainment artifacts.

The robustness to acoustic path changes has been assessed by recording the output of the device while an acoustic reflector mounted on a linear slide was moved quickly to KEMAR's ear. The "psychoacoustic annoyance" of the recorded output was calculated and the results show that three out of the six devices had clear artifacts.

A14

Do new hearing aid users prefer less low-frequency, high-frequency, or overall gain than experienced users?

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Software tools to help new users of hearing aids get used to their amplification have gained widespread acceptance in clinical audiology. Although these tools may vary in their implementation, they all reduce gain for the initial fitting from the prescribed targets. However, a recent literature review (Convery et al., 2005) found that the acclimatization effect has not been scientifically verified and that no studies have directly investigated acclimatization with appropriate control of all relevant parameters. On this basis a study was designed that aimed at determining 1) whether gain preferences of new and experienced hearing aid users differ overall, or if they differ only in the low or the high frequencies; 2) if gain preferences initially differ, at what point post-fitting do the gain preferences of these groups converge; and 3) if preferred gain does change post-fitting, is it related to changes in perceived loudness and does it affect speech recognition in noise performance.

Sixty new hearing aid users and twenty-five experienced hearing aid users were fitted with the same type of hearing instruments set with NAL-NL1, NAL-NL1 with a 6 dB high-frequency cut, and NAL-NL1 with a 6 dB low-frequency cut in three programs. Participants compared the listening programs in their everyday environments and determined their preferred listening program and volume control setting. Inexperienced subjects were monitored 3 weeks, 3 months and 12 months post-fitting; experienced subjects were monitored 3 weeks post-fitting. At each appointment, subjects also completed a gain preference test, a loudness perception test, and a speech recognition in noise test. Preliminary results suggest that 1) new hearing aid users prefer progressively less gain than prescribed by NAL-NL1 as the hearing loss becomes more severe; 2) when matching the audiometric profile of experienced and new users, new users prefer less than 2 dB less gain than experienced users at 3 weeks post-fitting; 3) there is no significant change in preferred gain, the level of sound rated "comfortable", or speech recognition performance over the first 12 months in new hearing aid users; and 4) about 55% of both new and experienced users prefer the program with a high-frequency cut. The clinical implications of data are discussed.

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A15

Hearing aid algorithm and hardware module based on a general purpose DSP

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We proposed a multi channel hearing aid algorithm and digital hearing aid module using one of the general purpose DSPs and verified the performance of the algorithm and the hardware module. The algorithm is comprised of adaptive feedback cancellation (AFC) and multi channel amplification. In hearing aid, feedback signal, called howling, is very annoying problem to the hearing-aid users and it limits the maximum gain of hearing-aid. To remove this feedback signal effectively, we developed the adaptive feedback cancellation algorithm which based on both normalized least mean square (NLMS) method and time-varying all-pass filter in the forward path to de-correlate input and output signals of the hearing aid. The multi channel amplification algorithm consists of transform part and compensation part. In transform part, input signal at time domain was convert into signal of frequency domain using modified discrete cosine transform (MDCT) method and then divided into 64bands signal considering critical band. In compensation part, wide dynamic range compression (WDRC) which make the signals fit to reduced dynamic range of hearing impaired person, automatic gain control (AGC), amplification and output compression were applied to the divided signals regarding hearing loss characteristics. And we developed hardware module, which mainly consists of DSP (TMS320C6711), codec (AIC23), SDRAM (8Mbyte) and flash memory (512Kbyte). The performance of the algorithm was verified by three kinds of tests. First, the effectiveness of algorithm was verified through computer simulation. Second, we ported the algorithm code to the developed hardware module for verifying real-time operation. Finally, subjective score test which listeners estimate the quality of the processed signal was done. It is verified that the developed algorithm can be a good hearing aid algorithm and the developed hearing aid module can be a hardware platform for estimating many hearing aid algorithms.

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A16

Effects of interaural time differences on stream segregation in normal-hearing and hearing-impaired listeners

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

In a recent overview of factors influencing the auditory system's ability to segregate sequentially occurring sounds into separate auditory streams, Moore and Gockel (2002) suggested that stream segregation may occur as soon as any perceptual difference between successive sounds is sufficiently salient. The present study was designed to investigate the influence of perceived spatial location – as manipulated by the sounds' interaural time differences (ITDs) – on certain aspects of auditory scene analysis in normal-hearing and hearing-impaired listeners.

In the first part of the study, obligatory sequential stream segregation for a group of

young normal-hearing listeners was assessed using a temporal discrimination task (Roberts et al., 2002). Participants listened to pairs of tone sequences, both composed of the same (AAAA… or BBBB…) or different alternating (ABAB… or BABA…) low-frequency pure tones, and were required to detect the sequence containing an anisochrony. Stream segregation was also investigated by means of a subjective measure; here, participants listened to sequences of repeated A and/or B tones triplets (AAA-AAA-…, BBB-BBB-…, ABA-ABA-…, BAB-BAB-…) and reported whether they perceived a single "galloping" (i.e., integrated) stream or two simultaneous (i.e., segregated) streams. In both experiments, successive tones of each of the four possible sequences were presented with identical but opposite ITDs of 0 to 2 ms. The introduction of an ITD discrepancy between successive tones consistently affected anisochrony detection, but less than for a spectral discrepancy. In contrast, the effect of ITD discrepancy on the subjective streaming measure was more idiosyncratic. Taken together, these data are consistent with the idea that perceived spatial location may be a sufficient but weak (obligatory) streaming cue.

The second part of this study (in-progress) aims to test the hypothesis that impaired speech intelligibility in "cocktail-party" situations, as observed in hearing-impaired listeners, may be partly due to impaired ITD-induced streaming capacities. Sensitivity to ITD discrepancies (0-1 ms) in a group of elderly listeners and age-matched normal-hearing controls was measured for (i) anisochrony detection and (ii) speech-in-noise identification. Preliminary results indicate that hearing-impaired listeners are indeed less sensitive to ITD cues in the temporal-discrimination task, but scores overlap with those of the control listeners. Results in terms of correlation between the two tasks will be discussed at the meeting.

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A17

A study of hearing aid gain functions based on a nonlinear nonlocal feedforward cochlea model

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A model based sound amplification method is proposed and studied to enhance the ability of the hearing impaired.

The model consists of mechanical equations on basilar membrane (BM) and outer hair cell (OHC). The OHC is described by a nonlinear nonlocal feed-forward model. In addition, a perceptive correction is defined to account for the lumped effect of higher level auditory processing, motivated by the intelligibility function of the hearing impaired. The gain functions are computed by matching the impaired model output to the perceptively weighted normal output, and qualitative agreement is achieved with NAL-NL1 prescription on clean signals. For noisy signals, an adaptive gain strategy is proposed based on the signal to noise ratios (SNR) computed by the model. The adaptive gain functions provide less gain as SNRs decrease so that the intelligibility can be higher with the adaptivity.

[This work was supported by NSF and NIH]

A18

Growth-of-masking functions for sinusoidal and complex-tone maskers with differing phase spectra measured in hearing-impaired and normally hearing listeners

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The slopes of growth-of-masking (GOM) functions, which are plots of masked threshold against masker level, vary with the presence or absence of sensorineural hearing loss, and also with the type of masker used. When measured in forward masking, these slopes are often less than unity for normally hearing listeners, and the slope is generally shallower for waveforms with a higher crest factor. Such differences with crest factor are often reduced or absent with sensorineural hearing loss. This is consistent with an explanation that they are due to cochlear compression and suppression.

Models of forward masking that employ a linear temporal integrator predict that in the absence of cochlear compression, GOM functions in forward masking should have slopes of unity. In the present study, we tested this idea by measuring GOM functions using normally hearing (NH) and hearing-impaired (HI) listeners. GOM functions in forward masking (0-ms masker-signal delay) were measured using as maskers complex tones (harmonics 1-40, 100-Hz fundamental frequency) with components starting in cosine or random phase, and on-frequency sinusoids. The signal was a 20-ms sinusoid with frequency of 1 or 2 kHz for all listeners except one, for whom the higher frequency was 1.5 kHz. For the NH listeners and some of the HI listeners, slopes of the GOM functions were significantly greater for the random- than for the cosine-phase maskers and slopes for the complex-tone maskers were less than for the sinusoidal maskers. These effects are explained in terms of cochlear compression and suppression. Some of the HI listeners showed no phase effect. For them, the slopes of the GOM functions were similar for the complex-tone and sinusoidal maskers. These effects are explained by the loss of cochlear compression and suppression. The GOM functions for the sinusoidal maskers had slopes between 0.45 and 0.78 and were typically in the range 0.6 to 0.7. The finding of slopes below one for listeners in whom peripheral compression was probably absent is not consistent with linear-integrator models of forward masking.

A19

Analysis of two simulated side-branched compressor strategies and their impact on sound quality perception as a function of hearing impairment

Andrew Dittberner, GN Auditory Research Lab; Todd Ricketts, Vanderbilt University; Jeff Bondy, GN Auditory Research Lab

From the perspective of the hearing impaired end user, compressors ensure audibility while keeping the signal level comfortable. To do this compressors in hearing instruments trade-off between processing complexity, frequency resolution, time delay, quantization noise, and distortion. While one compressor design can optimize intelligibility the same settings may worsen music appreciation or sound quality. In this work, an investigation was conducted on two types of compressors to explore their impact on perceived sound quality of music as a function of compressor type and hearing impairment.

Three groups of ten adult subjects were used: 1) normal hearing, 2) mild/moderate sensorineural hearing impaired, and 3) moderate sensorineural hearing impairment. Sound quality was evaluated using a round-robin, paired-wise comparison technique. Only the criterion of "overall sound quality" was evaluated. Paired comparisons were made separately for each of three recorded sound samples. These included short segments of Bluegrass music (Alison Krauss' Lucky One), Classical music (Mozart's Flute Concerto No. 2 in D) and a popular movie (Seabiscuit – including both voice and a ringing musical triangle). All segments were specifically chosen due to their inclusion of high frequency content. All signals included relatively high signal levels through at least 11,000 Hz.

Preliminary findings indicate a compressor preference between the different listener groups. Further, analysis revealed different distortion levels introduced by the compressors that were functions of frequency.

A20

New measurement procedures for characterizing speech amplification of hearing aids

Nikolai Bisgaard, Brian Dam Petersen, Johnny Andersen, Frank Rosenberger, Carsten Paludan-Møller, Volker Kühnel, Ivo Merks, ISMADHA working group of EHIMA, Denmark

Current pure-tone based hearing aid measurement procedures provide very limited information about how modern hearing aids with sophisticated signal processing amplify speech signals. Many hearing aids can be programmed into a special test-mode in order to be measured according to current standards. Such measurements will provide useful information about the basic electro-acoustic properties, but reveal little about the main objective for a hearing aid is performed. An industry expert group has been established to propose a new set of measurement procedures to be proposed as international standards. A related project concerns the development of standardized speech signal to be used with the proposed measurement procedures.

The proposed measurement procedures are based on 60 seconds of speech input after 20 seconds of settling. Signals are presented under quasi-free field conditions compensated for microphone positions at a level of 65 dB SPL. The hearing aid output is recorded in an ear-simulator and analyzed in 1/3 octave bands determining the short-time levels using 125msec time-windows. Amplitude statistics such as the 99 % and the 30 % percentiles as well as the average values are computed and presented as either REIG or REAR.

A21

Effect of simulated and actual age on F_0 DL and concurrent vowel identification

Kathy Pichora-Fuller, Tracy Anselmo, and Tara Vongpaisal, University of Toronto, Canada

We investigated the effect of age on voice fundamental frequency (F_0) difference limen (DL) and identification of concurrently presented vowels. Fifteen younger and 15 older adults with normal audiometric thresholds in the speech range participated in two experiments using intact stimuli. In Experiment 1, F_0 DLs were measured for a synthesized vowel. Younger adults had smaller (better) F_0 DLs than older adults. For the older group, age was significantly correlated with F_0 DLs. In Experiment 2, accuracy in identifying concurrently presented vowel pairs was measured. Vowel pairs were formed from five synthesized vowels with F_0 separations ranging from 0 to 4 semitones. Younger adults identified concurrent vowels more accurately than older adults. When the vowels in the pairs had different formants, both age groups benefited similarly from F_0 separation. When both constituent vowels had identical formants, F_0 separation was deleterious, especially for older adults. The pattern of errors indicates that the relative dominance of specific vowels was similar for both age groups. For both groups, there were no significant correlations between pure-tone average threshold and either F_0DL or accuracy in concurrent vowel identification. Neither were there significant correlations between F_0DL and accuracy in single or concurrent vowel identification. Age-related auditory declines, including larger F_0DLs, poorer identification of concurrently spoken vowels, and more difficulty with F_0 separation between identical vowels, may involve deficits in periodicity coding. Note that he pattern of reductions in performance due to age are not the same as the reductions found by others in cases of sensori-neural hearing loss.

Experiments 3 and 4 were identical to Experiments 1 and 2, respectively, except that jittered stimuli were presented. The same vowel tokens used in the earlier experiments were jittered to simulate the loss of synchrony or periodicity coding that we believe characterizes some forms of auditory aging. Each token was jittered to create a set of jittered tokens that were selected randomly for presentation to ensure that the listeners did not learn the specific distorted tokens. When listening to jittered stimuli, the performance of both age groups dropped, with performance of the younger adults dropping to a level that was below that observed for older adults in intact conditions. These findings bolster the interpretation that loss of periodicity or synchrony coding could explain some aspects of auditory aging.

A22

Continuous optimization of amplification and noise reduction in hearing aids using the Speech Intelligibility Index (SII) theory

Martin Hansen, University of Applied Sciences, Oldenburg, Germany; Carsten Paludan-Müller, Widex A/S, Denmark

Most adaptive noise reduction systems in hearing aids are controlled solely by the properties of the input sound, e.g. typically its sound level or its level distribution, and the resulting level dependent gain or amount of noise reduction have been optimized for one or more “average” acoustical situations. However, in situations where the speech sounds reproduced by the hearing aid are on or close to the edge of audibility for the hearing aid user, noise reduction techniques controlled by input levels may make speech inaudible and thereby have a negative impact on speech intelligibility.

A new noise reduction system will be presented, that results in a reduction of noise and at the same time yields an optimal improvement of audibility of speech signals. This goal is reached by including the individual hearing loss as a key factor and an integrated part of the noise reduction algorithm. In effect, the action of the new noise reduction algorithm is optimized by using the SII (ANSI S3.5-1997). The optimization is performed continuously, based on the individual hearing thresholds and running estimates of the noise spectrum and the speech spectrum, by adjusting the frequency dependent gain such that the SII is always maximized. The optimization is performed by an n-dimensional restricted non-linear maximum search in an n-dimensional gain-space. Compared to generic fitting rules which prescribe a level dependent gain (or amount of noise reduction) and which have been optimized for one or more “average” acoustical situations, our new approach yields a precisely and continuously optimized gain which takes into account both the hearing thresholds and the current speech and noise spectra which are individual to the user and its current acoustical surrounding. The net result is improved speech intelligibility without a compromise in the perceived sound quality. One particular and noteworthy observation is, that the overall audibility of speech can sometimes be improved, depending on the input signal and noise configuration, by reducing the gain in certain frequency bands. This can be explained by effects of upward-spread of masking which the SII takes into account.

In a laboratory test with 12 hearing impaired subjects the SII optimized noise reduction showed an average improvement of 1.8 dB SNR for speech recognition in noise, compared to no noise reduction. In a field test comparison we found that subjects generally preferred the SII optimized strategy to the input-controlled strategy. In an open questionnaire subjects reported that they had experienced a reduction of the perceived noise level while speech intelligibility was preserved or increased.

A23

Moving from acoustic scene analysis to auditory scene analysis with digital hearing aids

David Fabry and Stefan Launer, Phonak Hearing Systems

Modern hearing aids use increasingly sophisticated technology to adapt automatically to their environments. Recent examples include multi-band compression, adaptive directional microphones, noise cancellation, and active feedback cancellation. All are first approximations of “acoustic scene analysis”, which comprises a classification and decision-making process that is capable of recognizing a wide variety of sound environments to adapt the hearing aid response accordingly. The human brain, on the other hand, uses much higher level processing to segregate multiple sounds into auditory "streams" for outstanding speech recognition in difficult listening environments.

This session will focus on discussion of recent experiments conducted to develop a four-class acoustic scene classification system used in a commercially available hearing instrument.

In the evaluation phase, ROC curves were generated to find the classification system with the highest specificity and sensitivity.

Subsequently, human subjects were evaluated, and data reflect the need for personally-based optimization of hearing aid characteristics by varying the time constants of multichannel compression, adaptive directionality, and scene classification. Data from ongoing experiments will be presented that provides examples of how “self learning” feature parameters may be incorporated into modern digital hearing aids to evolve from “acoustic” scene analysis to “auditory” scene analysis.

A24

Vibration interference in ossicular chain due to different sound delivery paths of implantable middle ear hearing device

Jin-Ho Cho, IL-Yong Park, Young-Ho Yoon, Young-Jun Choi, Sang-Heun Lee, Kyungpook National University, South Korea

The implantable middle ear hearing devices (IMEHDs) which can resolve the problems of conventional air-conduction hearing aids are mainly composed of three parts of a vibration transducer, a signal processing module, and microphone. The external sound inputted into the microphone is amplified and filtered by signal processing module so that the vibration transducer attached on an incus long process can drives the ossicular chain directly according to the amplified electrical signal. In addition, the external sound transmission through the ordinary path from an ear canal to a tympanic membrane also happens in the middle ear of IMEHDs implanted. Such condition can lead the vibration interference due to the different time delays in two sound delivery paths when the vibration of a stapes by the ordinary sound delivery through the ear canal is competitive to that by the attached transducer with the low amplification gain.

In this paper, the vibration interference effect in the case of IMEHD implantation has been investigated using the ACRHS system, a fully-implantable middle ear hearing device (F-IMEHD), being developed in Korea as well as the computer program for simulating two different sound transmission paths. Also, we suggest the method of avoiding the loss of sound delivery caused by the interference effect with the experimental results by the use of human fresh temporal bones [This study was supported by a grant of the Korea Health 21 R&D Project, Ministry of Health & Welfare, Republic of Korea. (02-PJ3-PG6-EV10-0001)].

A25

The surprising lack of effect of added AM on low-rate FM detection at high carrier frequencies

Hugh Greenish, Michael A Stone and Brian CJ Moore, University of Cambridge, UK

Two mechanisms have been proposed for the detection of frequency modulation (FM). The first is through variations in the excitation pattern on the basilar membrane (Zwicker 1956; Moore and Sek, 1994) and the second is through variations in the temporal patterns of firing in the auditory nerve, i.e. variations in phase locking (Siebert, 1970). It is important to understand the role of phase locking, since recent evidence suggests that hearing-impaired people have a reduced ability to use phase-locking information.

Moore and Sek (1996) measured FM detection thresholds in the presence or absence of a amplitude modulation (AM) with a 6-dB peak-to-valley ratio, for modulation rates between 2 and 20 Hz, using a sinusoidal carrier presented at 70 dB SPL. The AM was intended to disrupt excitation-based cues, so that good performance would only occur if phase locking could be used to detect the FM. The hypothesis was that, as modulation rate increased, the 'sluggishness' of the system for decoding phase locking would reduce the reliability of this cue. Thus, the added AM should produce a greater disruption of FM detection for the high modulation-rate conditions than for low modulation-rate ones. A further prediction was that, for test frequencies above 4-5 kHz, where phase locking information was no longer available, the detrimental effect of the AM would be the same for all modulation rates. Their results were consistent with these hypotheses, although AM did interfere with detection at modulation rates as low as 2 Hz, suggesting that the detection mechanism was not purely based on phase locking at any modulation rate.

The current study extends this research by measuring low-rate FM detection limens in normally hearing subjects at low sensation levels, using carrier frequencies up to 10 kHz. Thresholds were measured in five different conditions. The first two replicated the high sensation level conditions used by Moore and Sek, while in the remaining three conditions the carrier was presented at 20 dB SL. In the first of the low level conditions, the carrier envelope was unmodulated; in the second there was a 10 dB sinusoidal AM superimposed; and for the final condition the envelope was modulated by a noise band centred on 2 Hz, with a peak-to-peak modulation depth of 40 dB. While the results at high sensation levels resemble those of Moore and Sek (1996), the effect of the added AM on performance at low sensation levels appears to be independent of carrier frequency even for very high frequencies and in the presence of a large amount of AM. Our results suggest that 10-dB depth of sinusoidal AM provides maximum excitation pattern disruption, with further increases of modulation depth giving no further impairment of performance [Supported by Deafness Research UK]

A26

Using a signal cancellation technique with impulse responses to assess adaptive directivity of hearing aids

Yu-Hsiang Wu and Ruth A. Bentler, University of Iowa

The method described in ANSI S3.35 for quantifying directivity of a microphone calls for presenting "probe" signals from a fixed speaker to a device under test (DUT) that rotates to different azimuths. This method is invalid for adaptive null steering directional systems because the probe itself changes the directional pattern. Our lab has adapted the ANSI method to better assess adaptive directivity. This new method (the SC method) combines a signal cancellation technique with a second speaker. The speaker rotates with the DUT throughout the test sequence and maintains a constant position relative to the DUT. The speaker emits a "jammer" signal which freezes the directional pattern. The measurement at each azimuth is obtained by two sequential recordings from the DUT, one using an input of a low-intensity probe (usually noise with a certain duration) and a high-intensity jammer (from the fixed and rotating speakers, respectively), and the other with an input of the same probe and a phase-inverted jammer. After canceling out the jammers, the remaining probe can be used to assess directivity. The SC method has been proven to be accurate and reliable. However, because a high-intensity jammer is needed to freeze the directional response pattern, the SC method cannot assess the directivity in conditions wherein jammer levels are low (for example, the response of an adaptive system to a fan noise behind it). In order to assess the directivity at any jammer level, we propose a new method using the signal cancellation scheme with impulse responses (IR method). The IR method is the same as the SC method, except that clicks are used as probe signals. The measurement is achieved by two inputs: one using a click and a jammer, and the other using the same click and a phase-inverted jammer. After canceling the jammer, the DUT response to the click (e.g., impulse response) can be used to quantify the directivity. Because of the short duration of clicks, the DUT responses can be captured almost before the directivity patterns change. By using the IR method, even directivity under silent environments can be assessed. With appropriate levels and bandwidths of jammers, important properties of multi-channel adaptive directional systems, such as threshold of null steering, bandwidth of multi-channel system, and so on, can be measured.

A27

Restoring localization for users of bimodal hearing instruments by introducing cross-frequency ILD cues

Tom Francart, Jan Wouters and Marc Moonen, Exp ORL, K.U.Leuven, Belgium

Users of certain configurations of bilateral hearing instruments, where high frequencies are absent in one or both ears, have difficulties using interaural level difference (ILD) cues for sound source localization. This is due to the fact that low frequency ILDs are absent for sound sources further away than about 1 meter.

Though it has been shown that humans are able to perceive ILDs in the low frequencies, bimodal system users possibly experience a difference in place of stimulation between both ears, because the residual hearing of most subjects doesn't extend beyond 500Hz and the electrode array of the cochlear implant is not fully inserted in the cochlea.

To investigate the possibility of perception of interaural level differences (ILD) by users of contralateral bimodal systems (cochlear implant in the one ear and hearing aid in the other) and bilateral cochlear implants, experiments were conducted with normal hearing subjects to assess the impact of a frequency shift in one ear on the just notable difference in ILD. Similar experiments were conducted with users of bimodal systems.

Stimuli were noise bands with a bandwidth of 1/3oct, the center frequency was shifted in one ear over 1/3, 2/3 or 1oct, relative to the center frequency of the noise band in the other ear. Noise bands were uncorrelated between both ears. An adaptive procedure (1up/2down) was used to find the just notable difference (JND) in ILD. First a reference was presented, equally loud on both ears (using dBA weighting); then the stimulus was presented with an ILD. As changes in frequency mapping often induce a significant learning effect, several runs were conducted of each condition, and it was verified that every subject was at saturation performance. The subject had to respond whether the stimulus was on the left or right side of the reference.

From the results, it is clear that subjects can still perceive an ILD, even if the frequency shift is as big as 1 octave, albeit less accurately. For the reference condition without frequency shift, JNDs are in the order of 2dB, depending on the base frequency. For the shifted conditions (1/3oct, 2/3oct, 1oct), JNDs are in the order of 3dB, 4dB and 5dB.

Preliminary results with users of bimodal systems suggest that they are able to perceive ILDs, although less accurately than normal hearing listeners. The results will be presented.

A28

Role of spectrally asynchronous auditory delays in speech perception for normal-hearing and hearing-impaired listeners

Amanda J. Ortmann, University of Pittsburgh and VA Pittsburgh Healthcare System; Catherine V. Palmer, University of Pittsburgh; Sheila R. Pratt, University of Pittsburgh and VA Pittsburgh Healthcare System

Recently, there has been a growing literature regarding the effects of auditory delay on speech perception. Current digital hearing aid technology introduces a delay between the arrival of the speech signal at the hearing aid microphone (the ear) and the delivery of the signal to the ear canal. The introduction of more complex signal processing that is aimed at improving speech perception and comfort continually introduces ever increasing delay. Hearing aid digital delay is characterized as being a spectrally-asynchronous acoustic delay. Spectrally asynchronous delay is defined by acoustic delay values that vary as a function of frequency bands within the speech spectrum. Despite the body of literature regarding auditory delay, it remains unclear whether hearing-impaired listeners are detrimentally affected by the introduction of a spectrally asynchronous delay to the speech signal.

The purpose of this study is to examine the effect spectrally asynchronous auditory delay has on the perception of voice and voiceless initial stop consonants in individuals with normal hearing and individuals with sensorineural hearing loss. A possible cue that could be used in differentiating voice and voiceless cognate pairs is Envelope Onset Asynchrony (EOA). EOA is defined as the time asynchrony between the onset of two frequency bands of energy (one being high passed at 3000 Hz and the other being low passed at 350 Hz). It is not known whether normal-hearing and hearing-impaired listeners use EOA as a cue for distinguishing the voicing features of speech. This study involves a categorical perception procedure to find the categorical boundary of voice-voiceless cognate pairs as the EOA is altered along a continuum.

Twenty normal-hearing adults between the ages of 21-60 have completed this study. As data collection is still ongoing seven moderately hearing-impaired have completed the study, however subjects will be recruited into the study until optimal power (β = 0.8) is achieved. The CV syllables /ba/, /pa/, /da/, /ta/, /ga/, and /ka/ were each filtered into two frequency bands (low pass and high pass). The EOA of these two bands were varied in 25 ms steps relative to the original token for each of the 6 stimuli. The participants performed a labeling task during which they listened to VC syllables that varied in EOA and then chose from two given labels which syllables they perceived. The participants also completed a same-different discrimination task using the same CV syllables as in the labeling task. Preliminary results indicate that both normal-hearing and hearing-impaired listeners perceive the EOA categorically, meaning that as the temporal onset asynchrony is varied between a low and a high frequency bands of speech, the perception of voicing changes from a voiceless phoneme to a voiced phoneme. Once data collection is complete, the influence of EOA on the perception of voice-voiceless features will be assessed by a final examination of the broadness of the labeling boundary slopes and discrimination peaks along the EOA continuum. The two groups will be compared to see if there are significant differences in the perception of spectrally asynchronous delays. A future goal of this research endeavor is to use this task to evaluate the impact of various digital delays as employed in amplification systems.

A29

Speech enhancement using a model of outer hair cell mechanism for digital hearing aid

Yoon Sang Ji & Young Woo Lee, Hanyang University, Seoul, Korea; Sung Hwa Hong, University School of Medicine, Korea; Jong Shill Lee, In Young Kim and Sun I. Kim, Hanyang University, Korea; Sangmin Lee, Chonbuk National University, Korea

In the field of speech processing for digital hearing aid, the speech enhancement is one of the important issues. Especially it has been noticed that the background noise causes remarkable reduction of speech intelligibility. So speech enhancement algorithms have been implemented in digital hearing aids to provide clean speech and enhance acoustic conditions leading to improved speech intelligibility by user. In this study, we proposed an algorithm which is a speech enhancement method using FSLI (Function of Spatial Lateral Inhibition) with spectral subtraction. FSLI is a psychophysical evidence about the lateral-hearing inhibition through psychophysical experiments. Responses of the basilar membrane are nonlinear. One consequence of this nonlinearity is that one tone can suppress the response to another tone that is close in frequency (Javel et al, 1978). Lateral inhibition is a common phenomenon involved in sensory reception of biological systems. The formants of input signal can be obtained by the FFT (Fast Fourier Transform), and the LPC (Linear Predictive Coding) is used to spectral envelope detection. The spectral envelope data pass through the convolution with the FSLI and a rectifier. The IFFT (Inverse FFT) is used to enhanced speech using the input speech's phase and amplitude component. So the spectral subtraction algorithm has been used for enhancing speech corrupted by background noise and FSLI is used for spectral contrast enhancement. The proposed algorithm was evaluated by computer simulation for noise reduction in speech which is contaminated by white noise (16kHz sampling). Simulation results show that the proposed algorithm was better speech enhancement performance than the conventional spectral subtraction algorithm and it is confirmed that proposed algorithm is effective for good performance on digital hearing aids. In this research, we estimate the possibility of application of the lateral suppression mechanism as the outer hair cell model to hearing aids in the background noise.

This study was supported by a grant of the Korea Health 21 R&D Project, Ministry of Health & Welfare, Republic of Korea. (02-PJ3-PG6-EV10-0001).

A30

Influence of the auditory efferent system on speech perception in noise: physiological and behavioral evidence

Srikanta K. Mishra and Mark E. Lutman,
ISVR, University of Southampton, UK

One of the functions of the efferent auditory system is to improve the detection of signals in noise (for review, Sahley, 1997). It is possible that reduction in the detrimental effects of background noise on speech perception may be proportional to the level of efferent activity, in particular, medial olivocochlear bundle (MOCB). If so, behavioral measures of speech-in-noise performance should be correlated with physiologic measures of efferent auditory system. Moreover, if objective measures correlate with speech-in-noise performance, they could be implemented into routine clinical procedures when attempting to determine hearing aid success. The present study investigated the involvement of olivocochlear feedback in speech-in-noise intelligibility in normal hearing subjects. The study examined; (i) the effects of contralateral acoustic stimulation on measures of speech-in-noise intelligibility; and (ii) its relationship with the strength of the olivocochlear feedback (assessed through contralateral acoustic suppression of otoacoustic emissions). The speech-in-noise measurements were performed using the four alternative auditory feature test (FAAF) in an adaptive paradigm. The analysis of data will be presented to discuss, if individual differences in efferent auditory system activity can account for intersubject variability in speech perception in noise.

Reference:

Sahley TL, Nodar RH, Musiek FE (1997). Efferent auditory system structure and function. San Diego: Singular Publishing Group.

Posters for Session B should be put up by 8 A.M. Friday, August 18, and taken down after 10 P.M. Friday, August 18 or before 7 A.M. Saturday, August 19. Presenters should be at their posters from 9:45 – 11:00 A.M.; 4:30 - 5:00 P.M.

POSTER SESSION B

Friday 8:00AM –10:00PM

B1

Bench-top SNR testing of open vs. occluded fittings

Robert Ghent and Michael Nilsson, Sonic Innovations, Inc.

The Noise Reduction Index (NRI) is a bench-top estimate of the signal-to-noise ratio (SNR) change of a mixed signal (target plus masker) through an audio system. The NRI makes use of multiple recordings obtained in a test environment consisting of a two-dimensionally-diffuse noise field with target signals presented from 0° azimuth. It has been used to characterize and quantify the SNR change of conventional hearing aids as well as advanced features such as directionality and noise reduction algorithms. In the present study, the NRI was measured on an open-ear hearing aid in both a sealed coupler (reference condition) and an open-coupler type fitting in order to assess the impact on changes to SNR from leakage through an open-tip fitting. Measurements were obtained in an unobstructed sound field and on a KEMAR. It is hypothesized that the low-frequency effects of an open fitting will reduce any SNR advantages the signal processing may introduce in the sealed coupler. Data will be presented showing how much of the change in SNR achieved in the coupler remains in the open-tip condition.

B2

Loudness judgments of non-musicians and musicians

Mead Killion, Etymotic Research, Inc; Edgar Villchur, Foundation for Hearing Aid Research; Brian Glasberg, University of Cambridge, UK

At IHCON four years ago, one of the participants stated: “Loudness doesn’t matter, “ to which another replied, “It certainly matters to musicians.” In this presentation we report loudness judgments of musicians and non-musicians. With non-musician subjects, we found highly reliable loudness judgments using an ascending-loudness trisection method. With Chicago Symphony Orchestra musicians, we obtained musical productions corresponding to musical notations of ppp to fff using an ascending-intensity production series. A bass trombonist produced tones over a 50 dB SPL range from pppp to ffff, with an almost exact 5.7 dB difference (1.5x loudness) between successive levels. Interestingly enough, non-musicians chose six levels using an arithmetic loudness progression, while the musicians produced tones using a geometric loudness progression. In addition to A-weighted SPL, one of the authors (BG) made measurements of the same musical tones using the a model of loudness applicable to time-varying sounds (BR Glasberg and BCJ Moore, J.Audio Eng Soc, 50, 331-342; 2002). A comparison between A-weighted SPL and the estimated loudness level in phons will be presented.

B3

VirtualHearing: A virtual acoustic environment for the evaluation of hearing aid algorithms.

Karl Wiklund, Simon Haykin, Ian Bruce, McMaster University, Canada

With the development of more sophisticated hearing aid algorithms, the question of effective testing and evaluation is becoming more pressing. New developments in hearing aids allow for greater consideration of the effects of spatially distributed noise and reverberation that exist in real acoustic environments. However, the testing of such algorithms under such conditions can be expensive and time-consuming and may require the use of specialized equipment. In addition, the need to test the performance of these algorithms for human users adds an additional layer of complexity to the task.

It is desirable therefore to perform much of this testing in a virtual environment. This environment should also simulate patient responses to acoustic inputs. The VirtualHearing system that we describe here is one such system. Based on acoustic simulation techniques developed for multimedia applications, and incorporating a computational model of the cochlea, VirtualHearing is a new software tool for hearing aid developers. This tool allows the user to define many different acoustic scenarios, and to control the level of reverberation, the intensity and direction of acoustic sources, as well as the level of hearing impairment suffered by the patient. The software also includes analysis tools to evaluate the quality of the processed speech, and also permits the user to develop and test custom processing algorithms. The cochlear model used also allows output files to be generated containing either the instantaneous neural spike rates, or the spike trains themselves. These neural outputs can then be used in other applications that simulate higher level processing in the auditory cortex. As a result we feel that this software is a valuable new development tool to hearing aid designers, and is one that will also be of great interest to those studying the neurobiology of hearing.

B4

Comparison of objective measures characterizing nonlinear hearing aid performance

Martin Hansen, University of Applied Sciences Oldenburg, Germany

In the recent past, several new methods for characterizing the nonlinear signal processing behaviour in modern hearing aids have been proposed. Also, the need for new appropriate test signals for analyzing the performance of a nonlinear hearing aid has been noticed in the community. This study investigates the interrelation of several new measures for characterizing nonlinear hearing aids when using speech and non-speech signals as input signals. Special focus is put on a new method presented by Olofsson & Hansen (IHCON 2002) which quantifies the amount of distortion from nonlinear signal processing in hearing aids. The measure was based on the use of special pairs of input signals, one being the Hilbert transform of the other, and the measure was calculated (only) from the two corresponding output signals. It could predict the subjectively perceived distortion of 12 simulated compression hearing aids.

A method is presented how the new Hilbert-pair-measurement can also be performed for real hearing aids. The method includes a means for meeting the requirement of an extremely precise relative timing for the pairs of signals, using a standard PC and soundcard equipment. A number of 8 different modern commercial hearing aids were fitted to different generic hearing losses and presented with different types of speech in quiet, speech in noise and test signals. Objective measures of compression and nonlinearity were collected for all possible combinations of hearing aid, fitting, and input signal. Here, the results of the Hilbert-pair-measure are presented and compared to other modern and standard analyses for hearing aids. The dependencies of the different measures on the choice of the input signal are discussed.

B5

Factors influencing localization with an electronic pass-through hearing protector

Brian Hobbs, Douglas Brungart, Nandini Iyer, and James Hamil, Air Force Research Laboratory

Traditionally, the goal of a hearing protection device (HPD) has been to attenuate noise enough to enable persons to maintain operations safely within loud noise environments. More recently, electronic pass-through HPDs have been developed that incorporate dynamic range compression, similar to that used in hearing aids, to protect the users from high-levels noise while preserving their ability to hear low-level ambient sounds in their surrounding environment. However, because these electronic devices alter spectral and temporal parameters, certain auditory abilities (for example, localization) can be compromised. Two binaural cues, interaural time differences (ITD) and interaural level differences (ILD) play an important role in helping listeners locate a sound in space, and both can be significantly altered by HPD use (whether passive or pass-through). In addition, localization cues can also be affected by limitations of the electronic pass-through technology (for example, bandwidth of the microphone, amplifier, driver, coupling, etc.) or by the configuration of the technology (for example, linear versus compression algorithms). In this study, we examined how well normal-hearing listeners were able to localize brief and continuous sounds with an electronic pass-through hearing protection device that allowed the manipulation of such parameters as bandwidth, compression type, attack time, and release time. The results are discussed in terms of their implications for the design of dynamic compression algorithms that protect hearing in loud environments but minimize the impact on the auditory situational awareness of the users.

B6

Effect of multiple compression channels on vowel identification

Stephanie Bor, Pamela Souza & Richard Wright, University of Washington

Although a current trend for digital hearing aids is for wide-dynamic range compression (WDRC) processing in larger number of compression channels, it remains unclear if more channels truly provide benefit to hearing impaired individuals. Previous researchers have suggested a possible degradation of vowel discrimination as the number of compression channels increases due to spectral flattening. The goal of the present study was to relate acoustic measures of spectral flattening on vowels to identification scores.

Stimuli for both the acoustic analysis and behavioral testing consisted of eight naturally produced vowels spoken by twelve talkers (6 male, 6 female). The vowels were digitally processed through WDRC simulation, using 1, 2, 4, 8 and 16 channels with an input of 75 dB SPL, compression ratio of 3:1 and a compression threshold of 45 dB SPL in every channel. A spectral contrast measure was developed to quantify the relationship between spectral flattening and the number of compression channels, and was defined as the difference in average peak and average trough intensities for the first and second formants (Bor et al., submitted). In order to assess the relationship between the spectral contrast measures and vowel identification, subjects with mild to moderately-severe, sloping sensorineural hearing loss were tested with an 8-alternative forced-choice identification task. The vowels were presented monaurally in quiet via supra-aural earphones. Overall RMS levels of the vowels were equated across conditions and each vowel presentation was varied by ±2 dB to eliminate intensity bias. Presentation order of the conditions and order of vowels within each condition was randomized for each subject.

Preliminary results (n=8) from these studies indicate a correlation between vowel identification and the measure of spectral contrast; specifically, lower spectral contrast measures are associated with lower identification scores. These results suggest that high numbers of compression channels may decrease vowel identification. [This work was supported by NIH RO1 DC006014.]

B7

Open ear hearing aid amplification in tinnitus therapy

U. Ambrosetti, IRCCS, UO Audiologia, Italy; M. Bettinelli, L. Del Bo & E. Domenichetti, Del Bo tecnologia per I'ascolto, Italy; E. Fagnani, IRCCS, UO Audiologia, Italy; A. Scotti, I clinica ORL, Italy; <u>O. Dyrlund</u>, GN Resound, Denmark

The aim of this study was to assess the efficacy of sound stimulation delivered by open ear hearing instruments (OHI) for tinnitus treatment using Tinnitus Retraining Therapy (TRT). Tinnitus patients falling within Jastreboff 's tinnitus categories 1 and 2 (Henry et al. 2002) and with mild moderate hearing loss in the 2-6 kHz frequency range were included.

The results collected through the questionnaires administrated (structured interview (Jastreboff et al. 2002) and self-administered questionnaire (Henry et al. 1996) show that the sound stimulation delivered by the OHI combined with TRT was be able to provide successful results within relatively short periods of time (average: 6.9 months). Therefore OHI seem to be an effective solution for sound enrichment in TRT for mild and moderate sloping hearing losses, and could substitute artificial sound generation in many cases.

References

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B8

Measuring intermodulation distortion in hearing aids using arbitrary broad-band input signals

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At IHCON 2002, Hansen & Olofsson proposed a method for measuring nonlinear distortion. The method was using two test signals, of which the second one was the Hilbert transform of the first one. When testing a hearing aid, the linear part of the response was defined as the parts of the two output signals, which were still related through the Hilbert transform, and the remaining parts were defined as distortion. This distortion measure was compared with results of a listening test on a number of systems and program materials. One of the conclusions drawn from the experiment was that low measured distortion, defined as described above, was a necessary but probably not a sufficient condition for a system to be free from perceptual distortion.

Here another measure of nonlinear distortion is described, namely intermodulation distortion from interactions of two independent broadband test signals, e.g. a speech signal and a noise signal. The method uses the same technique to separate the response signals at the output of the hearing aid, as previously described by Hagerman, Olofsson & Nästén (IHCON 2002), and Hagerman & Olofsson (Acta Acustica, Vol 90, pp.356-361). First a measurement is done with added test signals and then a similar measurement is performed with one of the signals phase inverted. The response signals from the hearing aid are then added and subtracted, respectively, to recover two output signals, which are filtered versions of the separate input test signals. After the separation, the amount of the output signal power, which contains intermodulation components from quadratic and cubic nonlinearities, is calculated as a function of frequency.

The two different methods of distortion estimation will be compared and results shown on compression hearing aids with fast and slow time constants, as well as measurements on noise-reduction hearing aids.

B9

Comparison among various systems for hearing aid measurements

Henrik L. Olsen, Oticon A/S, Denmark; Karolina Smeds, Karolinska institutet, Sweden; Åke Olofsson, Karolinska institutet, Sweden

When prescribing a hearing instrument, it is important to verify that the signal processing works on the client as intended. Real ear measurements are therefore routinely used when fitting hearing aids. It is, however, crucial, that measurements are done adequately. Many factors affect the estimated gain response of a hearing aid, e.g. the levels and type of input signal, dynamic response of the instrument and the equalisation method used. Another factor, which is often overlooked, is the way the gain response is defined when using a broadband signal. According to the standards, gain responses could be either auto-spectrum based (IEC 61669) or cross-spectrum based (IEC 60118-2) when assuming long-term spectrum averaging. However, most commercial equipment employs a gain estimation based on short term spectrum averaging.

The aim of this study was to compare different measurement systems. The main focus of the study is to document any differences among the measurement systems especially with regard to the choice of input signal, preconditioning time, estimation time and test reliability. Four commercially available equipments for hearing aid measurements were tried along with a laboratory based reference equipment. A commercially hearing aid configured with three non-linear and one linear hearing aid responses are used as measurement objects. With a few exceptions, the gain response between the different commercial equipments using similar signal were quite similar. Differences where observed when using different spectrum averaging (long-term versus short term) when estimating the gain-frequency response using speech or speech-like signals. To help understand these findings simulations of different estimation procedures of gain response where tried.

B10

A model of hearing aid benefit based on performance measures with hearing aid users

Victor Bray, Michael Nilsson, and Robert Ghent, Sonic Innovations, Inc.

Speech-recognition-in-noise measurements have been obtained on several hundred hearing aid subjects across multiple research clinics over several years. The research sites followed a standardized protocol and utilized a standardized test platform [Test Paradigm Manipulation During the Evaluation of Speech Recognition in Noise, IHCON Poster (2002); Development of a Test Environment to Evaluate Performance of Modern Hearing Aid Features, JAAA, 16:27-41 (2005)]. Performance measures were obtained on all styles of hearing aids (CIC through BTE) on subjects with varying degrees of sensorineural hearing loss (mild through severe) with various hearing aid technologies selectively engaged (multi-channel compression, directional microphones, and digital noise reduction).

This 'large n' data base is being analyzed to determine hearing aid benefit for speech-recognition-in-noise as a function of unaided hearing loss in quiet, unaided hearing loss in noise, and hearing aid technologies. It is hoped that the results will lead to a predictive model for hearing aid benefit for speech-recognition-in-noise with hearing aids. Preliminary results lead to interesting correlations between unaided thresholds in quiet and aided thresholds in noise (R^2 ranging from .18 to .23) and unaided thresholds in noise and aided thresholds in noise (R^2 ranging from .40 to .53), with respect to the various hearing aid technologies.

B11

A first attempt at a comprehensive Own Voice Qualities (OVQ) questionnaire

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Traditionally, hearing-aid users' problems with their own voice have been attributed solely to occlusion. Thus, among the broad population of audiologists and hearing-aid dispensers, the conception about hearing-aid users' own-voice problems still seems to be that all own-voice problems are due to occlusion, and accordingly that all own-voice problems have been solved by open fittings. However, since open fittings have become mainstream it has become clear – at least to the authors – that even hearing-aid users whose occlusion problems are essentially solved still may have issues and concerns about their own voice.

When trying to demonstrate (and prove) that there are own-voice issues left even for hearing-aid users who have negligible occlusion problems, a major obstacle turned out to be the lack of a suitable instrument for evaluating people's perception of their own voice. Hence it was decided to develop a new questionnaire – named Own Voice Qualities (OVQ) – for this purpose. Thus, the aim of the questionnaire is to enable identification of possible self-perceived own-voice problems in connection with hearing-aid use, including but not restricted to problems caused by occlusion.

Both the general contents and the specific wording of the individual items in the questionnaire have to a large extent been based on actual statements about own voice made by hearing-aid users, rather than the researchers' suppositions. These statements were mainly made during a focus group interview, where a group of hearing-aid users discussed a variety of own-voice issues, and during a number of individual interviews where hearing-aid users were asked about their perception of their own voice.

The OVQ questionnaire includes items on a variety of own-voice aspects, e.g., the sound quality of own voice, the interaction between own voice and the voices of other people in communication situations, other people's perception of the respondent's own voice, the level of own voice in different situations, ways of controlling the level of own voice, whispering, and the physical experience of own voice. In order to minimize the risk of test subjects misunderstanding the items, the questionnaire is supposed to be filled in during an interview session where the interviewer goes through all the individual items together with the respondent.

The first informal tests of the OVQ questionnaire have been encouraging and more formal studies involving the questionnaire have therefore been initiated. Preliminary data from these studies will be presented.

B12

The performance-perceptual test as a counseling tool

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Despite considerable improvements in hearing aid technology, hearing aid dissatisfaction is surprisingly high. A number of studies have shown that this is in part due to a variety of patient-based factors such as attitudes, expectations and personality. Previous data collected with the Performance-Perceptual Test (PPT) has shown that individuals who underestimate their hearing ability report more handicap than would be expected based upon their hearing impairment, while individuals who overestimate their hearing ability report less handicap than would be expected from their hearing loss (Saunders et al., 2004; Saunders and Forsline, in press). The purpose of this study is to determine whether simple counseling based upon discussion of PPT results can be used to better align perceived and measured ability to understand speech-in-noise; and, more importantly, to determine whether such counseling can decrease reported handicap and improve hearing aid satisfaction, regardless of its impact upon perceived hearing ability.

Hearing aid users complete the PPT for aided and unaided listening, along with standardized questionnaires measuring reported auditory disability, handicap and hearing aid satisfaction. Following this, subjects are randomly assigned to one of two groups. Subjects in Group 1 receive counseling from the experimenter in the form of an explanation and discussion of their PPT results which includes discussion of the extent to which they misjudge their hearing ability, its potential impacts and suggestions for adjusting their behavior. Subjects in Group 2 also participate in a discussion with the experimenter, but it does not include mention of the extent to which they misjudge their hearing ability. Two weeks after enrollment in the study subjects complete a second set of questionnaires. Ten weeks after study enrollment subjects return to the laboratory to rerun the test battery. Data will be presented in terms of the impact of the counseling across the two groups upon misjudgment of hearing ability, reported handicap and hearing aid satisfaction and benefit.

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B13

Making young ears old (and old ears even older): Simulating a loss of synchrony

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Age-related changes in the auditory system have been attributed to three independent factors: OHC damage, changes in endocochlear potentials, and loss of neural synchrony (Mills et al. 2006). The high-frequency hearing loss typically observed in presbycusis is results from one or both of the first two factors. While loss of neural synchrony has little effect on audiometric thresholds, it is thought to contribute to difficulties understanding speech in noise.

The goal of this research was to try and identify the consequences of a loss of synchrony on speech intelligibility in noise. Young and old adults with good audiograms in the speech range were presented with SPIN-R sentences in two SNR and three processing conditions: intact, jitter, and smear. The parameters of the jittering algorithm were chosen to simulate a loss of synchrony consistent with prior psychoacoustic and speech experiments on auditory aging. The parameters of smearing algorithm were chosen to match the spectral distortion produced by the jitter algorithm. For both jitter and smear, the distortions were restricted to the 0-1.2 kHz frequency band.

Previous findings of an age-related difference in the intact conditions were replicated. For both age groups, the jitter condition resulted in a significant decline in word identification. However, the smear condition resulted in a significant decline only for the older age group but the decline was not a large as that in the jitter condition. Since a difference in performance between jitter and smear must be due to phase distortion (i.e. a simulated loss of synchrony), the results for both age groups suggest that a loss of synchrony can have a deleterious affect on speech intelligibility in noise. Furthermore, the performance of the young in the jitter conditions was similar to that of the older adults in the intact conditions for low context sentences. Thus, the jitter condition appears to simulate this neural aspect of auditory aging in otherwise healthy young ears.

B14

Validity of data logging systems in hearing instruments

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Data logging – i.e., the recording of individual hearing instrument usage for later inspection by the dispenser – has become a feature widely available in most modern high end hearing instruments. It offers the acoustician the possibility to base his fine tuning or trouble shooting on objective data that he cannot obtain reliably from the hearing instrument wearer by other means. In order for the acoustician to make use of the information, he has to rely on the validity of the logged data with respect to the real world. Additionally, the correspondency of the logged data with the individual subjective impression is to be considered. If there are large systematic deviances the acoustician has to decide whether to fine tune the instrument based on the logged (technical) data or – conventionally – based on the direct feedback by the hearing instrument wearer.

The validity of different data logging systems was addressed in a first study. Seven hearing instruments possessing four different data logging systems were simultaneously exposed to a controlled sequence of stimuli reflecting distinct acoustic conditions. After 86 hours of continuous stimulus presentation the data were read out and analyzed. Results generally show a wide spread of accuracy between the different systems. For some features such as <time spent in environment: music>, the data logging showed values between 0% and 38% (true value: 36%), or for <time spent in environment: speech in quiet> between 17% and 67% (true value: 13%).

A second study addressed the correspondency between logged data and the subjective impression. 21 subjects wore one hearing aid with data logging capabilities during their daily lives for 2-3 weeks with the data logging feature turned on. Additionally, the subjects recorded the acoustic environment each day in a personal journal with a granularity of 30 minutes. The subsequent analysis revealed that most acoustic situations showed a relatively small mismatch (less than 7% for each particular environment) between the logged data and the data recorded in the journals. The only exception was the environment <speech in noise> where the logged data were on average 17% higher than the journal data.

Given the potential misadjustment of hearing instruments due to inaccurately logged data, it seems important that the logged data capture the real world conditions with an adequate accuracy. Additionally, systematic deviations for some logged features between the recorded data and the perceptual correlates should be considered by the dispenser during fine tuning.

B15

Consonant recognition at multiple input levels for severe loss, using clinically fit linear vs. non-linear hearing aids.

Evelyn Davies-Venn, Pamela Souza, Christopher Stecker, University of Washington

Improved speech audibility is one of the primary goals of multichannel wide dynamic range compression (WDRC) hearing aids. Some studies have suggested that the benefit of improved audibility may be offset by acoustic alterations, especially for listeners with severe hearing loss (Souza et al., 2005 Ear Hear 26:2 120-131). This study's goal was to evaluate consonant recognition and error patterns as a function of input intensity for both linear compression limiting (CL) and non-linear WDRC hearing aids. Such information will highlight the role and limitations of audibility in phoneme perception for the severely hearing-impaired population.

A within-subject repeated measures design was used to evaluate patterns of consonant recognition and confusion for severely hearing impaired listeners. Each subject was binaurally fitted with wearable hearing aids using individualized and clinically appropriate compression and gain parameters. The test conditions were compression limiting (CL) and four-channel WDRC amplification, and were counterbalanced across subjects. Block randomized consonant-vowel /Ci/ nonsense syllables were presented in soundfield at multiple input levels (50, 65, 80 dB SPL). Subjects were trained to select the stimulus heard from an on-screen display of 22 /Ci/ tokens. Sequential information feature analysis (SINFA; Wang & Bilger 1973) was used to analyze transmitted information for consonant place, manner and voicing for stimuli presented at three input intensity levels.

Results revealed benefits related to improved audibility with WDRC amplification only when the input signal was below conversational level (i.e. 50 dB SPL). Consonant recognition was better for WDRC than CL amplification at 50 dB SPL, but similar across conditions at higher input levels. Overall recognition scores increased from 50 to 65 dB SPL, and decreased from 65 to 80 dB SPL, in both conditions. SINFA revealed that WDRC transmitted greater information about consonant voicing, manner and place at low input levels (i.e. 50 dB SPL) compared to CL. At conversational levels (65 dB SPL) WDRC transmitted more information about voicing than CL amplification, but information about manner and place features were similar across conditions. At high levels (80 dB SPL) WDRC transmitted more information about voicing, but less information about place and manner, compared to CL. [Supported by NIH R01 DC006014]

B16

Interaction of hearing-aid compression with spatial unmasking

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For unaided listeners, speech that is spatially separated from interfering speech is more intelligible than speech that is co-located with interfering speech. This benefit of spatial separation is known as spatial unmasking. Spatial unmasking is elicited in large part by differences in perceived location of target speech and interfering speech. This study examines whether hearing-aid compression acting independently at the two ears reduces spatial unmasking by distorting interaural-level-difference (ILD) cues for location.

We addressed the question by measuring spatial unmasking of speech, with and without compression. Subjects heard diotic target sentences presented amidst two interfering sentences, and they were instructed to identify the sentence at the center of the head. All sentences on a given trial were spoken by different male talkers in the coordinate-response-measure corpus, were computer processed, and were presented via headphones. We measured spatial unmasking by comparing the speech reception threshold (SRT) obtained for co-located diotic interfering sentences with the SRT obtained for interferers made dichotic by the imposition of ILDs; each of the two interferers had ILDs favoring opposing ears. To ensure that any unmasking was due only to ILD and not to any improvement of signal-to-noise ratio, we fixed the masker level at the better ear across all trials and conditions.

Normal-hearing subjects obtained approximately 0.5 dB of unmasking for every 1-dB ILD imposed on the interferers. For a 16-dB ILD on the interferers, this unmasking reduced by 4 dB when stimuli were processed by a 15-band compressor having 3:1 compression, near-instantaneous attack time, and 50-ms ANSI release time. The amount of spatial unmasking varied with compression parameters. The reduction of unmasking was attributable to independent compressors at the two ears reducing ILDs and causing the perceived spatial separation between target and interferers to be smaller than in corresponding unprocessed conditions.

These findings indicate that compression, as implemented in current hearing aids, can reduce spatial unmasking. The implications of these results for the design of hearing aids will be addressed.

B17

Adaptive feedback cancellation in hearing aids with nonlinear feedback paths

Daniel J. Freed, House Ear Institute

The Normalized LMS (NLMS) algorithm is often used to estimate and cancel feedback in hearing aids. Because NLMS is designed to model linear phenomena, nonlinearity in the feedback path may undermine the performance of a NLMS-based feedback canceller. This study focused on one type of feedback path nonlinearity: clipping of the feedback signal arriving at the microphone.

Simulations were performed to investigate how the adaptation time of a NLMS-based feedback canceller was affected by clipping of the feedback signal arriving at the microphone. Adaptation time rose gradually as forward and feedback gain were increased, up to a critical point where the canceller failed to adapt because the NLMS filter coefficients consistently underestimated the target values. The canceller was then modified to clip the cancellation signal, so that the cancellation path explicitly modeled the nonlinearity in the feedback path. This change allowed successful adaptation at gain levels well above the critical point.

A second set of simulations investigated the effect of input signal level on the performance of a NLMS-based feedback canceller with feedback signal clipping at the microphone. Noise bursts were presented to a canceller after the NLMS filter coefficients had been initialized to target values. When the noise burst level was high enough to produce clipping of the resultant feedback signal, the canceller responded by reducing the magnitudes of the filter coefficients; at the conclusion of the noise burst, the canceller sometimes failed to readapt. Clipping the cancellation signal guaranteed successful readaptation and minimized the coefficient drop during the noise burst at moderate noise levels, but provided less benefit at higher noise levels.

A physical system was then constructed consisting of an open-ear hearing aid with a NLMS-based feedback canceller. The gain of the feedback path and the receiver driving level were controlled by analog amplifiers. The simulation measurements were repeated with the physical system, and a similar pattern of results was obtained. Canceller performance did not decline substantially even when the receiver was driven at high levels, suggesting that receiver distortion may not pose a major problem for NLMS-based feedback cancellers.

B18

The effect of speaker familiarity on adult and pediatric informational masking

Jason Galster, Todd Ricketts, Vanderbilt University

Speech recognition in noise ability varies significantly with changes in noise type. The effects of a broadband masking noise on speech are well documented, however, the effects of speech-on-speech masking are less predictable and may result in situations under which the listener is unable to disentangle the content of overlapping speakers. When performance in a speech-on-speech listening task is compared to that of a task where the speech maskers are altered or simulated in a manner which makes them unintelligible, listener performance may improve, even though the energetic properties of the masking speech remain grossly unchanged. This difference in performance has been referred to as informational masking. To date, research in this area has not approached the effects of informational masking as a function of speaker familiarity and listener age.

Studies have shown benefits of listener familiarity with subjects trained to recognize a voice. The current study has controlled for listener familiarity by using speakers and listeners within a family triad which included the mother, father, and a child (age 7-11). A modified version of the Coordinate Response Measure (CRM) was generated for this study. The CRM uses a speech-on-speech paradigm during which the listener is prompted to identify colors and numbers spoken by the target speaker. For this study the mother of the family triad served as the familiar speaker and a second female speaker, unknown to the families, served as the unfamiliar speaker. Test subjects consisted of each family's father and child. Test conditions were completed at three signal-to-masker ratios and used two competing maskers which were played forward and reversed to create conditions of differing informational content. Early results suggest that speaker familiarity may reduce informational masking effects, however pediatric performance shows high variability and significantly increased susceptibility to confusion in highly informational test conditions.

B19

Development of a modified HINT protocol and a new localization test for evaluating adults with vastly different binaural abilities

Jenny Chan, Andrew Vermiglio, Daniel Freed, Sigfrid Soli, House Ear Institute

A modified test protocol for the adaptive Hearing in Noise Test (HINT) and a new horizontal plane sound localization test have been developed for use by acoustic-hearing (AH) subjects and bilateral cochlear implant (CI) users with vast difference in their binaural hearing abilities. Signals are delivered either via sound field (SF) or via direct connect (DC) input in both of these tests. For AH subjects, headphones are used to deliver the DC input. For CI subjects, a prototype instrument is used to feed signals directly to the auxiliary inputs of their processors. Sets of calibrated head-related transfer functions (HRTFs) from sources at various azimuths in the horizontal plane were measured through KEMAR's ear canal microphone and through several CI microphone placements. The HINT speech and noise materials were processed with the HRTFs for 0º, 90º, and 270º azimuths to simulate spatially separated speech and noise sources. Three adaptive rules were developed to be used with CI subjects who usually do not achieve 100% intelligibility in quiet. These rules aimed at measuring performances at different points on the performance-intensity (PI) function. The three rules are: Rule 1 – allows no errors; Rule 2 – allows one error per sentence, which corresponded to sentence intelligibility of 75-99% ; and Rule 3 – allows two or three errors per sentence depending on sentence length, whichever produced intelligibility of 50-74%. Pairs of S/N ratios and percent intelligibility scores were obtained from 12 AH subjects to elucidate the relationship among these rules. By using the most appropriate rule, binaural directional hearing abilities of 5 bilateral CI subjects could be compared directly with those of AH individuals. A new adaptive sound localization test was also developed to evaluate simple absolute localization ability. In the SF, an impulse noise was presented randomly through 12 loudspeakers that were arranged in the rear horizontal plane of subjects' ears from 97.5° to 262.5°. The same stimulus was processed with HRTFs for the 12 azimuths in the DC test. For subjects who were unable to identify all of the source azimuths, a protocol to derive sector accuracy scores was established. Data from 12 normally hearing subjects and 5 bilateral CI subjects with widely varying localization abilities will be reported.

The same modified protocols are expected to be applicable to hearing aid users. A strong correlation between test results measured in the SF and via DC supports the use of DC methods as a practical means of measuring binaural directional hearing and sound localization abilities when well-controlled SF environments are unavailable.

B20

Two in-situ approaches to assessing the audibility of amplified speech for BAHA users

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Determining the audibility of amplified speech for bone conduction hearing aids has been a challenge for many years. Despite the many limitations of aided soundfield thresholds as a verification tool for aided speech, viable alternatives for BAHA have not emerged.

We propose 2 novel in-situ verification procedures for BAHA. The first procedure (Accel-o-gram) utilizes accelerometers to capture the accelerations associated with a subject's bone conduction threshold and aided BAHA output. The second procedure (SPL-o-gram) utilizes a real-ear probe microphone to measure the ear canal SPL associated with a subject's bone conduction threshold and aided BAHA output. The dependent variable for each approach, speech audibility, will be determined by subtracting the thresholds from the BAHA aided long term average speech spectrum (LTASS).

Data from 30 subjects will be analyzed using a 2 x 5 entirely within-subjects, repeated measures ANOVA. The first within-subjects independent variable is APPROACH having 2 levels (Accel-o-gram, and SPL-o-gram). The second within-subjects independent variable is FREQUENCY having 5 levels (250, 500, 1000, 2000 and 4000 Hz). Data will be analyzed for any main effects or interactions that may be present. Equivalent results for both procedures will be taken as evidence that both approaches yield valid measures of real head performance of BAHA. Additionally, speech intelligibility in noise data from the HINT will be presented to validate these objective approaches to BAHA verification.

B21

Effects of noise suppression of digital hearing aids

Tatsuo Nakagawa, Yokohama National University, Japan

In order to evaluate the effectiveness of noise suppression digital hearing aids, a noise suppression index(NSI) and a speech audibility in noise index(SANI) were developed. The NSI is defined as a ratio of noise power between the original stimulus and the amplified stimulus. The SANI is defined as a ration of speech power above noise between them. Using these two indexes, 6 kinds of commercial digital hearing aids were evaluated in a hearing aid test box. Different effects of noise suppression were found among these hearing aids for the different noise conditions. A subjective evaluation was also performed. Normal hearing subjects wore each hearing aid monaurally and listened to speech in noise from one speaker in a sound proof room. They compared the relative noise suppression effects of the hearing aids in random pairs. The subjective evaluations were in good agreement with the indexes. It is suggested that the indexes are useful to predict hearing-aid performance in noise.

B22

Time-varying compression amplification with enhanced peak-to-valley ratio

Zaker Siraj and Janet C. Rutledge, University of Maryland; Peggy B. Nelson, University of Minnesota

Multichannel amplitude compression processing is used to reduce the level variations of speech to fit the reduced dynamic ranges of listeners with sensorineural hearing loss. This processing, however, can result in smearing of temporal information, artifacts due to spectral discontinuities at fixed channel edges, and spectral flattening due to reduced peak-to-valley ratios. An implementation of a time-varying compression processing algorithm based on a sinusoidal speech model (Col-SM) was presented at a previous conference (IHCON 2000). The algorithm operates on a time-varying, stimulus-dependent basis to adjust to the speech variations and the listener's hearing profile. The algorithm provides fast-acting compression without artifact, has time-varying frequency channels, is computationally inexpensive and preserves the important spectral peaks in speech. A modification of that algorithm was presented at IHCON 2002 that used an LPC-based estimate of the spectrum to sharpen peak-to-valley ratios. In that paper spectral sharpening parameters were determined empirically, however the results from testing on the algorithm were mixed due to limitations with that implementation. Presented here is a Col-SM based algorithm that determines the optimal levels of key spectral peaks and valleys to achieve time-varying compression amplification with spectral sharpening. The amount of sharpening can be set individually based on the needs of the listener. Preliminary subject tests will be presented. [This work is supported by the National Science Foundation].

B23

Perceptual learning of amplitude-modulated tones: Evidence for selectivity in the modulation domain?

Christian Füllgrabe, Anthony Yii, and Brian C.J. Moore, University of Cambridge, UK

Most species-specific vocalizations show prominent fluctuations in amplitude over time. In continuous speech, these amplitude modulations (AMs) are found predominantly in the low frequency range, with a maximum at about 3-4 Hz that corresponds to the average syllabic rate. Data from many psychophysical studies indicate that these temporal-envelope cues are sufficient and perhaps even necessary for speech identification.

Given the importance of the temporal structure of speech, an increasing number of studies have attempted to determine the functional architecture of temporal-envelope processing and to specify the nature of the sensory representation of temporal envelopes. Recent electrophysiological, brain-imaging, and simulation studies suggest the existence of some form of selectivity in the AM domain, either in terms of (i) different (temporal vs. rate) coding mechanisms (Anderson et al., 2006), or (ii) an array of AM channels (i.e., a "modulation filter-bank"), each tuned to a different modulation frequency, which decomposes the temporal envelopes of sounds into spectral components (Dau et al., 1997).

The present study aimed to provide evidence for the existence of either form of selectivity by using a perceptual-learning paradigm. The primary goal was to determine (i) how multi-hour training on the detection of AM with a given (high or low) modulation frequency improves thresholds for this particular AM frequency, and (ii) if the learning effect generalizes to detection of AM at a different frequency, for which the subject has not been trained. Preliminary results show a clear

training effect for a low modulation frequency (5 Hz) but not for a high modulation frequency (97 Hz). However, post-training detection thresholds for AMs of untrained modulation frequencies also improved. The outcome of this study provides new insights into the way the human auditory system encodes temporal-envelope cues, and thereby may prove useful for the design of rehabilitation strategies for persons with auditory temporal-processing deficits. An additional (and more methodological) goal of this study was the evaluation of the relative contribution of perceptual and procedural learning to the early stage threshold improvements (Hawkey et al., 2004).

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B24

Development of a protocol for an aided version of the TEN (HL) test

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The threshold equalising noise (TEN) test was developed to identify regions of loss of inner hair cells (IHCs) or neurons, referred to as dead regions (DR). The TEN HL test was designed to reduce the overall loudness of the TEN by limiting the frequency range from 500 to 4000 Hz. However for subjects with severe and profound hearing loss it is sometimes not possible to generate sufficiently high levels of noise, through the audiometer. This can restrict the applicability of the TEN test for more extensive hearing losses. An alternative approach is to apply the TEN test in the sound-field, while the subject is wearing a hearing aid. The non-flat frequency response of the hearing aid should not be of great importance, as the test depends mainly on the signal-to-noise ratio. We report on the viability of performing the aided TEN test.

Twenty-five subjects (12 males and 13 females) aged between 12 and 19 years, with severe or profound hearing loss, participated in the study. Absolute and masked thresholds in TEN were measured (usually for the better ear) using warble tones presented via an audiometer and headphones. Testing was repeated in the sound field via a loudspeaker, with subjects wearing a hearing aid in the test ear. The criteria for identifying a DR were the same as for the "standard" TEN(HL) test: when the masked threshold is 10 dB or more above the TEN level per ERBN, and the masked threshold is 10 dB or more above the (aided) threshold in quiet.

Similar profiles of DRs were obtained, although criteria were met over a greater frequency range using the aided TEN technique for some subjects. Some subjects did not fulfil the criteria for a DR for sounds presented via the audiometer and headphones, but did fulfil the DR criteria with aided sound field presentation. No subjects met the criteria using headphone presentation, but not in the aided presentation. The results suggest that higher effective masking levels can be achieved through amplification than through headphone presentation, making the test more applicable for severe or profound hearing loss. Overall TEN levels are limited by the maximum output of the hearing aid, thus reducing the likelihood of hearing damage from the TEN.

B25

Performance testing of open ear devices in noise

Michael Nilsson and Victor Bray, Sonic Innovations, Inc.

Questions have arisen regarding the use of advanced signal processing techniques in open ear devices which do not occlude the ear or provide amplification at low frequencies. If the devices are designed to allow low frequency information freely in and out of the ear, and many environmental noises are low frequency in nature, can advanced signal processing features provide any benefit? HINT thresholds were measured on 12 subjects in unaided and aided conditions including various levels of noise reduction and adaptive directionality. The APHAB was also administered to evaluate perceived benefit from these devices. Data will be presented to indicate whether advanced signal processing can contribute to performance in open style devices.

B26

Morphometry of the middle ear and the inner ear: MicroCT measurements and analysis

Sunil Puria, Jae Hoon Sim, James Tuck-Lee, Minyong Shin and Charles R. Steele, Stanford University

Our goal is to develop anatomically based 3D biomechanical models of middle ears and cochlea for which morphometry data is critical. The computational models of the middle ear are important for improving implantable hearing aids. The models of the cochlea are important for understanding the effect of implanted short electrodes on residual hearing and for bone conduction hearing. To obtain morphometry of the ear, histological methods have been the primary technique. However, this technique is destructive and certainly not appropriate for in-vivo imaging of individual subjects. One of the most recent advances for obtaining anatomical information is computed-tomography with um resolutions (microCT). Here we describe methods to determine parameters, needed for computational models, from the microCT imaging modality. MicroCT images, at 10-20 um resolution (both in plane and out of plane), were obtained from cadaveric temporal bone ears of human, cat, chinchilla and guinea pig using a Scanco VivaCT 40 scanner. The high-resolution images (500 to 1500 slices) were used for 3D reconstructions of the ossicles, suspensory ligaments and tendon, tympanic membrane eardrum curvature and its relative position in the ear canal, tympanic membrane thickness, middle ear cavities, scala vestibule and scala tympani area functions, and primary and secondary osseous spiral laminae. Results indicate significant inter-subject variability amongst individual subjects and across species. Morphometry measurements will include calculations of: (1) principal axes and principal moments of inertia of the malleus-incus complex and stapes, (2) dimensions and angles of suspensory ligament and tendon attachments relative to the principal axis, (3) malleus-incus joint spacing, (4) Eardrum thickness as a function of position, (5) middle ear cavity volumes and location of septa and foramen (if any). The microCT imaging modality offers some distinct advantages over existing histological methods. These include: (1) elimination of stretching distortions commonly found in histological preparations, (2) use of a non-destructive method, (3) shorter preparation time (hours rather than 12-16 months), and (4) results already in digital format. [Work supported by the NIDCD of NIH]

B27

Using FFT-based frequency transposition to improve consonant identification in listeners with severe high-frequency hearing loss

Joanna Robinson, Tom Baer, Thomas Stainsby and Brian Moore, University of Cambridge, UK

People with severe high-frequency hearing loss have difficulty identifying consonants characterized by their high-frequency energy, as in “tip-sip-ship-chip”. 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 (Moore et al., 2000) – then amplification of frequencies well above f_e may not be beneficial, or even lead to deterioration of speech understanding. However, amplification of frequencies up to $1.7f_e$ is often beneficial (Vickers et al., 2001). Frequency transposition, recoding information from high to low frequencies, provides a possible means of conveying high-frequency information to such people.

We designed and evaluated an FFT-based transposition technique that was adapted to the value of f_e for each subject. Frequencies up to $1.7f_e$ were left unprocessed, preserving information about voice pitch. When high frequencies dominated the spectrum, high-frequency components within a certain target range were transposed to the range f_e to $1.7f_e$ and were presented ‘on top of’ any original frequency components falling within this range. Otherwise, no transposition occurred. This conditional transposition prevented the interfering influence of high-frequency background noise. In a control condition, stimuli were low-pass filtered at $1.7f_e$.

In a pilot study with five subjects some benefit was seen, but only one subject showed significant improvements. Individual consonant confusions suggested that adjustment of the algorithm’s parameters, such as the level of the transposed signal and the frequencies selected for transposition, could lead to greater benefit. These changes have been made for the current study.

Six listeners with high-frequency dead regions are being tested using vowel-consonant-vowel nonsense stimuli. Dead regions were diagnosed using the TEN(HL) test (Moore et al., 2004) and values of f_e were determined more precisely using “Fast PTCs” (Sek et al., 2005). The value of f_e varied from 0.8 kHz to 1.5 kHz. At the time of abstract submission, data were still being collected. To judge a potential benefit from transposition, performance for transposed and control stimuli will be compared regarding overall performance, information transmitted, the proportion of correctly identified fricatives and the types of confusions made.

In parallel, we are investigating potential benefits of this transposition algorithm in wearable hearing aids with the aim of testing the intelligibility of sentences and the ability to make grammatical distinctions (dog vs. dogs or your vs. yours). Making grammatical distinctions may be especially problematic for children growing up with a hearing impairment, who may then fail to learn such grammatical rules (Rees and Velmans, 1993). [This work was supported by the RNID].

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B28

Effect of dynamic compression characteristics on listeners' perception of reverberant speech

L.F. Shi, and K.A. Doherty, Syracuse University

The effect of different compression attack times (ATs) and release times (RTs) on listeners' perception of everyday speech remains unclear. Studies that have assessed AT/RT have used speech stimuli in quiet or in noise, which may not be sensitive enough to assess differences across AT/RT settings. Others have used settings that are not commercially available, which limits the clinical application of the results. The purpose of this study was to include both lab and field assessments of the effect of AT/RT on listeners' perception of reverberant speech.

Thirty listeners with a moderate sensorineural hearing loss participated in this study. They were monaurally fitted with a commercial behind-the-ear (BTE) hearing aid programmed in three different AT/RT settings: linear, fast, and slow. Stimuli were reverberant SPIN sentences that were pre-recorded to simulate an anechoic room, living-room, classroom, and hard hall (Sandridge et al., 2005). Listeners wrote down the last word of each sentence they heard and rated the clarity using a categorical scale. A subset of twenty listeners participated in the field study. They were binaurally fitted with the same BTE used in the lab part of the study. The BTEs were worn for two weeks, one in the fast compression setting and the other in slow/dual. Results showed that, in the lab, both fast and slow AT/RT settings yielded significantly higher speech intelligibility than linear, but no significant differences were seen between fast and slow. Also, slow AT/RT yielded a significantly higher real-ear aided response (up to 6 dB) than the fast AT/RT. In the field, higher benefit scores were obtained on the ease-of-communication and reverberant subscales of the APHAB for the fast compression compared to slow/dual. Last, at the end of the two week field study 85% of the listeners preferred fast compression over slow/dual.

B29

Predictions of speech quality under conditions of noise and distortion

Melinda C. Anderson, Kathryn H. Arehart, University of Colorado; James M. Kates, GN ReSound and University of Colorado; Lewis O. Harvey, Jr., University of Colorado

Noise and distortion produced by audio devices such as hearing aids reduce speech intelligibility and quality. The purpose of this study is to quantify and model the effects of signal degradation on the perception of speech quality in normal-hearing and hearing-impaired listeners. The stimuli were sentences subjected to eight different levels of additive noise, peak clipping, and center clipping distortion. The subjects listened to all possible comparisons of pairs of the 24 degraded sentences, and for each comparison indicated their preference. Multi-dimensional analysis is used to model the subject decision spaces. Theoretical explanations are given to describe these decision spaces. The data are also modeled using a one-dimensional metric to predict the subject quality judgments. The one-dimensional metric is an extension of the procedure developed by Kates and Arehart to model speech intelligibility. The one-dimensional and multi-dimensional analyses are able to accurately model the quality perception of both normal-hearing and hearing-impaired listeners. [Work supported by GN ReSound and Centers for Disease Control]

B30

Phase effects on masking period patterns of normal and hearing impaired listeners: Influence of stimulus presentation level

Melissa A. Woods and Jennifer J. Lentz, Indiana University

In this experiment, we compared Masking Period Patterns (MPPs) for positive Schroeder-phase, negative Schroeder-phase, and cosine-phase maskers with harmonic components presented at equal sensation levels to determine whether differences in within-period forward masking between normal-hearing and hearing-impaired listeners contribute to the shape of the MPP. Stimuli presented at the same sensation level will have a similar release from forward masking between the two groups. If differences in the release from forward masking contribute to the shape of the MPP, MPPs are expected to be similar for normal-hearing and hearing-impaired listeners when measured at the same sensation level.

MPPs were measured for six normal and six hearing impaired listeners using positive (m+), negative (m-) Schroeder, and cosine-phase harmonic maskers composed of tones ranging between 800 and 3200 Hz, with a fundamental frequency of 50 Hz. Masker stimulus levels were fixed at 15 dB SL re: audiometric thresholds for hearing impaired listeners and at 15 dB SL and 55 dB SL re: audiometric thresholds for normal hearing listeners. Detection thresholds were measured for 5-ms 2000-Hz tone bursts having onset times of 150, 154, 158, 162, or 166 ms post masker onset, thereby sampling the masker's 20-ms period every 4 ms. The "peakiness" of the MPP is described as the maximum threshold minus the minimum threshold, or the "modulation depth" of the MPP. MPPs for the normal-hearing and hearing-impaired listeners were relatively flat for the negative Schroeder-phase maskers at both stimulus levels. Hearing-impaired listeners showed slightly smaller modulation depths than normal-hearing listeners for the positive Schroeder-phase and cosine phase maskers presented at 15 dB SL. The slightly smaller modulation depths of the hearing-impaired listeners when tested at equal sensation levels supports the hypothesis that within-period forward masking plays a role in the amount of modulation present in the MPP. In addition, the cochleae of hearing-impaired listeners, despite having lost nonlinearity, still might produce a peaked stimulus representation. Results will be discussed in terms of the role that forward masking and cochlear nonlinearity play in the representation of harmonic stimuli over time.

B31

The Mandarin Early Speech Perception Test (MESP): An assessment tool for children with hearing impairment

Yun Zheng, Kai Wang, Shixi Liu, West China Hospital of Sichuan University, China; Sigfrid D. Soli, House Ear Institute

Early identification and intervention are crucial for children with hearing impairment. Age- and language-appropriate assessment of speech perception is one of the major elements in the complete audiological evaluation of children who may be hearing impaired. Eisenberg et al. (2006) recommend a hierarchy of speech tests, beginning with the Early Speech Perception (ESP) test (Moog and Geers, 1990), that span the entire period of early speech and language development; however, no such hierarchy of tests is available in languages other than English. This presentation describes the development of the Mandarin Early Speech Perception (MESP) test for use in early identification and intervention of hearing impairment in Chinese children. The MESP is comprised of five ordered subtests: speech detection, speech pattern perception, beginning word identification, word identification through vowel recognition, and word identification through consonant recognition. The words in the pattern perception subtest are distinguished by their temporal envelopes, while the words in the following subtests are distinguished by increasingly complex spectral and temporal cues. The pattern perception subtest is further divided into word sets contrasting in syllable stress and number, and in Mandarin tones 1-4. A set of 111 Mandarin words presumed to be within the vocabulary of 2-3-year-old children was selected for initial evaluation. Eleven normally hearing children (average age: 2.5 yr, SD = 0.3) were tested with these words using a live voice picture pointing task. Children were first shown pictures of each of the words and asked to name the word. Next, each word was spoken, and the child was asked to point to the appropriate picture. Words that were not within their vocabulary or that could not be accurately identified were eliminated, leaving a total of 59 words. A second sample of six normally hearing children (average age: 2.6 yr, SD = 0.1) was successfully tested with these words. A recorded version of the MESP comprised of these words is being prepared for use in the development of age-appropriate norms in China. Results of the norming study and the initial clinical experience with the MESP will be reported.

B32

Preservation or distortion of interaural cues by adaptive noise reduction algorithms for hearing aids: a perceptual evaluation

T. Van den Bogaert, L. Wouters, ExpORL, KULeuven, Belgium; T.J. Klasen, S. Doclo, and M. Moonen, SISTA-SCD, KULeuven, Belgium

Hearing impaired persons often localize sounds better without hearing aids than with their hearing aids [1,2]. This is not surprising, since hearing aid signal processing is not designed to preserve interaural cues. As an example, noise reduction algorithms currently used in hearing aids are typically designed to maximize monaural signal to noise ratio and not to preserve interaural cues. This may put the hearing aid user at a disadvantage as well as at risk. Sound localization is important in sound segregation in noisy environments (a.k.a 'the cocktail party effect'), and in other real-life situations, such as traffic.

In [3], a binaural noise reduction algorithm was presented which is specifically designed to preserve the interaural transfer functions (ITF) of a speech and a noise source. Different weighting can be applied to optimize noise reduction or interaural cue preservation. The focus of this submission is the perceptual evaluation of this algorithm in terms of localization performance, compared to other approaches, using different tuning parameters and different speech and noise scenarios. We will compare this binaural algorithm with the standard binaural multichannel Wiener Filter [4] (without ITF preservation), with an adaptive (monaural) GSC based approach and a condition without processing (omnidirectional BTE microphone signals). Localization tests are carried out with normal hearing subjects. These perceptual evaluations will be presented at the conference.

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Posters for Session C should be put up by 8 A.M. Saturday, August 19, and taken down after 10 P.M. Saturday, August 19 or before 7 A.M. Sunday, August 20. 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 10:00PM

C1

Electric acoustic stimulation – performance of EAS users of a combined speech processor, the DUET

Ilona Anderson, Marcus Schmidt, Marek Polak, Peter Lampacher, MED-EL, Austria

Electric Acoustic Stimulation (EAS) is becoming a more accepted method of treating individuals with a ski-slope type hearing loss and who gain minimal benefit from hearing aid amplification. In EAS, users are implanted with a shorted length cochlear implant (up to 18 to 20mm), which stimulates the mid to high frequency range. The low frequencies may be conveyed using the natural hearing in the cases of normal to mild hearing loss in the low frequency or may be amplified with a hearing aid, modified to amplify the low frequencies. Speech perception results in EAS users show synergistic effects between electric and acoustic stimulation especially in background noise. Improved music perception is also seen in EAS users. However, experience has shown that not all potential users used EAS due to not accepting two devices in the same ear.

To overcome this, a combined hearing aid and cochlear implant speech processor was developed to allow for EAS stimulation using one device. Design features such as compression, frequency range, signal processing and fitting have been implemented based on previous experience from EAS studies or findings in related fields.

This study aimed to examine the effectiveness of these design features in EAS users. Several design parameters have been evaluated and/or compared with the hearing aid and speech processor used by EAS patients prior to changing to the DUET. Results show equal or superior results with the DUET. Patients report on increased user comfort.

The DUET EAS hearing system is an effective device for EAS users. While the design features examined in this study are proven to be effective; an ongoing learning process is expected with growing experience in the young topic of EAS. The DUET has flexibility in both the electric and acoustic part to adapt to this learning process.

C2

Why classroom demonstrations of binaural loudness summation fail

Mary Florentine and Michael Epstein, Northeastern University

Motivation for the present experiment came from the observation that classroom demonstrations of binaural loudness summation for speech in a sound field never yielded the magnitude of the effect that was reported in the literature for tones presented via headphones (for review, see Marks, 1978 and Hellman, 1991). According to the literature, a tone or noise presented to both ears is perceived louder than the same sound presented to only one ear (Fletcher and Munson, 1933). Generally, it is assumed that this ratio is equal to two for dichotic tones at the same loudness; but some studies have suggested a lower ratio ranging from 1.3 to 1.8 (Scharf and Fishken, 1970; Zwicker and Zwicker 1991; Whilby et al., 2006). The classroom demonstrations differed from the previous experiments in stimuli (tones vs. speech), mode of presentation (earphones vs. sound field), and psychophysical procedure (magnitude estimation vs. introspection). Could it be that the stimuli, mode of presentation, or visual speech cues from the professor caused the different results in the laboratory experiments and the classroom demonstrations? To gain insight into this question, eight normal listeners were presented stimuli as a function of level with earphones and in a sound field. The stimuli were 1-kHz tones, recorded spondees without visual cues, and live spondees with visual cues. Average data show more loudness summation for tones presented binaurally via headphones than in the other five conditions. Possible reasons for these findings will be discussed. [Work supported by NIH-NICDC grant R01DC02241].

C3

A glimpse at a big picture: The prevalence of hearing aid use in adults, by gender and audiometric configuration

Greg Flamme and Lindsay Ciletti, Western Michigan University

Hearing aids are used by only a small part of the population with substantial hearing impairment. In a prior study, the probability of hearing aid use exceeded 0.5 only for those with average thresholds exceeding 45 dB HL. Along with other factors, a listener's audiometric configuration could play a role in a person's willingness to use hearing aids, and knowledge of the predominant audiometric configurations within underserved populations could inform subsequent research and development.

The purpose of this study was to examine the variety of audiometric configurations present in data from the National Health and Nutrition Examination Survey (NHANES, 1999 - 2004) and the Keokuk County Rural Health Study (KCRHS). The NHANES study was designed to provide a nationally representative sample of the health status of the non-institutionalized US population. Participants in the KCRHS consisted of 20% of the population in a rural Iowa county. All participants (3862 females, 3454 males) were divided into groups with maximally similar bilateral audiometric configurations using cluster analyses. Separate analyses for each gender were performed due to substantial gender-related differences in hearing loss magnitudes and configurations. For example, 12 clusters were necessary to describe female configurations accurately, while the number of clusters required for males was limited to 20 for practical reasons. Within each gender, the prevalence of hearing aid use will be presented, and this study's implications for rehabilitation research and development will be discussed.

C4

Helping people hear, and preserve their hearing, in noisy places

Mead Killion, Etymotic Research; Frank Dunn, FDPDCS; Jack Goldberg, Metrionix; Greg Flamme, Western Michigan University Kalamazoo; Kelly Kodama, ZOE Design

Some 10% of hearing-aid wearers have greater than 15 dB SNR loss, making it impossible for them to carry on a conversation in noise. A novel multi-talker wearable wireless "Companion Mics TM" system has been developed to provide a 15-20 dB increase in SNR for three to four talkers as heard by the listener. An estimated 98% of hearing aid wearers and 85% of cochlear implant wearers can enjoy conversations in noisy surroundings with this device. Solving a separate but related problem – safely listening to music in noisy places -- requires providing the listener with information they don't normally have: The 75-80 dBA SPL noise levels found in aircraft cabins, for example, may induce the listener using inexpensive open-ear airline earphones to increase the levels to 85-95 dBA in order to understand the movie dialog or enjoy the available music. A level of 91 dB for 4 hours exceeds by 200% the recommended daily exposure using NIOSH-1998 recommended criteria. Noise-excluding earphones help reduce risk, but some listeners appear to have become addicted to loud sounds (Florentine et al., Ear & Hearing 1998;19;420-428), and listeners seldom know when their ears are it risk. An inexpensive pocket dosimeter has been developed to provide warning for both sound-field and earphone-listening exposures.

C5

A novel approach to reduce transient noise in hearing aids: Technical concept and perceptual evaluation

V. Hamacher, E. Fischer and M. Latzel,
Siemens Audiological Engineering Group,
Germany

One of the main issues for hearing aid users is background noise, which often affects speech understanding and listening comfort. Therefore, directional microphones and noise reduction algorithms are essential processing stages for modern hearing aids. Directional microphone processing utilizes spatial information about the sound field. It enhances sound from the desired front-direction by attenuating sounds coming from other directions.

In contrast to this, noise reduction algorithms are based on the different signal characteristics of speech and noise. After decomposition of the noisy signal into several frequency bands, the early modulation based methods apply a long-term smoothed attenuation to those subbands for which the modulation of the envelope differs from patterns typical for speech. More effective recent methods continuously estimate the power density spectrum of the noise and then apply short-term attenuation factors in each subband calculated from spectral subtraction or Wiener filter formulas, which were designed to maximize the signal-to-noise ratio. Both methods have in common, that they are only effective for stationary noise; i.e., all types of noise (e.g. car noise), for which the temporal envelope fluctuation is significantly lower than for speech. Consequently, all transient types of ambient noise, e.g. hammering, rustling of paper or clanking of dishes are not attenuated. Since research has shown that transient noise is often rated by hearing instrument wearers as annoying as stationary noise, it should be attenuated by the hearing aid processing by a reasonable amount.

In this paper, a novel hearing aid algorithm for suppression of transient noise is presented. The algorithm is capable of reducing transient noise nearly instantaneously (processing delay below 1 ms). Doing this without introducing artefacts to the speech signals is the crucial point since speech itself is a transient signal. The basic processing steps are to continuously calculate the signal envelope in different frequency bands, to analyze them using a speech model and finally, to reduce the detected transient non-speech sounds. The attenuation mainly affects the loudness-carrying peaks of the transient noise. The amount of attenuation depends on the ratio of the peak level to the long-term overall RMS level (i.e. the more transient, the more attenuation). Beyond conceptual and technical insights in the basic functioning of the new algorithm, this paper will present results from perceptual studies conducted with hearing impaired at independent research sites. Overall, these data show that the described algorithm can significantly reduce the annoyance of everyday life related transient noises without adverse effects on speech intelligibility, speech quality and localization ability.

C6

The relationship between a pure tone hearing loss classification scheme, the articulation index and hearing in noise performance

Andrew Vermiglio and Sigfrid Soli, House Ear Institute

The goal of this study was to examine the relationship between categories of hearing loss, the Articulation Index (AI) for average speech, and performance on the Hearing in Noise Test (HINT). Study of this relationship may reveal the extent to which the audiogram and the AI are useful in predicting speech intelligibility in noise for individuals with hearing impairment.

In the field of audiology, the audiogram has been used as a gold standard to describe the ability to hear. However, a weak relationship exists between the pure tone hearing loss classification scheme and differences in speech intelligibility in noise performance. This disparity may have a significant impact on any evaluation the uses only the audiogram.

Audiometric and HINT data were collected from four test sites. Two hundred and seventy eight subjects were tested. HINT data were collected in a simulated soundfield under headphones using the HINT for Windows software. The HINT noise front, noise side and composite scores were used to measure speech recognition in 65 dBA speech-shaped noise.

The implications of a classification scheme based on audibility and hearing in noise performance will be discussed as they relate to test norms, the definition of control groups in hearing research, patient counseling, hearing aid fittings, CAPD classification, and hearing conservation studies.

C7

Adaptive processing enhancing sound quality

Melissa Dominguez, DBK Acoustics; Jeff Bondy & Andrew Dittberner, GN ReSound

One of the more interesting results to come out of benefit assessment for hearing aids is that several algorithms provide better sound quality and listening comfort but no benefit to intelligibility. While difficult to quantify, increasing the ease of listening, especially in noisy conditions, will make a patient more likely to accept a hearing aid. Increased cognitive load brought on by hearing impairment is often suggested as the reason for the hearing impaired person's difficulties in noise.

Unlike noise reduction circuits which started out trying to increase intelligibility of a signal, we started out trying to increase the ease of listening in noise. Our algorithm is based on leveraging the strength of the hearing impaired have for high level linguistic information while reducing the amount of confounding information in the acoustic environment. Our algorithm was implemented by capturing the microphone signal from a BTE, processing it in Matlab, and then playing it out of the same BTE's receiver. The sampling rate, bit depth, fixed point processing and noise floor are used to mimic a typical hearing aid implementation. A 2AFC sound quality comparison against six different manufacturers' devices showed an advantage not only for this algorithm against all others, but also above the unprocessed signal. HINT scores and several offline metrics are also given. Our algorithm is an altered Kalman filter, working on temporal dynamics, thus it could provide an added benefit in conjunction with spectral subtraction circuits.

C8

Low delay processing for open-ear hearing aids

Peter Blamey and Brenton Steele, Dynamic Hearing Pty Ltd., Australia

It takes longer for a signal to travel from the input of a hearing aid microphone to the eardrum of the listener than for the same sound to reach the eardrum by the direct air-conduction path. Delays for digital hearing aids reported in the literature range from 0.9 to 9.8 ms. For open-ear systems and vented earmolds, the direct and delayed signals add together at the eardrum.

The duration of the delay is important because it may affect the acoustic and perceptual properties of the combined signal. The acoustic effects include the "comb filter" effect, summation of intensity, and echo (for long delays). The perceptual effects can include increased loudness, perception of a tonal quality at the comb filter frequency, hollowness, reverberation, and echo (for long delays).

Delays may also affect the susceptibility of the hearing aid to feedback oscillations, the frequencies at which oscillations can occur, and the effectiveness of feedback cancellation and feedback suppression algorithms.

Hearing aids with delays from 0.75 to 20 ms were evaluated in studies with listeners having normal or impaired hearing. Open-ear, vented, and occluded sound delivery systems were also compared. Several aspects of performance were evaluated: sound quality for the listener's own voice, music, and speech in quiet; speech perception in noise and in quiet; and susceptibility to feedback oscillations.

The perceptual effects of delay were not as great as the perceptual effects of the different sound delivery systems. Listeners with normal hearing were more sensitive to delay than listeners with impaired hearing.

C9

Adjustments in fricative production by young children following a change in hearing aids

Sheila Pratt and Allison Hearn, University of Pittsburgh

Consistent auditory feedback is particularly important for speech development and maintenance in infants and young children (Pratt, 2005; Uchanski & Geers, 2003). As a consequence, changes in hearing aid configuration and signal processing have a high likelihood of influencing the speech production of pediatric patients.

The purpose of this study was to evaluate the impact of switching hearing aid circuitry on children's production of voiceless fricatives. Speech production data were collected from a group of 17 children while they were wearing their previously owned linear hearing aids, immediately after being fitted with more advanced technology that included WDRC, and then four weeks later. During each session the children said a series CVC words in the phrase "One, two, three______". The children's productions were recorded onto cassette audio tape and later analyzed acoustically. For this study, four words with initial unvoiced fricatives (feet, seat, sheet, heat) were extracted from the pre-fitting, fitting, and 4-week-post recordings. In addition, the word beet was included as a non-fricative comparison. The spectral and relative intensity levels of the fricatives and following vowel were assessed.

The acoustic characteristics of the children's speech were altered consequent with the changes in the children's amplification.

Some of the differences, like those observed with spectral variance, were evident only at the fitting, while most were consistent with a more systematic response to the change in amplification. Similarly, coarticulation of the voiceless fricatives, as measured with locus equations, appeared to deteriorate on the day of the fitting but recovered by the 4-week session. In this sample the sh-sound seemed to be the most sensitive to changes in amplification and likely to change.

C10

Health-related quality of life and hearing: Results from a multi-center study

Martin Kinkel, KIND Hoergeraete, Germany; Markus Meis, Hoerzentrum Oldenburg, Germany; Karsten Plotz, Hoerzentrum Oldenburg and Ev. Krankenhaus Oldenburg, Germany; Norbert Dillier, Univ.-HNO-Klinik Zuerich; Juergen Kiessling, Univ.-HNO-Klinik Giessen, Germany; Horst Hessel, Cochlear GmbH, Germany

In view of the demographic changes, sensorineural hearing impairment is increasingly becoming a serious social problem. The psychosocial effects of untreated hearing impairment often result in social withdrawal and cognitive deficits as well as a decrease in quality of life both in occupational and private activities. These effects can be positively influenced by providing hearing aids or cochlear implants.

As part of a cross-sectional multi-center study, a survey was conducted involving 2,260 hearing impaired persons, including 910 CI patients and 1,500 patients with different types of hearing impairment and treatment. Quality of life was measured using generic instruments taken from research conducted on health-related quality of life as well as from the HoerTech Fragebogeninventar© (HoerTech inventory), i.e. questionnaires specifically designed for hearing aid fitting. Out of 1,000 subjects, 358 were fitted unilaterally, 323 bilaterally, 94 bimodally (CI and hearing aids), and 225 were not fitted with either system. A significant benefit from bilateral fitting was seen in all profiles in several subscales.

C11

The efficacy of an asymmetrical directional-microphone fitting

William Whitmer, Andrew Dittberner, Maureen Coughlin & Melissa Dominguez, GN Auditory Research Laboratory

Several recent studies have shown that a bilateral hearing-aid fitting with omnidirectional microphone processing in one ear and directional microphone processing in the other, an asymmetrical fit, can provide both the directional advantage of bilateral directional fit in the laboratory with the ease-of-listening preference of bilateral omnidirectional devices in the field. In another recent study, it was found that many users preferred a unilateral omnidirectional fitting in loud situations to bilateral omnidirectional or bilateral directional. A set of experiments tested the application of these findings. Hearing-impaired listeners gave their setting preferences under realistic normal-level and loud conditions for unilateral-omnidirectional, bilateral-asymmetrical, or bilateral-directional modes. Listeners switched between settings while listening to single and multiple speech targets in diffuse background noise. Microphone-processing preferences were dependent upon listener hearing loss and age, but were similar to previous findings. A post-hoc analysis of stimulus-preference interactions using a modified form of the speech-transmission index showed a modest ability to predict preference based on acoustic environment.

C12

Learning in repeated measurements of speech reception thresholds for sentences in speech-spectrum noise

E. William Yund, VA Northern California Health Care System; David L. Woods, VA Northern California Health Care System and University of California, Davis

Stable measurements of the speech perception in the same individuals are critical for studying the objective benefits of hearing aids, acclimatization and auditory rehabilitation training in the hearing impaired. In a previous study [Wilson, Bell & Koslowski (2003), J. Rehabil. Res. Dev. 40, 329-336], sentence thresholds in quiet improved across five sessions, showing large improvements even for sentences that were never repeated. Here, we investigate the measurement stability of the Hearing In Noise Test (HINT) in normal-hearing individuals over test sessions on five different days within a period of two weeks. For each subject, some of the HINT sentence lists were repeated in all five sessions, while others were presented uniquely in either the first or last session. Although repeated HINT lists showed consistent performance improvements over the five sessions, performance on unique HINT lists was the same in the first and last sessions. Factors that may account for differences in results from the previous study include (1) higher-intensity speech, (2) testing in noise, and (3) test procedures. Independent of the explanation for the differences, the present results indicate that the HINT provides stable measures of speech perception in noise as long as the same HINT lists are not repeated for the same subjects. [This work was supported by VA RR&D.]

C13

The importance of binaural listening in adverse acoustic environments: Impact of signal processing requirements

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Binaural hearing has increasingly become a hot topic among hearing aid researchers and manufacturers. Binaural hearing refers to the sensing of sounds by the two ears, physically separated by an inter-cranial distance of approximately 15 cm in the average human. The physical locations of the two ears give rise to intensity and time of arrival differences dependent on sources locations. It is well established that binaural hearing influences the ability to understand speech amidst multiple competing sounds, as well as producing situational awareness. As well as, being important in themselves, they also affect safety, naturalness, and comfort.

The poster will illustrate the disruption of spatial cues by directional hearing aids, particularly how localization cues are further degraded by reverberation. In addition, it will review ideas based on combining microphone responses located on each side of the head in order to improve signal-to-noise ratio in adverse listening situation, e.g. multiple competing sounds and reverberant environments. A sound demonstration will include conventional beamformers, and blind source separation based on correlation analysis between signals arriving at the two ears, recorded in a dummy KEMAR head. Finally, sound recordings whereby a pre-recorded multi-talker diotic sound such as the result of a blind source separation and a restored spacialised sound with similar objectively measured signal-to-noise ratio will demonstrate the importance of binaural (dichotic) listening. [Ongoing research work supported

by the CRC for Cochlear Implant and Hearing Aid Innovation, Australia, The National Acoustic Laboratories, Australia and The University of Sydney, Australia].

C14

Reference audiograms for programming hearing aids to be measured by new speech amplification measurement procedure

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The performance characteristics of modern hearing aids are primarily set by the programming software using an audiogram as input. The fitting rule applied transforms the audiogram data in to amplification settings. A novel approach to defining reference conditions for hearing aid characterization uses a set of standard audiogram representative of typical hearing aid users. This approach is built into a new set of measurement procedures to be proposed.

Analysis of a large database of audiograms results in the extraction of a set of typical audiograms that covers the entire spectrum of losses.

C15

A "level playing field" analysis of hearing aid functions across manufacturers

Jeff Bondy and Andrew Dittberner, GN ReSound; Todd Ricketts, Vanderbilt University; Melissa Dominguez, DBK Acoustics

The complexity of digital hearing aids, the concealed calculations in fitting software and the lead time of standards has made it increasingly hard to judge one hearing aid against another. The inability to compare hearing aids against one another has allowed underperforming devices to be given to patients. Audiologists can unknowingly fit patients with 3 dB less real-world directivity, or drop audibility by activating feedback suppression. We have developed a benchmarking strategy that allows the audiologist to assess the perceptual benefits of different algorithm packages across manufacturers to aid them in providing the best advantage to their clients.

Our testing includes analysis of feedback suppression, adaptive directivity and noise reduction. For feedback suppression, there are large differences in the amount of headroom and the sound quality of each device. Headroom, useable gain, HINT scores, 2AFC blind sound quality assessment, and signal fidelity scores are presented. For directivity each device underwent a diffuse field test with white noise and multi-talker babble, a single source, adapted response measurement, and two new measures for multiple talker locations. The perceptual efficacy of noise reduction is also measured. The metrics presented are much more consistent with real-world benefit than numbers typically advertised. No manufacturer names are presented, the protocol and range of measures is hoped to give the reader an apple to apple comparison of the field.

C16

Cognitive effects of noise reduction

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Signal processing strategies that attempt to aid listening by improving the signal-to-noise ratio have been used in assistive auditory devices for decades, yet there is little evidence that they are effective in improving the speech reception threshold. One possible explanation for the lack of benefit of noise reduction algorithms is that they are performing an essentially similar function to the natural processing of the listener's auditory system. In other words, a listener faced with a noisy speech signal performs a type of signal extraction or noise reduction, as long as cognitive resources are available. Applying a signal processing strategy is often redundant, as it provides little more information than what the listener's system has already extracted. We propose that the benefits of signal processing in auditory devices may be in reducing cognitive effort in situations with a high cognitive demand, such as when multiple tasks are performed simultaneously. To that end we have used the dual-task paradigm to evaluate different noise reduction strategies. The auditory task involves listening to and repeating sentences or words in a noisy background. Two competing tasks where developed in order to investigate the effects of noise and noise reduction on cognitive effort. The first task was a simple driving game presented on a computer, while the second task was a memory test. Using performance in the competing task as an indicator of mental effort, we evaluate different noise reduction algorithms, both in their effect on speech intelligibility and mental effort.

C17

Comparison of test results between automated and manual audiometry using the tympany otogram

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Automated audiometry is a technology that is coming of age, but questions still arise about how comparable automated results are to results obtained using manual audiometry. A study comparing thresholds measured manually by audiologists and automatically using a computer-based system (the Otogram) is underway to determine the comparability of manual versus automated audiometric scores. As a part of this study, multiple tests on the same individuals will be run on the Otogram to evaluate test-retest repeatability on an automated system. For both sets of data, results will be statistically analyzed using paired comparisons to determine the level of variability between test results.

C18

The objective and subjective evaluation of multi-channel expansion in wide dynamic range compression hearing instruments

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Rationale:

Research has demonstrated that single-channel expansion improves subjective evaluations for WDRC hearing instrument users (Plyler et al., 2006, 2005a) despite degrading the recognition of low-level speech (Plyler et al., 2005a, 2005b). Expansion effects may differ, however, if multi-channel expansion is used instead of single channel expansion. Specifically, expansion parameters such as expansion threshold and expansion ratio may vary across channel in multi-channel systems in order to reduce amplification in restricted frequency regions as opposed to across the entire spectrum. Consequently, multi-channel expansion may preserve audibility of high frequency speech cues necessary for accurate feature identification while reducing amplification for low-level environmental noises and noises generated by the hearing instrument. As a result, multi-channel expansion may maintain subjective improvements for WDRC hearing instrument users without degrading the recognition of low-level speech; however, the effects of multi-channel expansion on the objective and subjective evaluation of WDRC hearing instrument users remain unknown. Therefore, the purpose of the present study was to determine if multi-channel expansion affected the objective and subjective evaluation of WDRC hearing instruments used for a two-week trial period.

Methods:

Twenty current hearing instrument users were fitted binaurally with four-channel digital ITE products. Three memories of the hearing instruments were programmed for each subject. Each instrument had one memory in which expansion was activated in all four channels (four-channel), one memory in which expansion was restricted to channels one and two only (<2000 Hz) (restricted), and one memory in which expansion was deactivated in all four channels (off). All other fittings parameters were held constant across the three memories. Each subject utilized the hearing instruments for a two-week period before returning to the laboratory for the objective and subjective evaluations. Objective evaluations were conducted in quiet using the Connected Speech Test and in noise using the Hearing in Noise Test at 40, 50, and 60 dB SPL. Subjective evaluations were conducted by having each participant (a) rate their satisfaction regarding the amount of noise reduction they perceived daily and (b) indicate which expansion condition they preferred overall after a two-week trial.

Results:

Listeners performed significantly better in quiet (CST) and in noise (HINT) for the off condition than for either multi-channel condition; however, restricting expansion to channels one and two improved objective performance in quiet and in noise relative to the four-channel condition. Conversely, satisfaction ratings were significantly greater for both multi-channel conditions than for the off condition; however, satisfaction ratings were similar for the restricted and the four-channel conditions. Overall, listeners preferred any form of multi-channel expansion to no expansion; however, overall preference was similar for the restricted and the four-channel conditions.

Conclusions:

Hearing instrument users prefer the use of multi-channel expansion despite the fact multi-channel expansion may significantly reduce the recognition of low-level speech in quiet and in noise. Although restricting expansion to channels one and two (i.e., 2000 Hz and below) maintained subjective benefit for WDRC hearing instrument users, the recognition of low-level speech was not completely preserved.

[The authors would like to acknowledge the support of Starkey Laboratories for providing the hearing instruments.]

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C19

Effect of speech presentation level on acceptance of noise in full-time, part-time, and non-users of hearing aids

Melinda Freyaldenhoven, Patrick Plyler, James Thelin, Mark Hedrick, and Schuyler Huck, The University of Tennessee

Acceptable noise level (ANL) is a measure of willingness to listen to speech in the presence of background noise. Previous ANL research has demonstrated that ANL is directly related to hearing aid use. Specifically, individuals who accept high levels of background noise (i.e., have small ANLs) are likely to become successful hearing aid users (i.e., wear hearing aids on a full-time basis), and individuals who cannot accept background noise (i.e., have large ANLs) are likely to become unsuccessful hearing aid users (i.e., wear hearing aids part-time or not at all) (Nabelek et al, 1991). It should be noted that ANLs are conventionally measured at one speech presentation level, the listeners' most comfortable listener level (MCL). When measuring ANLs in this manner, a prediction as to whether a listener will be a successful or unsuccessful hearing aid user can be made with approximately 85% accuracy (Nabelek et al., in press).

There are several potential limitations to the current prediction of hearing aid use. First, the model assumes that in daily listening situations, hearing aid users only listen at one level, their MCL; therefore, the model gives no information on how background noise acceptance is changed if the signal of interest is above or below MCL. Second, there are two groups that make up the unsuccessful hearing aid user group: part-time hearing aid users and non-users of hearing aids. Part-time hearing aid users are defined as individuals who wear their hearing aids occasionally, and non-users are defined as individuals who never wear their hearing aids. These two groups cannot be differentiated using a single ANL measure. Third, although successful hearing aid use can be predicted with 85% accuracy, an accurate prediction cannot be calculated for 15% of the population. For example, the most frequent ANL value is 10 dB, and the current predictive model indicates a 50% probability of success for listeners with ANLs of 10 dB. Therefore, the use of a single ANL measure may be insufficient for listeners with the most common ANLs. Consequently, the purposes of the present study were to determine if ANL measured at high and low speech presentation levels (i) could differentiate the three hearing aid groups (full-time, part-time, and non-users), and (ii) could better predict hearing aid use than a single ANL measure.

Twenty-five full-time, 21 part-time and 23 non-users of hearing aids participated in this study. Unaided ANLs were measured at MCL (i.e., conventional ANL) and at speech presentation levels of 40, 45, 50, 55, 60, 65, 70 and 75 dB HL. The effect of speech presentation level on ANL was evaluated in two ways: (i) averaging ANL values at all speech presentation levels (called global ANL) and (ii) calculating the slope of the ANL function across the fixed speech presentation levels (called ANL growth). The results demonstrated that global ANLs differentiated the three hearing aid groups in the same manner as conventional ANL; however, ANL growth differentiated full-time hearing aid users from non-users only. Individual data analysis further demonstrated that both global ANL and ANL growth may better differentiate part-time hearing aid users from the other two groups when compared to conventional ANL. These results indicate that the effects of speech presentation level on acceptance of noise show potential in differentiating part-time hearing aid users from the other two hearing aid user groups. The results further demonstrated that conventional ANL predicted hearing aid use with 68% accuracy.

Furthermore, the accuracy of the prediction decreased for both global ANL and ANL growth. These results indicated that the predictive probability of ANL measured at multiple speech presentation levels was comparable to the prediction obtained for conventional ANL.

C20

Speech recognition in noise is correlated to cognitive function

Thomas Lunner & Elisabet Sundewall-Thorén, Oticon A/S, Denmark

Results from speech recognition-in-noise tests typically show large variations across subjects, although the pure-tone hearing thresholds may be of similar magnitude. In a previous study (Lunner, 2003) it was shown that performance on a visual working memory test predicted speech recognition performance after accounting for pure-tone hearing thresholds.

Here we present results from more than 200 test subjects with moderate sensorineural hearing loss. The subjects' performance in sentence-in-noise tests (Hagerman sentences, Dantale II, and HINT) were measured, both under aided and unaided conditions, as well as under unmodulated and modulated background noise conditions. Also, we present results from cognitive tests, which all measure the ability to simultaneously process and store incoming, visual stimuli (Reading Span, Visual Word Monitoring Score, and the Rhyme test). Results show correlations typically in the range .40 to .70 between speech recognition in noise performance and cognitive performance.

This indicates that individual cognitive performance may explain at least a part of the variance seen in speech-in-noise tests across clinical samples of hearing impaired hearing aid users. Assessment of cognitive function may therefore be an important predictor for real-world performance in difficult communication situations.

Lunner, T. (2003). Cognitive function in relation to hearing aid use. Int.J.Audiol., 42 Suppl 1, S49-S58.

C21

Histogram analysis of occlusion effect measurements during speech

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The occlusion effect is a well-known problem for hearing aid users. The users often perceive their own voice as booming and hollow when their ears are occluded by earmolds. Usually the occlusion effect is remedied by venting, which attenuates the low frequency part of any sound source within the ear. Conventionally, the occlusion effect is measured by use of sequential single vowel vocalisations. However, as this study will show, the single vowel method is not exhaustive and imprecise since the occlusion effect depends on many different aspects of the signal.

We present systematic measurements of the variables of the objective occlusion effect during running speech. The measurement method comprises a simultaneous bilateral measurement, with one ear occluded and the other open. The sound pressure is simultaneously monitored by use of a probe tube microphone in each ear while the user e.g. reads aloud from a text passage. The objective occlusion effect is calculated as the ratio between the time-frequency spectra of the signal in the closed ear relative to that of the open ear. For the measurements to be representative the probe tube must be inserted at equal depth in each ear and the ears must be symmetric. Measurements prove that this assumption holds at low frequencies.

The simultaneous bilateral measurement offers a unique opportunity to analyse the occlusion effect as a function of time. The temporal aspect of the occlusion effect is implemented in the analysis by use of a histogram approach. This histogram analysis depicts the distribution of the occlusion effect at each frequency instead of the conventional single value. In this way, not only the average frequency dependent occlusion effect is observed from the data, but also the spread is assessed. Furthermore, by discarding non-speech time segments, the result of the method is made independent of pauses in the speech, coughs, swallowing etc.

The measurements show that the occlusion effect has complex dependencies. The well-known dependencies on insertion depth, vent size, and vowel type are reproduced by the measurements. Moreover, it is shown that variables such as pitch and vocal tract area function also affect the amount of occlusion perceived by the user.

Furthermore, the objective occlusion effect measurement is correlated with laser vibrometer measurements of the open ear canal wall movements during vowel vocalisation. These measurements show good correspondence between the amount of occlusion effect and the vibrations of the ear canal wall at different vowels.

C22

The benefit of high-frequency bandwidth for hearing-impaired people under speech-in-noise conditions

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Through a number of years there has been some discussion of whether hearing-impaired people can benefit from an increased high-frequency (HF) bandwidth of a hearing aid. At the same time the general development in the hearing aid industry has been towards increasing the bandwidth of the hearing aids. This development has perhaps been more technologically motivated than it has been audiologically motivated.

One area where such an increased bandwidth might give advantage could be under speech-in-noise conditions. The purpose of this study was to test the hypothesis that increased HF bandwidth improves speech recognition in noise and results in less perceived effort when doing the task.

Eleven subjects participated in the experiment. All subjects were fitted binaurally with the hearing aids. All subjects had mild-moderate sloping/ski-slope hearing losses.

Two types of hearing aids were tested – one with a HF cutoff at approximately 8 kHz and one with a cutoff at approximately 6 kHz. The hearing aids had equal gain characteristics except for the bandwidth. Both hearing aids were fixed in a cardioid directional pattern and their noise management systems were active. It was verified that the noise management systems of the hearing aids were damping the stimuli equally.

The experiment was conducted according to the procedure known as the Ease-of-listening test [Behrens et al. 2005]. The task of the subjects was to recognize Danish Hagermann sentences in modulated ICRA noise presented at 70 dB SPL with individual signal-to-noise ratios. The recognition percentage was recorded. After the session with each of the two hearing aid types, the subjects were asked to rank the perceived effort involved in doing the task. This was done on a scale going from “no effort” to “largest possible effort” with no ticks in between.

The results show that on average the 6 kHz hearing aid was ranked at 73% of max effort, while the 8 kHz hearing aid was ranked at 60% of max effort. This difference is significant according to a pair-wise one sided t-test with $p=0.001$. In terms of recognition percentage the 6 kHz hearing aid resulted in an average recognition percentage of 83%, while the 8 kHz hearing aid resulted in an average recognition percentage of 88%. This difference is significant according to a pairwise one sided t-test with p=0.001. So the hearing-impaired subjects found it much easier to follow the speech using the hearing aid with the highest bandwidth and they also performed better with it.

[Behrens et al. 2005] Thomas Behrens, Elisabet Sundewall & Martin Vestergaard (2005): Test of a Method for Measuring Listening Effort, External Research Report 052-08-03, Oticon A/S Research Centre 'Eriksholm', Kongevejen 243, DK-3070 Snekkersten, Denmark.

C23

Longitudinal trends in subjective measures of benefit and satisfaction

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The question of whether or not auditory acclimatization occurs for self-report outcome has not been resolved. While some studies provide evidence in favour of auditory acclimatization for self-report outcome, several studies have failed to demonstrate auditory acclimatization in the self-report domain. However, in most studies where no acclimatization was found, the evaluations did not start until 3 to 6 weeks post-fitting. Kuk et al. (2003) failed to show evidence of acclimatization in the self-report domain, and concluded that future studies should examine self-report benefit and satisfaction closer to the initial fitting. Munro & Lutman (2004) reported a study in which one of their subject groups showed a statistically significant improvement in self-report outcome, while in another group, using a different version of the same questionnaire, no such improvement was observed. They suggested that self-reporting is unfit for measuring changes in auditory performance over time because the result will depend on the type of questionnaire used.

To unravel some of the divergent findings regarding the existence of auditory acclimatization in the self-report domain, I administered several self-report questionnaires repeatedly over time starting immediately post fitting. The study investigated the extent to which self-report benefit and satisfaction changed over 13 weeks, and it also assessed the validity of and relationships between various measures of self-report outcome. Four outcome inventories and a questionnaire on auditory lifestyle were given to 25 hearing-aid users, and assessments took place 1 week, 4 weeks and 13 weeks after hearing-aid provision. The results showed that, for data collected early post-fitting, some subscales have no face value. Furthermore, for first-time users who used their hearing aids more than 4 hours per day, self-report outcome improved over time in some scales, although there was no change in amplification during this time. These results indicate that the way in which hearing-aid users assess outcome changes over time. The results are consistent both with studies where changes were found and where no changes were found in self-report outcome; in fact they can explain why in some studies changes over time are not observed while in others they are. The practical consequence of the results is that early self-report outcome assessment is misleading for certain self-report outcome schemes. [The work was funded by the William Demant Foundation]

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C24

Development of an audio-visual dual-sensory assist device using mutual information optimization

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Allowing listeners to observe the face of a speaker enhances their ability to understand speech in noise. One way in which visual information contributes to speech understanding is through the common modulations occurring in the audio and visual signals that may be integrated in the central processing systems of the listener. The time-linked characteristics of auditory and visual information reflected in the speech signal form the basis for the algorithm we are developing into a new technology to improve speech intelligibility and sound localization in noisy environments. We will describe a prototype system that is currently under development. The inputs to the system consist of video recording of a human speaker and dual-channel speech recorded from microphones mounted within a KEMAR manikin's ears. A mathematical model has been built that creates audio and video filters using the maximum mutual information between the audio and video streams as the optimality criterion (Fisher et al. 2000, Advances in Neural Information Processing, 13). When data pass through the audio filter, only sounds that correlate with the video are presented to the listener's ear. Concurrently, the video filter created using mutual information may be coded into the audio-stream to provide auditory cues relating to sound localization. We will discuss the development of the optimization algorithm and the improvements achieved in signal-to-noise ratio of speech and localization of sound point sources. These results will be compared with outcomes from a statistical method of fusing audio-visual data using canonical correlation analysis (Kidron et al. 2005, Proc. IEEE Computer Vision & Pattern Recognition, 1:88-96). [This work was supported by NCRAR grant VA RR&D C2659C].

C25

Frequency-specific expansion of temporal cues for lexical-tone identification in Cantonese

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Our previous research showed that the temporal envelope and periodic components (TEPC) in the high-frequency region of speech (1.0-4.0 kHz) are more important for lexical tone identification (ID) than those components in the low-frequency region (<0.1 kHz). This suggests that lexical tone ID in Cantonese can be improved by manipulating TEPC in frequency-specific regions. The present research aims at a new signal processing method that can increase the salience of the TEPC cues and improve lexical tone ID for the users of hearing prosthesis.

Two listening tests have been carried out using different lexical-tone stimuli. In both cases, the stimuli contain Cantonese lexical tones with the phonetic contexts of /ji/ and /wai/, uttered by one male and one female speakers. The stimuli were processed by filtering the full-band speech signal into four or five frequency bands (<0.5 kHz, 0.5-1.0 kHz, 1.0 -2.0 kHz, 2.0 -4.0 kHz and 4.0-8.0 kHz), followed by full-wave rectification, and low-pass filtering (cut-off at 0.5 kHz) to extract the TEPC from each band. The TEPC were nonlinearly expanded in the time domain. In Test 1, the manipulated band-specific TEPC were used to modulate speech-spectrum-shaped white noise. In Test 2, the processed TEPC were used to modulate the original intact fine structure within the respective frequency bands. The stimuli were masked by background noises at a low signal-to-noise ratio in Test 2. In both tests, different combinations of TEPC-modulated frequency bands were presented to the subjects for lexical tone ID.

Ten Cantonese-speaking subjects (5 males and 5 females) with normal hearing participated into Test 1, and another 5 males and 5 females participated into Test 2. In Test 1, it is observed that the accuracy of lexical tone ID was the best with expanded TEPC in the high-frequency bands (1.0-2.0 kHz and 2.0-4.0 kHz) and the worst when the TEPC-modulated noises were presented in the low-frequency bands (<0.5 kHz and 0.5-1.0 kHz). In Test 2, the stimuli with expanded TEPC in the high-frequency bands above 1.0 kHz were most accurately perceived and those with expanded TEPC in the low-frequency bands below 1.0 kHz gave the worst accuracy. These findings suggest that TEPC expansion in high-frequency regions is a potentially effective way to improve the lexical tone ID for hearing prosthesis and could possibly be utilized in hearing aids or cochlear implant devices for speakers of tonal languages such as Cantonese or Mandarin.

C26

The effect of noise vocoder signal processing on consonant recognition in normal hearing and hearing-impaired listeners in noise

Yang-soo Yoon, David M. Gooler and Jont B. Allen, University of Illinois at Urbana-Champaign

One of the factors that accounts for the high performance in speech recognition for cochlear implant listeners in quiet is the high transmission rate of speech temporal information that is processed by the noise vocoder. A little attempt is made to measure the effect of the noise vocoder signal processing on the recognition for individual consonants in noise. Thus, the goal of this study was to compare the recognition on each of 16 consonant-vowel syllables (CV), processed by the noise vocoder, both in hearing-impaired listeners (HI) and normal-hearing listeners (NH) as a function of signal-to-noise ratio (SNR). For each CV sound, the temporal envelope of speech was extracted from each of 26 critical bands by the Hilbert transformation. Further, the processed stimuli were temporally degraded by low-pass filtering (LPF) to produce four conditions: No LPF, 16Hz LPF, 8Hz LPF, and 4 Hz LPF. Performance was measured in the form of confusion matrices. Results showed that the magnitude of the mean percent error P_e *(SNR)* was a function of the LPF cutoff frequency applied to the temporal envelope and the SNR for both NH and HI. As expected, P_e *(SNR)* in NH < P_e *(SNR)* in HI for all stimulus conditions across SNR. For both NH and HI, four consonants (/Tha/ as in thin, /tha/ as in that, /va/, and /fa/) were perceived least accurately, that is, $P_e^{(i)}$ *(SNR)* on each CV > P_e *(SNR)*, regardless of stimulus conditions at each SNR. In contrast, /sha/, /jza/, /ma/ and /na/ were the most robust in noise, that is, lower P_e *(SNR)*) at SNR ≤ 0 dB. Consonants /ka/ and /za/ were the most susceptible to noise; the difference in P_e *(SNR)* was the greatest between SNRs of –8 dB and 8 dB. The results also showed that a similar subset of additional CVs (/ba/, /da/, /ga/, pa/, and /ta/) were systematically affected by LPF of the temporal envelope, but the recognition performance of these subsets of CVs differed for the two groups of listeners.

C27

Signal-to-noise ratio loss and consonant perception in hearing impairment under noisy environment

Yang-soo Yoon, David M. Gooler and Jont B. Allen, University of Illinois at Urbana-Champaign

It is known for hearing-impaired (HI) listeners that the ability to understand speech in noise is not entirely dependent on their hearing threshold. It is also known that the signal-to-noise ratio (SNR) loss (also called speech loss) is more correlated with the speech understanding in noise in HI listeners. The goals of this study were to (1) measure the relationship between the hearing threshold and SNR loss and (2) to compare SNR loss with the recognition performance on 16 American English consonant-vowel (CV) nonsense syllables as a function of SNR (-12, -6, 0, 6, 12, & Quiet). 16x16 confusion matrices were collected and analyzed from 16 sensorineural HI listeners. Results indicated a systematic relationship between SNR loss and hearing threshold. The distribution of SNR loss as a function of pure-tone average (PTA) over .5, 1, 2, & 4 kHz showed that 16 listeners could be divided into two groups: SNR loss > 5 dB was associated with PTA > 40 dB HL; the majority of listeners with SNR loss < 5 dB had a PTA < 40 dB HL. Furthermore, the group with > 5 dB SNR loss showed a wider range of SNR loss (> 10 dB), whereas the group with < 5 dB SNR loss had a range of 3 dB. However, performance on CV recognition, the mean percent error Pe(SNR) is not correlated with SNR loss. No significant difference in Pe(SNR) was observed between the two groups across SNRs, but a significant main effect of SNR was observed in all groups. Pe(SNR,SNR loss) for individual listeners showed a weak correlation with SNR loss.

C28

Psychometric per utterance and per listener confusion patterns in listening experiments

Andrew Lovitt and Jont Allen, University of Illinois at Urbana-Champaign

The Miller Nicely 1955 experiment was repeated, as closely as was physically possible, however results were analyzed on a per utterance (e.g., talker), per listener, basis. Using information theoretic measures, the results illustrate that responses, in some cases, can vary significantly across talkers and listeners. For many of the utterances, as the SNR is lowered, some identifications change systematically into another sound (denoted morphing). For some utterances all listeners report the same result, while for others the subjects formed response groups. These per-talker and per-listener differences represent specific processing strategies, that are critical to understanding the data.

C29

Objective measures of auditory stream segregation in normal-hearing and hearing-impaired listeners using multi-tonal complexes

Deanna S. Rogers, Jennifer J. Lentz, Indiana University

This study investigated the effects of hearing loss on broadband auditory streaming abilities using an objective paradigm. An alternating sequence of stimuli (A_B_A_B_A_B…), where A and B represent 6-component, multi-tonal stimuli and _ denotes a silent interval, were used. Frequencies in the A stimulus ranged between 1000 and 4000 Hz (generated at a frequency ratio of 1.32), whereas the initial frequency for the B complex was 1000, 1030, 1060, 1080, 1100, 1200 or 1400 Hz (also generated at a frequency ratio of 1.32). The A and B stimuli were 60 ms in duration with 20 ms ramps, and the silent interval was 60 ms in duration in the standard stimulus. Each standard sequence was composed of 11 A_B_ cycles, leading to an overall duration of about 1.5 sec. Signal sequences differed from the standard sequence in that after the 5th cycle in the sequence, a delay (Δt ms) was added to the silent interval, thus increasing the duration of the silent interval. Over the next four AB cycles, the delay was progressively increased by an additional Δt, leading to a final cumulative delay of 4Δt on the 9th A_B_ cycle. During the final two cycles the silent interval reverted back to the original duration (60 ms). A 2AFC procedure was used in which subjects indicated which of two stimulus intervals contained the temporal change. A 2-down, 1-up procedure was used to estimate thresholds for each of the starting frequencies of the B complexes. For both groups of listeners, thresholds (Δt) increased with increases in the initial frequency of the B complex. Thresholds also were higher for the hearing-impaired listeners than for the normal-hearing listeners for the conditions in which the B stimulus differed from the A stimulus (i.e. all initial B frequencies except 1000 Hz). The higher thresholds for the hearing-impaired listeners than for the normal-hearing listeners suggest that hearing-impaired listeners are having more difficulty comparing temporal changes across frequency regions. However, thresholds were not different between the two groups for the condition in which the A complex and the B complex had the same frequencies (i.e. the initial B frequency of 1000 Hz). Because thresholds are similar here, within-channel temporal processing deficits cannot explain the higher thresholds for all other conditions. Results will be discussed in terms of how this difficulty comparing temporal changes across frequency channels might lead to difficulties segregating and grouping sounds in a complex environment.

C30

Speech enhancement algorithm for hearing aid users based on modified spectral subtraction and companding

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Generally, digital hearing aid users often complain of difficulty in understanding speech in the presence of background noise. Therefore, various speech enhancement algorithms have been applied in digital hearing aids for improving speech perception. In this study, a speech enhancement algorithm utilizing modified spectral subtraction and companding is proposed for digital hearing aid users. The spectral subtraction algorithm has been used for enhancing speech corrupted by background noise. The companding based on two tone suppression has been used for spectral contrast enhancement. We combined two algorithms sequentially for noise reduction and formant enhancement. A crucial component of spectral subtraction is an estimation of noise power and a subtraction rule. The basic spectral subtraction algorithm employs a speech activity detector to update noise statistics. Therefore, tracking of varying noise levels might be slow and confined to periods of no speech activity. It is not proper for hearing aid user in real environment that noise characteristics change continuously. We applied minima controlled recursive averaging (MCRA) noise estimation that track the noise spectrum well in the non-stationary noise and modified a bias of noise spectrum for decreasing the musical residual noise. The biases of each frequency bin were adjusted by the SNR between the smoothed speech power spectrum and estimated noise spectrum per each frame. The formants of speech with noise suppression were obtained using linear prediction coding (LPC) and peak extraction algorithm. The extracted formants were enhanced using companding. Objective and subjective evaluation under various environmental conditions (such as white noise, car noise, factory noise and speech babble from -5dB to 10dB SNR) confirm the profit of the proposed algorithm. The noise was taken from the NOISEX'92 database and the speech signal was taken from K-HINT database. Noise suppression was achieved, while retaining weak speech components and avoiding the musical residual noise phenomena. The results suggest the proposed speech enhancement algorithm may be beneficial for hearing aid users in noisy listening.

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