IHCON
2004

International Hearing Aid Research Conference 2004

August 25 - 29, 2004

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LAKE TAHOE, CALIFORNIA

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Student Scholarship Recipients

Name	Institution	Field of Study
Mitzarie Carlo	VA Medical Center, Bay Pines	Digital Noise Reduction Strategies
Suzanne Carr	Boston University	Speech Intelligibility
Rong Dong	McMaster University, Canada	Noise Reduction/Speech Enhancement
Jason Galster	Vanderbilt University	Directional Microphones
Melanie Gregan	University of Minnesota	Psychoacoustics
William Hodgetts	University of Alberta, Canada	Prescriptive Fitting
Lorienne Jenstad	University of Washington	Hearing Aid Prescription Protocols
Karolina Kluk	University of Cambridge, UK	Psychoacoustics
Courtney Lane	Rice University	Hearing Aid Algorithms
Sheng Liu	University of California, Irvine	Signal Processing
Ewen MacDonald	University of Toronto, Canada	Visual Speech Cues
Srikanta Mishra	All India Institute of Speech & Hearing, India	Pediatric Device Fittings
Elisabet Molin	KTH, Signals, Sensors, and Systems, Sweden	Auditory Models
Kartik Narayanan	All India Institute of Speech & Hearing, India	EMI Protected Hearing Aids
Hua Ou	University of Iowa	Music Perception
Jessica Rossi-Katz	University of Colorado at Boulder	Speech Perception
Arocena Alejo Suarez	LAFIVES, School of Medicine, Uruguay	Audiology Pathophysiology
Tim Van den Bogaert	Katholieke Universiteit Leuven, Belgium	Bilateral Signal Processing
Keiichi Yasu	Sophia University, Japan	Signal Processing
Muhammed Zilany	McMaster University, Canada	Signal Processing

Daily Schedule

WEDNESDAY, AUGUST 25

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

THURSDAY, FRIDAY & SATURDAY, AUGUST 26-28

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 29

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 25

WELCOME AND KEYNOTE ADDRESS

7:30PM – 8:45PM

Welcome Remarks: Sig Soli
Larry Humes

KEYNOTE ADDRESS

Tammo Houtgast	Speech reception in noise by hearing impaired listeners

THURSDAY, AUGUST 26

SESSION ONE

8:00AM – 9:45AM

AUDITORY PHYSIOLOGY, AGING, AND HEARING IMPAIRMENT

Moderator: Eric Young

Richard Schmiedt	Age-related hearing impairment is a consequence of voltage loss, not hair cell loss
Robert Frisina	Neural processing of complex sounds
Eric Young	Information about speech segments in the normal and impaired auditory nerve

POSTER SESSION A 9:45AM – 11:00AM

THURSDAY, AUGUST 26

SESSION TWO

11:10AM – 12:20PM

Michael Heinz	Auditory-nerve rate responses are inconsistent with common hypotheses for the neural correlates of loudness recruitment
Jeff Bondy	Auditory nerve clues for hearing aid algorithms

SESSION THREE

5:15PM - 7:00PM

AUDITORY PERCEPTION IN HEARING-IMPAIRED LISTENERS

Moderator: Michael Stone

Marina Rose	The relationship between stream segregation and frequency discrimination in normally hearing and hearing-impaired subjects
Jessica Rossi-Katz	Contribution of frequency region on the identification of competing vowels in listeners with cochlear hearing loss
Benjamin Hornsby	Effects of informational masking on hearing aid benefit

FRIDAY, AUGUST 27

SESSION FOUR

8:00AM – 9:45AM

HEARING AID OUTCOME MEASURES

Moderator: Stuart Gatehouse

Anna Nabelek	Relationship between acceptance of background noise and hearing aid use
Stuart Gatehouse	Hearing disability in the population rehabilitation: Targets for speech hearing, spatial hearing and qualities of hearing
Susan Scollie	Factors of hearing aid outcome in children

POSTER SESSION B 9:45AM – 11:00AM

SESSION FIVE

11:10AM – 12:20PM

Wouter Dreschler	Reliability of client-based adjustments of amplification
Harvey Dillon	Speech intelligibility in dead regions, healthy regions, and slightly sick regions

SESSION SIX

5:15PM – 7:00PM

A SESSION ON COMPRESSION

Moderator: Jim Kates

Michael Stone	Some mechanisms by which fast-acting dynamic range compression degrade intelligibility in a competing speech task
Guido Smoorenburg	The effect of amplitude compression, time constants, and number of channels on speech intelligibility in noise by the moderately hearing-impaired
Lorienne Jenstad	Quantifying the effect of wide dynamic range compression on the temporal envelope of speech in noise

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SATURDAY, AUGUST 28

SESSION SEVEN

8:00AM – 9:45AM

SOUND LOCALIZATION AND UNIQUE APPROACHES TO HIGH FREQUENCY HEARING LOSS

Moderator: Judy Dubno

Gabrielle Saunders	Impact of hearing loss, age, and signal frequency upon sound localization ability
Gitte Keidser	The effect of signal processing features in hearing aids on horizontal localization accuracy

William Yund	Changing speech perception and localization in new hearing aid users with wide dynamic range multichannel compression and linear fittings

POSTER SESSION C 9:45AM – 11:00AM

SESSION EIGHT

11:10AM – 12:20PM

Hugh McDermott	Frequency compression can enhance speech perception for hearing aid users
Martin Vestergaard	Benefit from amplification of high frequencies: Can auditory acclimatization explain missing benefit in patients with ski-slope hearing loss?

SESSION NINE

5:15PM – 7:00PM

HIGH FREQUENCY HEARING LOSS AND NOISE REDUCTION ALGORITHMS

Moderator: Sigfrid Soli

Thomas Behrens	Relations between cognitive function, compression time constants, and speech perception in different background noise types
Jan Wouters	Noise reduction approaches for improved speech perception
Laurel Christensen	The impact of noise reduction algorithms on speech intelligibility

SUNDAY, AUGUST 29

SESSION TEN

8:00AM – 9:10AM

FROM THE EARMOLD TO THE BRAIN

Moderator: Larry Humes

Ewen MacDonald	Smearing and jittering: A comparison of two techniques to simulate hearing loss
Chris Schmitz	Enhanced BTE directivity using a directional microphone array

BREAK 9:10 AM – 9:30 AM

SESSION ELEVEN

9:30 AM – 10:40 AM

Daniel Freed	Comparative performance of adaptive anti-feedback algorithms in commercial hearing aid and integrated circuits
David Fabry	Digital inside and out: Results from several ongoing experiments

ADJOURNMENT

Oral Program

Wednesday, August 25

KEYNOTE ADDRESS

7:45 PM **SPEECH RECEPTION IN NOISE BY HEARING IMPAIRED LISTENERS**

Tammo Houtgast, VU University, Amsterdam

Sensorineural hearing impaired listeners often suffer from a reduced speech reception in noise. Detailed studies show that this group cannot be considered as homogeneous. For some, speech reception is essentially normal when taking their raised hearing threshold into account. For others, speech reception is worse than expected on the bases of their raised hearing threshold alone, suggesting additional effects of 'supra-threshold deficits'.

Thursday, August 26

SESSION ONE

AUDITORY PHYSIOLOGY, AGING AND HEARING IMPAIRMENT

Moderator: Eric Young

8:00 AM **AGE-RELATED HEARING IMPAIRMENT IS A CONSEQUENCE OF VOLTAGE LOSS, NOT HAIR CELL LOSS**

Richard Schmiedt, Medical University of South Carolina

Aged animals reared in quiet show an audiometric configuration similar to that associated with elderly humans: a mild, flat loss at low frequencies coupled with a sloping loss at high frequencies. Physiological measures, including auditory-nerve recordings and distortion product otoacoustic emissions, indicate that a lowered endocochlear potential (EP) is the cause of this type of audiometric configuration, not hair cell loss. Indeed, artificially lowering the EP with chemical agents in young animals with normal complements of hair cells and nerve fibers yields a hearing loss similar to that of aged animals. Unlike mammalian hair cells, the lateral wall cells generating the EP can replicate, making them strong candidates for replacement therapies using stem cells. [Work supported by NIA.]

8:35 AM NEURAL PROCESSING OF COMPLEX SOUNDS

Robert Frisina, University of Rochester School of Medicine & Dentistry

Hearing loss results from problems or damage to the ear (peripheral auditory system) or the brain (central auditory system). Here, the basic structure and function of the central auditory nervous system will be highlighted as relevant to effects of permanent hearing loss on complex sound processing, where assistive devices (hearing aids) are called for. The parts of the brain used for hearing are altered in two basic ways in cases of hearing loss: 1) Damage to the inner ear can reduce the number and nature of input channels that the brainstem receives from the ear, causing plastic changes in the central auditory system. This plasticity may partially compensate for the peripheral loss, or produce new abnormalities such as distorted speech processing or tinnitus. 2) In some scenarios, damage to the brain can occur independent of the ear, as may occur in cases of tumors, metabolic disease or aging. Implications of deficits to the central auditory system for speech perception in noise and hearing aid use will be provided to set the stage for subsequent presentations on hearing aid specifications and usage.

9:10 AM INFORMATION ABOUT SPEECH SEGMENTS IN THE NORMAL AND IMPAIRED AUDITORY NERVE

Eric Young, Johns Hopkins University, Teng Ji, Rush Medical College, Ian Bruce, McMaster University, Michael Heinz, Johns Hopkins University

Studies of neural encoding of speech have usually focused on representation of formants. However, for many consonants the formant transitions are only a part of the signal and other sounds, associated with aspiration or frication, are also important. Also, real speech consists of a rapid sequence of sounds with different spectro-temporal characteristics. Here, responses of auditory nerve fibers, in both normal and acoustically traumatized ears, were studied using the stimulus "Five women played basketball," a natural utterance. For analysis, the stimulus was segmented into sections corresponding to phonemes or pairs of phonemes, with approximately constant spectral shape (50-70 ms in duration). The population response (rate versus BF) was computed for each segment, in BF bins representing equal steps along the basilar membrane. Similar phonemes cluster together in this analysis. Confusion matrices were constructed using Poisson simulations of the rate responses. The confusion matrices show about 2.8 bits of information about segment identity in the rate responses. In impaired ears with a moderate (40-60 dB) high frequency loss, the information drops to about 1.8 bits. CEFS processing, using the algorithm suggested by Miller et al. (JASA, 106:2693, 1999) increases the information to about 2.3 bits. The difference is primarily due to loss of representation of information about F2 in the impaired ears and restoration by the signal processing, mainly due to high-pass filtering. This way of quantifying performance degradation provides a direct way of evaluating hearing-aid algorithms. [Supported by NIH grant DC00109.]

Thursday, August 26

SESSION TWO

AUDITORY PHYSIOLOGY, AGING, AND HEARING IMPAIRMENT

11:10 AM AUDITORY-NERVE RATE RESPONSES ARE INCONSISTENT WITH COMMON HYPOTHESES FOR THE NEURAL CORRELATES OF LOUDNESS RECRUITMENT

Michael Heinz and Eric Young, Johns Hopkins University

A number of perceptual phenomena related to normal and impaired level coding can be accounted for by the degree of basilar-membrane (BM) compression. However, auditory-nerve (AN) fibers do not provide a simple representation of the BM input-output function. An improved understanding of the neural coding of level may provide insight into the difficulty current hearing aids have in restoring normal speech perception while overcoming loudness recruitment. The present results suggest that loudness recruitment cannot be accounted for based on summed AN rate responses due to the significant effect that inner hair cell damage can have on AN activity growth.

11:45 AM AUDITORY NERVE CLUES FOR HEARING AID ALGORITHMS

Jeff Bondy, Ian Bruce, Sue Becker and Simon Haykin, McMaster University

There have been many attempts to use quantitative models to derive hearing-aid algorithms or use for offline evaluation. Most have had limited success, often not being able to deal with the variance among hearing impairments, and not being able to frame effective strategies to deal with competing speech. Our suggestion is by studying the auditory nerve response one has the best chance of understanding why the same audiograms produce different intelligibility and why hearing impaired people get very little advantage from the temporal dips of competing speech. Our first stop is to detail the dynamic, nonlinear processes lost with hearing impairment and their effects on the auditory nerve response. We have built a predictive intelligibility measure similar to the Articulation Index based on the nerve response, called the Neural Articulation Index. By using this measure we can predict the parameters of current linear fitting strategies, notably NAL-RP. We show that NAL-RP can be described as a processing strategy that returns the median auditory nerve representation. In a sense hearing aid processing may be deemed successful by how well it re-establishes the auditory representation after the cochlea. This leads to some understanding of how to tweak processing for people with the same audiogram, but different levels of hair cell damage. From here, we try to quantify what is happening, or not happening, in the sensorineural impaired cochlea in relation to confusion between phones in different noise regimes. We specifically look at the competing speech regime because of the massive intelligibility deficit that sensorineural impaired people have versus normal hearing people. By looking at the auditory response to complex stimuli, we highlight losses in the nonlinear dynamic response are not reintroduced with modern linear or nonlinear hearing-aids. The adaptation/suppression characteristics of the impaired auditory system remains dramatically impaired with these algorithms. We give a theory about the loss of auditory segregation being the main culprit in the sensorineural impaired person's difficulty to unmask competing speech, and discuss an algorithm that attempts to reestablish the time-adaptive processing lost in the damaged cochlea.

Thursday, August 26

SESSION THREE

AUDITORY PERCEPTION IN HEARING-IMPAIRED LISTENERS

Moderator: Michael Stone

5:15 PM **THE RELATIONSHIP BETWEEN STREAM SEGREGATION AND FREQUENCY DISCRIMINATION IN NORMALLY HEARING AND HEARING-IMPAIRED SUBJECTS**

Marina Rose, Aston University, UK, and Brian Moore, University of Cambridge, UK

Hearing-impaired people often complain of difficulty in understanding speech in the presence of background sounds. Mackersie et al. (2001) showed that the ability of hearing-impaired subjects to identify a target sentence in the presence of a competing sentence is correlated with the fission boundary (FB) at which a sequence of pure tones alternating between two frequencies cannot be heard as two separate streams (Rose and Moore, 1997). It is important therefore, to examine the factors that might lead to increased FBs in hearing-impaired subjects. Here, we examined the relationship between stream segregation and frequency discrimination in normally hearing and subjects with cochlear hearing loss. It is possible that the FB is determined by the discriminability of successive sounds (Moore and Gockel, 2002), in which case increases in frequency discrimination threshold caused by cochlear hearing loss should be associated with increases in the fission boundary. The frequency range examined was 250 to 8000 Hz for the normally hearing subjects and 250 to 2000 Hz for the hearing-impaired subjects.

5:50 PM **CONTRIBUTION OF FREQUENCY REGION ON THE IDENTIFICATION OF COMPETING VOWELS IN LISTENERS WITH COCHLEAR HEARING LOSS**

Jessica Rossi-Katz and Kathryn Arehart, University of Colorado at Boulder

In the identification of competing vowels, listeners with normal hearing (NH) benefit from differences in fundamental frequency (F0) up to approximately 2 semitones. The benefit from F0 differences below 2 semitones is primarily mediated by within-formant grouping of the low frequency (first formant region) portion of the vowels and to a lesser extent by across-formant grouping of the high frequency (second and higher formants) portion of the vowels (e.g. Culling and Darwin, 1993). Previous work in our laboratory on the perception of harmonic complexes and double vowels suggests that the cues that facilitate perceptual separation of double vowels in NH listeners may be less salient to listeners with cochlear hearing loss (HI). The current study focuses on the role of frequency region on listeners' ability to use F0 differences in the segregation and identification of competing vowels. Following a procedure similar to Culling and Darwin (1993), NH and HI listeners identified standard double vowels and double vowels where the F0 of the first formant differed from that of its higher formants (F0-inconsistent). The difference in F0 between simultaneous vowels was restricted to either the first-formant region of the vowel (F2-F5 SAME); the second and higher-formant regions of the vowel (F1 SAME); or both regions of the vowel (SWAPPED). Differences in F0 ranged from 0 to 8 semitones. To assure audibility of all formants, the vowels were individually amplified via a prescriptive formula for the HI listeners. Performance was measured in terms of the percentage both vowels were correctly identified in both the standard and F0-inconsistent conditions. Results show that some HI listeners differentially weight low and high frequencies. This differential use of frequency region in the perceptual separation of competing speech will be considered in light of the spectro-temporal processing deficits that accompany cochlear hearing loss. The potential influence of these deficits on the mechanisms underlying perceptual separation of competing sounds will also be addressed.

6:25 PM EFFECTS OF INFORMATIONAL MASKING ON HEARING AID BENEFIT

Benjamin Hornsby and Todd Ricketts, Vanderbilt University

The negative impact of background noise on persons with hearing loss (HI) and the limited benefits of hearing aids in noise are well documented. These limitations are due, in part, to reduced audibility resulting from elevated thresholds and "energetic masking" (EM) effects of the background noise, which combine so that portions of the speech are inaudible. Factors other than audibility, however, also impact the speech understanding of elderly hearing aid wearers (i.e., Humes, 2002). This study investigates the role that one such additional factor, "informational masking" (IM), may play in limiting aided speech understanding in the HI.

Friday, August 27

SESSION FOUR

HEARING AID OUTCOME MEASURES

Moderator: Stuart Gatehouse

8:00 AM **RELATIONSHIP BETWEEN ACCEPTANCE OF BACKGROUND NOISE AND HEARING AID USE**

Anna Nabelek, Samuel B. Burchfield, Joanna W. Tampas, and
Melinda C. Freyaldenhoven, University of Tennessee

A novel measure of listener reaction to background noise, acceptable noise level (ANL), is being investigated as a predictor of hearing aid use. ANL is determined by listeners selecting their most comfortable listening level (MCL) for a story recorded by a male voice. Next, speech babble is added and the listeners select the maximum background noise level (BNL) that is acceptable while listening to and following the story. The difference between the MCL and BNL is defined as the ANL (ANL= MCL-BNL), expressed in dB. ANL evaluates the *willingness* to listen in noise, instead of *understanding* in the presence of noise. 191 subjects were fitted binaurally with a variety of hearing aids that were selected by clinicians independent of the study, which were selected to best suit each subject's individual needs. For 56 of the subjects, ANLs were collected over a three-month period in three test sessions and compared with speech perception in noise (SPIN) scores. Both ANLs and SPIN scores were not significantly different over the three-month period, indicating no acclimatization effect to hearing aids. An additional 135 experienced binaural hearing aid users were tested during a single test session. Overall, the ANLs were not significantly different with and without hearing aids while the SPIN scores were better with than without hearing aids, reflecting benefit of amplification. The subjects' pattern of hearing-aid use was determined by a questionnaire as follows: 1) full-time; wearing hearing aids whenever needed, 2) part-time; using hearing aids only occasionally, and 3) non-use; complete rejection of hearing aids. Mean ANLs were 7.7 and 7.4 dB for full-time users (N=69) and 13.5 and 13.0 dB for part-time users (N= 69) without and with hearing aids, respectively. For non-users mean ANLs were 14.5 (N=53) and 14.6 dB (N=26) without and with hearing aids, respectively. The group ANLs were compared with SPIN scores, abbreviated profile of hearing aid benefit (APHAB) scores and audiometric data. The mean ANL for full-time users was significantly smaller (more background noise was accepted) than for part-time and non-users. The SPIN scores were not significantly different among the three groups. The correlation analyses indicated that ANLs were not related to SPIN scores, age, and PTA. ANLs were weakly correlated with ease of communication (EC), reverberation (RV), and background noise (BN) APHAB subscales. On the basis of ANL, full-time uses can be predicted with 81% confidence either with or without hearing aids. Part-time and non-use can be predicted with 82% confidence, although means of separating these groups are unclear. SPIN and APHAB scores did not predict use of hearing aids. Further investigation will involve separation of part-time and non-users, possible means of lowering ANLs to generate more full-time users, and determination of physiological responses related to large inter-subject variability of ANLs in listeners with normal and impaired hearing. [Work supported by NIH/NIDCD.]

8:35 AM HEARING DISABILITY IN THE POPULATION REHABILITATION: TARGETS FOR SPEECH HEARING, SPATIAL HEARING AND QUALITIES OF HEARING

Stuart Gatehouse, MRC Institute of Hearing Research, UK

We have recently reported a new inventory, the Speech, Spatial and Qualities of Hearing Scale (SSQ), for the assessment of auditory disability (Gatehouse, S & Noble, W. 2004. The Speech, Spatial and Qualities of Hearing Scale (SSQ), International Journal of Audiology, 43:85-99; Noble, W & Gatehouse, S. 2004. Interaural asymmetry of hearing loss, Speech, Spatial and Qualities of Hearing Scale (SSQ) disabilities, and handicap, International Journal of Audiology, 43:100-114.). In addition to traditional contexts of speech-hearing, the SSQ includes monitoring multiple speech streams (divided attention), suppressing unwanted speech streams (informational masking), switching attention between speech streams, appreciation of distance and movement as well as static location, sound source segregation, sound identification, listening effort and concentration, and appreciation of prosody. We found that these non-traditional elements form important drivers of the experience of auditory handicap over and above traditional speech contexts. Modern signal processing in hearing aids maybe hypothesised to degrade the cues upon which functioning in these perceptually more complex environments depends. More complex and demanding environments and tasks are also likely to be influenced by non-auditory attentional and cognitive capacities. It is important to be able to dissociate direct auditory consequences from those, which will inevitably accompany the deficits as people age.

9:10 AM FACTORS OF HEARING AID OUTCOME IN CHILDREN

Susan Scollie and Richard Seewald, University of Western Ontario, Canada, Harvey Dillon and Teresa Ching, National Acoustic Laboratories, Australia

Recent work has described a multifactorial structure, in which large sets of outcome measures from adult hearing aid users have been described using a small number of factors (e.g., Humes, 1999; 2003). The primary analysis in this work has been Principal Components Analysis (PCA). Extracted factors have been interpreted as speech recognition performance, hearing aid use, and subjective benefit/satisfaction, as one example (Humes, 2003). This work is important because it defines a multidimensional construct which may be used to simply yet comprehensively describe and understand hearing aid outcome. This has the potential to inform the design of future studies in hearing aid outcome. Unfortunately, there has been no report in the literature that describes whether these specific factors would generalize to children who use hearing aids. We have applied PCA to outcome measures obtained from a pool of children enrolled in an 8-month hearing aid trial. The outcome measures collected include speech recognition in quiet and noise, loudness perception, and various questionnaires. Preliminary results replicate, to a large extent, the adult dimensions that have been reported in the literature. Interestingly, some systematic differences also exist, which may be interpretable based on child versus adult lifestyles (e.g., listening in school).

Friday, August 27

SESSION FIVE

HEARING AID OUTCOME MEASURES

11:10 AM **RELIABILITY OF CLIENT-BASED ADJUSTMENTS OF AMPLIFICATION**

Wouter A. Dreschler, Academic Medical Center, The Netherlands
Gitte Keidser, Elizabeth Convery and Harvey Dillon, National Acoustics Laboratories, Australia

The fitting of modern hearing aids is becoming increasingly complex, because the hearing aid is expected to adapt to different acoustical environments, either under user control or automatically. This requires active feedback of the hearing aid user about his preferences for different acoustical environments obtained for instance, by paired comparisons using CD's with samples of real-life sounds. This approach is time consuming and relatively indirect, because the real user environments can only be simulated to a certain degree in the clinic or in the laboratory. Future developments may facilitate a more direct approach if the subject will be allowed to control the most important fitting parameters in his/her own acoustical environment. The concept of a trainable hearing aid is promising in this respect.

In this study the listener is given access to a set of controls and while listening to selective sound samples that represent real-life listening situations the client is able to adjust the amplification characteristics to the preferred settings. 24 subjects with mild to moderate-severe symmetrical sensorineural hearing loss with flat and sloping audiograms participated in a series of laboratory tests using videos to simulate six different acoustical environments. The study investigates the effectiveness of four different ways to provide the user with optimal control of the amplification curve based on the resulting values of test-retest reliability and based on subjective judgements from a questionnaire. Also, we applied different (roved) starting points to investigate the convergence of the procedure.

The analysis of the results showed significant differences between the four controllers with respect to the test-retest reliability, but not with respect to the finally preferred amplification characteristics. The subjective judgements were more favourable for the controllers that yielded the better test-retest reliability. However, the finally preferred amplification characteristics proved to be strongly determined by the starting point. The clinical implication of these findings will be discussed.

11:45 AM SPEECH INTELLIGIBILITY IN DEAD REGIONS, HEALTHY REGIONS, AND SLIGHTLY SICK REGIONS

Harvey Dillon, Teresa Ching, Frances Lockhart, Emma VanWanrooy, Lydia Lai, National Acoustic Laboratories, Australia

When hearing loss becomes sufficiently severe, hearing-impaired people have reduced ability to understand speech, even when it is totally audible. This reduced ability has been referred to as hearing loss desensitization. The occurrence of desensitization seems certain, but its extent, and its relationship to frequency selectivity and the presence of dead regions, is uncertain. Also uncertain is the impact of background noise and type of speech material on apparent desensitization.

Friday, August 27

SESSION SIX

A SESSION ON COMPRESSION

Moderator: Jim Kates

5:15 PM SOME MECHANISMS BY WHICH FAST-ACTING DYNAMIC RANGE COMPRESSION DEGRADE INTELLIGIBILITY IN A COMPETING SPEECH TASK

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

Using a cochlear implant simulator (Shannon *et al.* 1995), we (Stone & Moore, 2003) have reported that wideband fast-acting compression led to poorer intelligibility than slow-acting compression in a competing speech task. This may be partly due to reduced modulation depth and reduction in temporal contrast. Compression speed was varied by using different pairs of attack and release times. In a subsequent experiment (Stone & Moore, 2004), we have shown that attack times less than about 2 ms in a wideband compressor are deleterious to intelligibility, but our range of attack times did not show that we had improved intelligibility to its maximum possible. This result prompts the question whether it is not only the spectral artifacts produced by fast gain changes but also the distortion of temporal envelope shape by the compressor that degrades intelligibility. In experiment 1 reported here, we extend the range of attack times tested from 2.5 ms up to several hundred ms, while keeping other parameters of the compression system near-constant.

Stone & Moore (2004) also reported an experiment where fast wideband compression was applied to the target and background either before or after mixing. The former reduced the modulation depth of each signal but maintained the independence between the two signals, while the latter introduced "co-modulation" between the two signals. Using simulations with 6 and 11 channels, intelligibility was consistently higher when compression was applied before mixing. A potential counter-argument to this result is that wideband compression, in the condition of compression applied before mixing, could have enhanced the temporal coherence because the whole signal, from a single source, is modulated by the gain signal, which itself was entirely derived from that single source. Grouping processes in the auditory system are known to use the across-frequency common fate of the temporal envelope as a cue. Wideband compression in this condition could have enhanced the temporal coherence of the individual speech signals, and thereby improved performance. A possible counter-argument is that the across-frequency coherence of speech from a single talker is already high, so little extra could be added by wideband compression. We therefore repeated the experiment applying compression before or after mixing of the target and background, but replaced the wideband compression with independent multi-channel compression. Since the compressors were independent, any additional across-frequency temporal coherence in the condition of compression before mixing was confined to each channel. For the 6-channel simulation, intelligibility was higher for compression applied before mixing. For the 11-channel simulation, there was little difference between intelligibility in the two conditions. Comodulation between target and background produced by the compression rather than reduced temporal contrast appears to be the major cause of reduced intelligibility.

5:50 PM THE EFFECT OF AMPLITUDE COMPRESSION, TIME CONSTANTS, AND NUMBER OF CHANNELS ON SPEECH INTELLIGIBILITY IN NOISE BY THE MODERATELY HEARING IMPAIRED

Guido F. Smoorenburg and Rolph Houben, University Medical Center Utrecht, The Netherlands

We investigated systematically the influence of several compression parameters on speech intelligibility in speech-shaped noise by the moderately hearing impaired. Experimental conditions included all combinations of compression ratio (CR = 1/2, 2/2, 2/3, 3/3 for low/high frequencies), attack and release times (Ta/Tr = 4/4, 4/40, 40/40, 4/400, and 40/400 ms), and number of channels (1, 2, and 6). Twenty subjects with moderate sensorineural hearing loss took part in the experiment. With two-channel compression the best average speech reception threshold (SRT) was found at a compression ratio of 2/3 and Ta/Tr = 40/40 ms. It was 0.7 dB better than linear amplification. With six-channel compression the best results (SRT 0.4 dB better than linear amplification) were obtained at larger time constants (Ta/Tr = 40/400 ms) and CR=2/2 and 2/3. With single-channel compression the best SRT was equal to linear amplification. It was found for the longest time constant (Ta/Tr = 40/400 ms) and CR = 2/2. In fluctuating noise the best speech reception threshold was found with single-channel compression at CR = 2/2 and Ta/Tr = 4/400 ms. The SRT was 0.9 dB better than in linear amplification. The results suggest that it is almost impossible for a hearing aid dispenser to optimise the parameter settings of a modern hearing aid.

6:25 PM **QUANTIFYING THE EFFECT OF WIDE DYNAMIC RANGE COMPRESSION ON THE TEMPORAL ENVELOPE OF SPEECH IN NOISE**

Lorienne Jenstad and Pamela Souza, University of Washington

The temporal envelope of speech carries important cues for speech identification (Shannon, et al. Science, 1995), such as voicing and manner of phonemes (Rosen, PhilTran, 1992), prosodic information (e.g., Flynn, et al. JSLHR, 1998), and sound source segregation in noise (Crouzet & Ainsworth, Eurospeech, 2001). Wide-dynamic-range compression (WDRC) processing in hearing aids is known to affect the temporal envelope of speech (e.g., Jenstad & Souza, submitted; Souza & Kitch, EarHear, 2001; Van Tasell & Trine, JSHR, 1996) by reducing the modulation depth of the speech signal. Some populations of listeners may be more susceptible to these changes, such as older listeners who require greater modulation depth than younger listeners (e.g., Fitzgibbons & Gordon-Salant, JAAA, 1996) and listeners with severe to profound hearing loss who place greater reliance on temporal cues (e.g., Rosen, et al. Acta Oto Suppl, 1990; Souza, et al. submitted).

Saturday, August 28

SESSION SEVEN

SOUND LOCALIZATION AND UNIQUE APPROACHES TO HIGH-FREQUENCY HEARING LOSS

Moderator: Judy Dubno

8:00 AM **IMPACT OF HEARING LOSS, AGE AND SIGNAL FREQUENCY UPON SOUND LOCALIZATION ABILITY**

Gabrielle Saunders, M. Samantha Lewis, Betsy Rheinsburg, Akiko Kusumoto
National Center for Rehabilitative Auditory Research

Auditory localization is a critical component of daily function, both for reliable identification of the direction of alerting signals, and during social communication such as for locating a talker in a crowd. The auditory system uses different mechanisms for localizing high and low frequency signals. In this study we evaluate the impact of hearing loss and age upon localization of low-, mid- and high-frequency narrowband noises and speech-shaped noise.

8:35 AM THE EFFECT OF SIGNAL PROCESSING FEATURES IN HEARING AIDS ON HORIZONTAL LOCALIZATION ACCURACY

Gitte Keidser, Harvey Dillon, Liz Convery and Lyndal Carter, National Acoustic Laboratories, Australia, Kristin Rohrseitz and Volkmar Hamacher, Siemens Hearing Instruments, Germany

Accurate localization in the horizontal plane depends on the presence of appropriate interaural time differences or interaural level differences or both. Advanced hearing aids contain signal processing features that affect the level and timing of signals at their outputs in a manner dependent on the input signal. Because hearing aids on the left and right side of the head receive different input signals, even identically adjusted hearing aids can alter the signal level timing by different amounts at any instant. Consequently, the more adaptive the hearing aid characteristics, the more the potential for the advanced processing to adversely affect localization accuracy.

An experiment was performed to investigate the separate effects on localization of multichannel compression, multichannel adaptive noise suppression, and adaptive directional microphones. The first two of these features should affect interaural level differences, and the third should affect interaural level and time differences. Measurements were performed with 20 loudspeakers in an anechoic chamber, using pulsed broadband noise signals. The presentation will show the effects of each processing scheme, measured on 12 hearing impaired subjects. These results will be related to the physical differences measured at the input and output of the hearing aids on each side of the head.

9:10 AM CHANGING SPEECH PERCEPTION AND LOCALIZATION IN NEW HEARING AID USERS WITH WIDE DYNAMIC RANGE MULTICHANNEL COMPRESSION AND LINEAR FITTINGS

E. William Yund and Christina M. Roup, VANCHCS, Helen J. Simon and Al Lotze, Smith-Kettlewell Eye Research Institute

Some auditory cues used in speech perception and sound localization become progressively less available as hearing impairment develops. As a result of this loss of auditory information, speech perception and localization adapt to optimize performance with the auditory cues that remain. This suggests that the auditory information reintroduced by the new hearing aid may not be of immediate value without an additional period of adaptation or re-learning. It has been difficult, however, to demonstrate that a significant period of acclimatization is required for the listener to obtain maximum benefit from new hearing aids. It also has been suggested that there may be differences in the time course of acclimatization for different types of hearing aids, but again there is little empirical evidence for or against this possibility. The present study, carried out over the last three years, was designed to clarify some aspects of these issues.

SESSION EIGHT

SOUND LOCALIZATION AND UNIQUE APPROACHES TO HIGH-FREQUENCY HEARING LOSS

11:10 AM FREQUENCY COMPRESSION CAN ENHANCE SPEECH PERCEPTION FOR HEARING-AID USERS

Hugh McDermott, The University of Melbourne, Andrea Simpson and Adam Hersbach, CRC for Cochlear Implant and Hearing Aid Innovation

Many people with sensorineural hearing impairment have poorer sound sensitivity at high frequencies than at lower frequencies. Amplification benefits most such people, provided that their impairment is not too severe. However, attempting to improve high-frequency audibility with amplification alone often fails to increase speech recognition for aid-users with severe high-frequency loss. Strategies developed to address this problem include techniques that shift high-frequency components of sound down to lower frequencies. Although several variations of frequency lowering have been reported, the common aim is to improve audibility by taking advantage of regions with better hearing thresholds.

Recently, a sound-processing scheme has been developed that applies frequency lowering only to signal components having relatively high frequencies. Below a predetermined cut-off frequency, sounds are amplified with appropriate frequency shaping and compression. Above the cut-off frequency, signals are compressed in frequency in addition to undergoing amplification. The frequency compression applies progressively larger shifts to components having increasingly higher frequencies. Consequently, the signal bandwidth at the output of the hearing-aid is narrower than at the input.

Experimental findings demonstrate that this scheme improves speech intelligibility for some aid-users having severe high-frequency hearing loss. In one experiment, 17 experienced aid-users wore the frequency-compression device in place of their conventional instruments for approximately 5 weeks. Their recognition of monosyllabic words was evaluated with each aid. Eight of the subjects obtained a significant score improvement ($p < 0.05$) with frequency compression; only one subject's score decreased significantly. The remaining subjects had similar scores with each device.

Because some of the observed improvements could have resulted from the better audibility provided by the frequency-compression device rather than from the frequency lowering *per se*, a second experiment was conducted. In that study, high-frequency gain was increased with no frequency shifting. It was found that the higher gains generally failed to provide benefits equivalent to those obtained with frequency compression, in some cases because of the onset of feedback oscillations.

The experimental frequency-lowering scheme has now been transferred to a digital, fully programmable, behind-the-ear instrument. Preliminary results of ongoing evaluations of the new device suggest that it is also capable of providing aid-users with significant perceptual benefits, as well as subjectively acceptable sound quality.

[Partly supported by the Garnett Passe and Rodney Williams Memorial Foundation.]

11:45 AM **BENEFIT FROM AMPLIFICATION OF HIGH FREQUENCIES: CAN AUDITORY ACCLIMATIZATION EXPLAIN MISSING BENEFIT IN PATIENTS WITH SKI-SLOPE HEARING LOSS?**

Martin D. Vestergaard, Research Centre Eriksholm, Denmark

Sounds brought back to audibility by hearing aids are expected to provide usefulness for the wearer in terms of better auditory function. While the rationales underlying provision of amplification to hearing-impaired patients are well known, it sometimes happens that audibility does not provide the expected benefit. The auditory acclimatization effect has been suggested for explaining why audibility fails to provide functional benefit to some users of hearing aids; however, the notion has not been consistently confirmed.

Saturday, August 28

SESSION NINE

HIGH FREQUENCY HEARING LOSS AND NOISE REDUCTION ALGORITHMS

Moderator: Sigfrid Soli

5:15 PM **RELATIONS BETWEEN COGNITIVE FUNCTION, COMPRESSION TIME-CONSTANTS, AND SPEECH PERCEPTION IN DIFFERENT BACKGROUND NOISE TYPES**

Thomas Behrens, Elisabet Sundewall, and Graham Naylor, Oticon Research Centre, Eriksholm, Denmark

No consensus exists regarding optimal time constants for compression amplification. We present data here from two studies indicating that 1) Optimal time constants may be different for different listeners, 2) Changing time constants makes a big difference or very little, depending on the type of background noise, 3) Aspects of cognitive function are correlated with optimal time-constants for the individual listener when aided, 4) Aspects of cognitive function are increasingly implicated in speech-in-noise performance as temporal variations in the noise or amplification increase.

Common to both studies are: blinded crossover design, 2-channel digital HA platform, Fast (40 ms) versus Slow (640 ms) release time in both channels as the sole contrast, 9-10 weeks acclimatization period with each setting.

The first dataset has 50 subjects, and is a subset of the dataset generated in the Gatehouse et. al. study, in which the FAAF test (target word in carrier sentence) was used. The second study with 23 subjects used a different speech testing paradigm (Hagerman sentences), allowing estimation of the SRT and slope of the psychometric function.

Subjective outcome data from the field test periods also lend support to the hypothesis that aspects of cognitive function have a material influence on optimal release time constants for the individual hearing aid user in everyday use.

5:50 PM NOISE REDUCTION APPROACHES FOR IMPROVED SPEECH PERCEPTION

Jan Wouters, Simon Doclo, Thomas Klasen, Jean-Baptiste Maj, Marc Moonen, Lies Royackers, Ann Spriet, Tim Van den Bogaert, Katholieke Universiteit Leuven, Belgium

Speech understanding in noisy listening conditions is a big problem for hearing aid users. Several noise reduction signal processing schemes have been investigated for application in hearing aids, but few implemented in commercial devices. Some single microphone strategies have been introduced yielding limited benefit. Configurations of two microphones are commonly implemented and demonstrate benefit in some sound scenes. Multi-microphone systems can take advantage of the spatial information of the desired and the jammer sound sources in addition to spectro-temporal information. At present, adaptive directional microphone systems are available in commercial hearing aids, allowing to adapt to changing jammer directions and to track moving noise sources. The small sizes of these arrays in hearing aids as well as specific signal processing aspects, however, introduce an increased sensitivity to errors in the assumed signal model (microphone mismatch, speech distortion, voice activity detection, room acoustics, …).

In our research we have been focussing on two and three microphone configurations for behind-the-ear (BTE) devices in combination with robust algorithms economic enough to be implemented in signal processors of hearing aids and cochlear implants. The degree and the robustness of the benefit in realistic, difficult listening environments is of major importance. The performance measures used are speech intelligibility weighted speech-to-noise ratios and distortion on the physical side, and speech reception thresholds on the perceptual side.

In general, the multi-microphone noise reduction approaches studied consist of a fixed spatial pre-processor that transforms the microphone signals to speech and noise reference signals, followed by an adaptive processing stage. In the two-stage adaptive beamformer or generalized sidelobe canceller (GSC) the second stage is an unconstrained adaptive noise canceller that uses the output of the first stage, and is only allowed to adapt during non-speech periods. The performance of this adaptive noise reduction strategy has been compared with an adaptive directional microphone, state-of-the-art in modern commercial digital hearing aids. Improvements in speech reception threshold between 7 and 12 dB have been obtained for the GSC in different realistic multi-source scenes.

These developments have been extensively evaluated with normal hearing, hearing impaired and cochlear implant subjects. Results will be given.
Since the fixed and the adaptive stage of the GSC-technique or two-stage adaptive beamformer rely on a-priori assumptions, this may give rise to undesired speech distortion and a reduced noise cancellation. Recently, we developed optimal filtering techniques which are more robust against signal model errors. A first technique is based on singular value decomposition (SVD). Although this approach has a high computational complexity, we were able to show performances better than GSC in difficult listening conditions (speech coming from some non-zero angle in a background of 3 different noise sources in a reverberant room). Secondly, a novel multi-microphone optimal filtering noise reduction scheme (spatially pre-processed speech distortion weighted multi-channel wiener filter) has been developed. The latter technique has an improved robustness of the fixed stage by incorporating statistical knowledge about the microphone characteristics in the design of the adaptive filter, and is computationally less expensive. Experimental results show increased benefit.

The former monaural adaptive noise reduction techniques have been extended to bilateral systems. First results will be discussed.

6:25 PM THE IMPACT OF NOISE REDUCTION ALGORITHMS ON SPEECH INTELLIGIBILITY

Laurel Christensen and Andrew Dittberner, GN ReSound Group, Yingyong Qi, Qualcomm

Noise reduction algorithms (directionality and spectral noise reduction) have been implemented in digital hearing aids to provide listening comfort and enhance acoustic conditions leading to improved speech intelligibility in noise by the end user. Research in this area has focused on whether these enhanced acoustic conditions lead to improved or degraded speech intelligibility in noise. Standard electroacoustical measurements, real ear measurements and human-listener-based judgments are utilized to evaluate the efficacy of different noise reduction strategies. Though each of these methods has their place in traditional models of hearing aid fitting and evaluation, an alternative, computer-based method is proposed to effectively predict whether a noise reduction strategy will provide potential improved or degraded speech intelligibility in noise for the end user. The purpose of this study was to look at noise reduction algorithms using a new type of analysis method that allows quantification of changes in the spectral content of an amplified signal in the presence of background noise (SNR). Results from this methodology were clinically validated to see if the predicted values were perceptually relevant.

Sunday, August 29

SESSION TEN

FROM THE EARMOLD TO THE BRAIN

Moderator: Larry Humes

8:00 AM SMEARING AND JITTERING: A COMPARISON OF TWO TECHNIQUES TO SIMULATE HEARING LOSS

Ewen MacDonald, Kathy Pichora-Fuller, Bruce Schneider, Hans Kunov, and Willy Wong, University of Toronto, Canada

Older listeners have hearing problems related to spectral and to temporal processing declines. Listeners with cochlear hearing loss exhibit broadening of auditory filters associated with loss of outer hair cells. Older adults, even those with audiometric thresholds within normal clinical limits in the speech range, exhibit declines consistent with loss of temporal synchrony coding. We examined two different simulations of auditory aging, ‘smearing’ to simulate declines in spectral processing, and ‘jittering’ to simulate declines in temporal processing. Smearing is a technique based on physiological models of broadened auditory filters in cochlear hearing loss (Baer & Moore, 1993). Jittering simulates the effects of reduced neural synchrony or periodicity coding presumed to be the basis for age-related differences in measures such as masking-level differences and frequency discrimination (Schneider & Pichora-Fuller, 2001). Since it is impossible to modify the frequency structure of a signal without also modifying the time domain structure (and vice versa) each method will introduce an unintended distortion. The physical and perceptual consequences of this unintended distortion were explored.

8:35 AM ENHANCED BTE DIRECTIVITY USING A DIRECTIONAL MICROPHONE ARRAY

Chris Schmitz, Nandini Iyer, Michael Lockwood, Charissa Lansing, Douglas Jones, University of Illinois at Urbana-Champaign

Understanding speech in noise is perhaps the greatest challenge for hearing-impaired listeners. Directional microphone technologies have proven to be the most effective means of addressing this problem. A directional microphone or a directional response formed from a linear array of two or more omni microphones must always form an axially symmetric free-field response. The head-related transfer function (HRTF) introduces an asymmetry, however, which cannot be fully exploited by such an array. A new three-element BTE array (o-g-o array) consisting of a gradient directional microphone and two omni-directional microphones exploits the asymmetry of the HRTF to obtain better directivity on the head.

Sunday, August 29

SESSION ELEVEN

FROM THE EARMOLD TO THE BRAIN

9:30 AM COMPARATIVE PERFORMANCE OF ADAPTIVE ANTI-FEEDBACK ALGORITHMS IN COMMERCIAL HEARING AIDS AND INTEGRATED CIRCUITS

Daniel Freed and Sigfrid D. Soli, House Ear Institute

Adaptive control of feedback is an increasingly common feature in hearing aids, but there are few reports comparing the effectiveness of different implementations. We compared the performance of eight commercially available adaptive anti-feedback algorithms; six were implemented in complete hearing aids, and two were implemented in integrated circuits intended for use in hearing aids. We also included our own experimental algorithm, implemented in a laboratory system. All evaluations were performed on KEMAR using BTE devices and a skeleton earmold with a fully open Select-A-Vent.

We performed three experiments to address the following questions for each algorithm: (1) How effectively does the algorithm prevent oscillation and reduce suboscillatory peaks in the frequency response under static acoustic conditions? (2) How quickly and effectively does the algorithm adapt to changing acoustic conditions? (3) How robust is the algorithm when presented with tonal input signals?

In Experiment 1, we measured added stable gain and real-ear aided gain for a 50-dB(A) white noise input, both with and without a telephone handset held against KEMAR's ear. In Experiment 2, we measured the time required to suppress oscillation after a hat was suddenly placed on KEMAR's head. In Experiment 3, we presented 50-dB(A) pure-tone inputs at 500, 1000, 2000, and 4000 Hz and measured the level of extraneous frequencies in the output.

We found dramatic differences in performance among the algorithms. In the static acoustic conditions, the amount of added stable gain achieved by the algorithms ranged from 0 dB (no benefit) to 18 dB, and the real-ear aided gains showed wide variation in the ability to reduce suboscillatory peaks. The response to changing acoustic conditions ranged from virtually instantaneous adaptation to a failure to adapt at all. Pure-tone input signals had no deleterious effect on some algorithms, but produced instability or excessive extraneous frequencies in others. Algorithms that performed well in one measurement condition often performed poorly in another, such that no single algorithm was clearly superior overall.

These results show a need for continuing research on anti-feedback algorithms. Clinicians and hearing aid users should temper their expectations of anti-feedback features in current commercial products.

10:05 AM DIGITAL INSIDE AND OUT: RESULTS FROM SEVERAL ONGOING EXPERIMENTS

David Fabry, Phonak Hearing Systems, Thomas Powers, Siemens Hearing Instruments, Robert Sweetow and Jennifer Henderson-Sabes, University of California, San Francisco, Robert Brey and Richard Robb, Mayo Clinic

Digital manufacturing of hearing instrument shells comprises three main stages: 1) scanning of the ear impression of external auditory canal, 2) software modeling of shell surface, vent pathways, and canal length, and 3) three-dimensional printing of the finished shell. This presentation will focus on the results of several ongoing experiments focused on technical improvements with the digital shell process.

The results of several experiments designed to better characterize the external auditory canal will be discussed. The first experiment used 330 earmold impressions taken from hearing aid patients with their jaw positioned in open, neutral, and closed position, to determine the location and extent of movement in the cartilaginous portion of the external auditory meatus. The results suggest that considerable movement exists in the lateral dimension, which suggests that a critical component of any future efforts should be to characterize the elasticity and bony structure underlying the cartilaginous tissue.

To that end, a second experiment was conducted to evaluate the canal dimension using computed tomography (CT) and magnetic resonance imaging (MRI). The results from a single subject determined that ear scans could be developed from the CT and MR images, but that both suffer from shortcomings. Although these strategies may have application for pediatric hearing aid fitting, future efforts will need to be directed towards more clinically useful approaches (e.g. optical scanning techniques) for scanning ear canal dimensions.

The final issue related to scanning of the ear impression relates to an intermediate step between the current process and the elimination of the ear impression altogether. A desktop scanner was placed in a remote facility and ear impressions were scanned by four audiologists and compared with those conducted by an experienced technician. The average difference between these scanned impressions across individuals averaged less than 0.4 mm.

An experiment underway at the present time is evaluating the use of scanned impressions, using different jaw positions, to evaluate whether the “align and compare” technique may be used to compensate for feedback and/or occlusion, by “oversizing” or “undersizing” the shell on the basis of dynamic ear canal movements. Data collection is in progress, and results will report whether the use of digital scans, in combination with shells altered by the shape change with jaw movement will provide improved comfort, fit, feedback, and/or occlusion.

Poster Program

Posters for Session A should be put up by 8 A.M. Thursday, August 26, and taken down after 10 P.M. Thursday, August 26, or before 7 A.M. Friday, August 27. 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

Prevalence and pattern of hearing aid use in cochlear implant recipients

Robert Cowan, CRC for Cochlear Implant and Hearing Aid Innovation, Australia, Juliet Chin-Lenn and Mansze Mok, University of Melbourne, Australia

Seventy-one adult cochlear implant users, all of whom had thresholds of less than 90dBHL at 500Hz in the non-implanted ear, were surveyed in regards to their pre-operative and post-operative use of hearing aid(s). Results showed a significant percentage of subjects continued to use a BTE in conjunction with their cochlear implant. Studies showed advantages in listening to speech in noise, in localisation, and in quality of sound. Interestingly, the bimodal ratings given were higher for those subjects with a shorter period of non-use of the hearing aid after receiving an implant. However, results showed that only a small percentage of these subjects had their hearing aid specifically adjusted to facilitate bimodal listening with a cochlear implant and hearing aid together.

A2

An integrated approach to algorithm design for enhancement of audibility, intelligibility and comfort

Bert de Vries, GN ReSound, The Netherlands

Over the past decade, signal processing for industrial hearing aids has evolved from analog electronics for amplitude compression to sophisticated digital systems that integrate multiple adaptive algorithms for beamforming, feedback cancellation, amplitude compression, noise reduction and signal quality enhancement. Commonly, these modules are designed independently and consequently the opportunities for undesired interactive effects abound. In this contribution we present an approach to integrate adaptive modules for amplification, amplitude compression, noise reduction and signal quality (distortion) control.

A3

Evaluating the performance benefit of adaptive, directional, interference-canceling systems in everyday listening environments

Joseph Desloge, Martin Zimmer, Patrick Zurek, Sensimetrics Corporation

Many advanced digital signal processor (DSP) powered hearing aids now include directionally adaptive interference cancellation among the tools used to improve target-signal intelligibility in the presence of noise. Specifically, they operate by adaptively attenuating interference sources (jammers) that are spatially separated from the desired target signal location. Performance of these systems depends upon several factors, including the number of microphones in the adaptive system, the number of independent jammer sources in the environment and the level of environmental reverberation. The purpose of the current study was to investigate the performance of a two-microphone adaptive noise-canceling system within everyday listening environments in order to explore how much 'adaptive benefit' (relative to simple, non-adaptive microphones such as cardioid or hyper-cardioid) could typically be obtained with realistic jammer and reverberation configurations. Four classes of environment were considered: outdoors, household, parking garage, and public establishment (e.g., restaurants, cafes.). Sources were either environmental noises (e.g., household appliances, restaurant noise) or a controlled white noise source placed at known distances from the microphones. In all situations, no signal was present at the target location (i.e., all signals were jammers) in order to allow for maximal jammer cancellation and prediction of maximal adaptive benefit. Two microphone recordings were made in the various environments, and the signals were processed offline. Adaptive processing systems included systems based upon both the Griffiths-Jim Generalized Sidelobe Canceller and the more general Linearly-Constrained, Minimum-Variance architecture – all implemented using several adaptive filter lengths. Results indicate average cardioid-relative, intelligibility-weighted, adaptive benefit levels on the order of 1-4 dB for household rooms, 1-7 dB for outdoors, 1-5 dB for the parking garage and 0.5-2.0 dB for public establishments. [Work supported by NIOSH].

A4

A three dimensional instrument-based approach to estimating the directivity index and predicting the directional benefit of directional microphone systems in hearing aids

Andrew Dittberner, GN Resound, Ruth Bentler, University of Iowa

Directional microphones in hearing aids (DMHAs) differ from each other in terms of their spatial attenuation characteristics. To quantify these differences, spatial directivity patterns and directivity indices are derived. Directivity Index (DI) computations on DMHAs have been presented in the literature in three different ways: (1) mathematical-derived DI, (2) empirically-derived free field DI, and (3) empirically-derived KEMAR DI. The conventional methodology used to compute empirically derived DIs assumes that spatial data gathered in the horizontal plane of the device under test are equivalent to spatial data in other planes of reference. However, one cannot assume such symmetry when computing a KEMAR DI for a DMHA.

A5

Modulation-based filtering in digital hearing aids

Wouter A. Dreschler, László Körössy; Alex E. Hoetink, Academic Medical Center, The Netherlands

In many digital hearing aids modulation-based filtering is applied as a method of noise reduction. Although there are many different methods of implementing this type of filtering, currently there are no standards for their technical evaluation. It has been shown that ICRA noises are able to do so and knowledge about the differences between aids may be of use in the interpretation of the subjective results. Therefore, we documented the technical functionality of modulation-based filtering in a variety of 12 hearing aids from different brands, using ICRA noises.

A6

The performance-perceptual test as an indicator of hearing aid satisfaction

Anna Forsline and Gabrielle Saunders, National Center for Rehabilitative Auditory Research

The Performance-Perceptual Test (PPT) enables a direct comparison of measured and perceived ability to understand speech in noise, using the same test materials, adaptive test procedures and scoring methods for each. More specifically, the Hearing In Noise Test is run in two conditions: (a) using the recommended test procedure and (b) allowing the subjects to judge whether they think they can just understand everything that is said and altering the S/N accordingly. This yields two speech reception thresholds in noise (SRTNs), known as the Performance SRTN and the Perceptual SRTN respectively. A third value can be computed: the difference between the actual and perceived SRTNs. This is the Performance-Perceptual Discrepancy (PPDIS). It is a measure of the accuracy with which individuals estimate their hearing ability. The test is further described in Saunders et al (2004).

A7

The effect of digital noise reduction time constants on speech recognition in noise

Jason Galster, and Todd Ricketts, Vanderbilt University

It has been established that many Digital Noise Reduction (DNR) algorithms available in commercially hearing aids offer improvements for listening comfort in noise. However, these systems have repeatedly not shown significant benefit for speech recognition ability in noise. Isolated studies have shown that one DNR system, which uses extremely fast attack time constants, may offer improvements for speech recognition in noise. The possible reasons for this improvement, however, remain unclear.

The purpose of this study was to compare speech recognition performance across DNR algorithms in two commercially available hearing aids, with contrasting DNR time constants. Sixteen subjects participated in this study. Speech recognition testing was completed in an anechoic environment using HINT sentences in a background of steady state noise. Test conditions included programs for each hearing aid with DNR active and inactive. The two hearing aid models were matched for audibility, compression threshold, compression ratio, compression time constants and absence of active expansion processing. The hearing aids were contrasted by the time constants of the active DNR algorithms. One device used a slow attack time constant (>100ms) while the other used a fast attack time constant (<50 ms). Speech recognition scores revealed a significant improvement for the fast acting DNR processing, but not the slow acting. Electroacoustic analysis revealed that in a background of steady state noise, gain for sinusoidally amplitude modulated pure tones and HINT speech stimuli were not significantly different when the fast acting DNR system was both active and inactive, suggesting similar audibility for speech in noise across the two settings. This study was not designed to directly address the parameters responsible for the source of benefit from this fast acting DNR system; however, the primary variable between test devices was the active DNR time constants. This relationship leads to several hypotheses regarding the source of speech recognition benefit. Future studies will revolve around adjustment of time constants and investigation of cross channel masking and non-simultaneous masking effects within these active DNR algorithms.

A8

Investigations into the locus of nonlinear gain control in the auditory system

Michael Gordon and Bruce Schneider, University of Toronto at Mississauga, Canada

A number of studies have demonstrated that the auditory system employs nonlinear gain control to modulate its response to a broad range of stimulus intensities (e.g., Parker, et al., 2002), but the precise location and nature of this mechanism remains uncertain. To address these issues, we investigated the extent to which gain control in one ear was influenced by auditory events in the other ear. Listeners were asked to make absolute judgments of intensity for a set of four acoustic stimuli: 25, 30, 35 & 40 dB SPL presented to the left ear. Subsequent test sets included an 80 dB SPL stimulus presented either to the left (ipsalateral) or the right (contralateral) ear. Ipsalateral presentations of the 80 dB SPL stimulus produced a greater reduction in discriminability between the original four tones than did contralateral presentations. This result suggests that the gain control mechanism may act peripherally, modulating sensitivity separately within each ear. Additional testing, using the precedence effect to vary the illusory ear of presentation (while actually presenting the 80 dB SPL stimulus to both ears), resulted in equivalent ipsilateral and contralateral losses in sensitivity, further suggesting that the gain control mechanism may be located in the auditory periphery.

A9

Compression-dependent differences in hearing aid gain and predicted speech audibility between speech and non-speech input signals

Rebecca Warner Henning, University of Wisconsin, and Ruth Bentler, University of Iowa

A primary purpose of fitting hearing aids is to improve the audibility of speech; however, hearing aid gain is typically measured using standardized non-speech signals, e.g., swept pure tones, speech-weighted broadband noise, or modulated noise. When compression hearing aids are tested with these non-speech input signals, the measured gain can be quite different than if a real speech input signal, matched in overall rms level to the non-speech signal, were used (Scollie et al., 2002; Stelmachowicz et al., 1996). This study systematically evaluated the effects of release time, compression ratio, and number of compression channels, as well as interactions of these parameters, on the gain difference between speech and several common non-speech hearing aid test signals. The magnitude of error when using gain measured with a non-speech signal to estimate speech audibility, using the Speech Intelligibility Index (SII), was also evaluated.

\A10

Segregating speech from noise: The role of low-frequency audibility

Su-Hyun Jin, University of Wyoming, Peggy B. Nelson and Melanie J. Gregan, University of Minnesota

In a recently completed study of hearing-impaired (HI) listeners' speech perception in noise (Jin and Nelson, 2004), two factors emerged as highly related to amplified sentence recognition in the presence of modulated noise: low frequency audibility and auditory filter bandwidths. Nine young adult listeners with sensorineural hearing loss participated in the series of experiments, which included the following measurements: recognition of consonant-vowel (CV) and sentence stimuli in quiet and in steady and modulated noise, forward masked thresholds, auditory filter bandwidths, and auditory fusion. HI listeners were matched for their hearing thresholds in the 2k–4k Hz region. Eight young adults with normal hearing (NH) sensitivity served as controls. Amplified speech recognition performance of the HI listeners was equal to that of the NH listeners in quiet and in steady noise, but was significantly poorer in modulated noise. Thus, even when amplification was adequate for full understanding of speech in quiet and in steady noise, HI listeners experienced significantly less masking release from the modulated maskers. Our findings indicated that those listeners with greatest hearing losses in the low frequencies were poorest at understanding amplified sentences in modulated noise. In addition, those HI listeners with wider auditory filters (in the 2k–4k Hz region) were poorer than HI listeners with near-normal auditory filter bandwidths. These two findings are consistent with the hypothesis that strong spectral representation of voice pitch is necessary for auditory segregation of speech from noise (e.g., Qin and Oxenham, 2003). This finding seems somewhat at odds with traditional noise-reduction signal processing algorithms in digital hearing aids, which often reduce low-frequency amplification in the presence of significant background noise. Additional results from HI and NH listeners will be presented, in which we systematically vary the audibility of low-frequency regions of speech while measuring masking release. Implications for amplification strategies will be discussed. (This work was supported by the NIDCD, the University of Minnesota, and the University of Wyoming.)

A11

Viscoelastic foam for perfect fit of dynamic ear canals

Vasant Kolpe, Robert Oliveira, William Parrish, Martin Babcock, and Michael Venem, Hearing Components

The superior acoustic attenuation property of viscoelastic polyurethane foam over conventional hard acrylate and soft silicone rubber was established and presented at the last IHCON 2002. The experiments were carried out in an Acoustic Test Fixture (ATF) at body temperature on several precise thicknesses of earmold materials. In axial compression, smooth hard acrylate is 10 -12 dB inferior and silicone 5-7dB inferior to viscoelastic foam.

The need for compliant foam and its rheological ability to keep pace with deformation is more appreciated today with the current ear canal volume data, the jaw held in open and closed position. The study documents variation in ear canal volumes, right and left ear, of the same patient.

We have tested several viscoelastic foams with the ATF above at body temperature and found that they provide a gradation in "noise" attenuation increasing from 10 dB to 33 dB. The "noise" signal is as available on the Fonix® 6500-CX hearing aid tester. It appears that the highly elastic foam at low frequency has lower attenuation. The slow recovery foams provide higher attenuation.

Earlier studies on dynamic properties of compliant viscoelastic foam reveal that higher values of "noise" attenuation, approximately 30 dB, result from energy dissipation capability at higher frequencies. Further low and high frequency dynamic modulus measurements of graded attenuation foams are in progress. These acoustic and rheological measurements will provide further understanding of compliant viscoelastic foam as an earmold.

A12

Improving speech intelligibility in babble for patients with multiple sclerosis

M. Samantha Lewis, Michele Hutter, David Lilly, Stephen Fausti, Portland VA Medical Center, Dennis.Bourdette, Oregon Health and Science University

Multiple sclerosis (MS) is a disease of the central nervous system that affects over 400,000 Americans. It should be noted that only 2-6% of the population with MS has peripheral hearing impairment. Despite this statistic, almost half of the population with MS complains of difficulty hearing, especially in the presence of background noise. With these thoughts in mind, the purpose of the present investigation was twofold: (1) to examine the speech intelligibility in noise of adults with MS and of those without MS and (2) to evaluate the effects of a frequency modulation (FM) system on speech intelligibility in noise for adults with MS.

A13

A modified audibility noise variance of the speech recognition sensitivity (SRS) model

Hannes Müsch, Sound ID, and Søren Buus, Northeastern University

We report further development on the speech recognition sensitivity (SRS) model [H. Müsch and S. Buus, J. Acoust. Soc. Am. 109, 2896 – 2909 (2001)], which is a model for predicting speech intelligibility. The aim of the SRS model is similar to that of the Articulation Index (AI) and the Speech Intelligibility Index (SII). By fitting the model to consonant recognition performance of normal-hearing listeners as reported in nine independent studies, we explored the optimal relation between the speech-excitation to noise-excitation ratio in the auditory periphery and a related SRS model parameter, the audibility noise variance. The best predictions were obtained when the audibility noise variance was directly proportional to the noise-excitation power. We propose that this modified relation replace the relation suggested in the original SRS formulation.

A14

Innovations in hearing aid systems: Why do we search for panaceas?

Naylor Graham, Oticon Research Centre, Denmark

This presentation addresses the question of how to provide hearing aid (HA) clients with an optimal fitting, as opposed to an acceptable one.

It will be argued that much of the research carried out to develop improved HA systems has been based on the assumption that what is good, is good for everybody. Likewise the popular Prescription Rules implicitly support the idea that the optimum audiological goal of a HA is the same for all clients. Thus research becomes a search for a 'Panacea'. This argument will be supported by evidence from the research literature concerning innovations in HA systems – 1) Historically, smaller improvements with each new generation of HA systems 2) Typically very small N in research study designs 3) Focus on group mean effects 4) Rare acknowledgement that individual differences might be a real result.

It will be suggested that a more appropriate way of viewing amplification schemes is to categorise them as 'General Purpose' or 'Targeted' schemes. General-Purpose schemes address classes of needs that are shared by almost all clients (e.g. re-establishment of audibility). Targeted schemes address classes of needs or abilities specific to some sub-group of clients. A Targeted scheme has the possibility to achieve significantly greater benefits for its targeted clients than a General Purpose scheme can for those same clients. Likewise, for the wrong group of clients, a Targeted scheme may provide markedly less benefit than a General Purpose scheme. These principles will be illustrated with data from the recent study by Gatehouse et.al. (IHCON 2000).

Well-designed General Purpose schemes achieve good results for a large proportion of clients with little clinical effort. Hence the success of the modern non-linear Prescription Rules such as DSL[i/o], IHAFF, NAL-NL1. Targeted schemes promise greater benefits but at the price of extra effort spent on client classification and the risk of poor outcomes when mis-targeted.

In some clinical contexts, application of a General Purpose scheme is the most prudent approach, and in other contexts application of Targeted fitting is appropriate.

New features are constantly being added to hearing aids. None of them is likely to be good for everyone, all the time. Modern advanced devices thus provide a host of possibilities for Targeted fitting. The secret of success with each new advance lies in identifying candidate clients and providing tools for the targeting process, a task for researchers and manufacturers that is at least as great as the technical development of features.

A15

Modeling individual ear canal geometries and the effect of ventilation in hearing aids

Morten Nordahn, Widex A/S, Denmark

The geometry of individual ear canals interacts with the dimensions of a ventilation canal in determining the acoustic properties and hence the actual gain of the hearing aid. Previous modeling studies of the ventilation effect have not included this interaction. In this study, a comprehensive model of the entire acoustic circuit of the hearing aid, ear and plug has been applied to study the acoustic changes that the drilling of a vent entail in the performance of a hearing aid in the individual ear. The model is designed as a two-port circuit with a high degree of flexibility. Every part of the physical circuit, including receiver, tubing, ear canal, ear drum, venting, and leakage has a corresponding block in the model, which can be altered according to the individual situation that is being modeled.

A16

It's the ear, not the impression

Bob Oliveira, Bob, Greg Hoeker, Martin Babcock, Vasant Kolpe and Bill Parish, Hearing Components

Ten years of our R&D sponsored by NIDCD has resulted in a better understanding of ear canal dynamics with jaw motion. Additionally, this sponsorship has supported the development of practical solutions to improve the physical fit of hearing aids.

Laser scanning of ear impressions and EarQuant software has documented that substantial size and shape changes occur in the soft part of the canal as a result of changes in jaw position. Within the same individual remarkable asymmetry can occur between ear canals caused by jaw position. 50.7% of the 67 subjects in this study had a 10%, or more, positive or negative, volume change in the cartilaginous region of at least one ear with jaw opening! MR images offer insight into a biomechanical mechanism of action. A simple difference in the relative location of the TMJ in relationship to the ear canal can account for a positive or negative volume change from a closed to open jaw position. The implications are that it is the ear canal geometry change with jaw position that is a primary cause for poor physical fits of hearing aids. Blaming the impression is often the wrong focus.

To solely rely on tight (the usual technique is to capture open jaw impressions) earmolds and shells to overcome these changes in the ear canal would result in about 1 in 8 patients likely having poor fitting resulting in feedback, since ~12% of subjects in this study had more than a 10% negative volume change with open jaw impressions. Traditional materials, e.g., acrylates and silicone rubbers, for earmolds and shells are at most flexible. A more reasonable solution in some situations would be to use compliant materials that can expand and contract in situ to maintain comfortable effective sealing for earmolds and shells.

A17

Speech coding for wireless communication in the hearing aid environment

Harald Pobloth and Bastiaan Kleijn, Royal Institute of Technology, Sweden, Bert de Vries, GN ReSound, The Netherlands

Potential attractive applications of wireless communication of speech signals in the hearing aid environment include binaural and remote processing. Today, these applications have not been realized due to the power demands of wireless communication. An important way to reduce power consumption is to minimize the number of bits that need to be communicated, while retaining satisfactory speech signal quality. Standard speech coding algorithms are not appropriate since the hearing aid environment imposes a unique set of (latency and computational) constraints that are much more severe than, for instance, for a mobile phone application.

A18

AI diagnosis

Christine Rankovic, Articulation Inc.

The articulation index (AI) is a metric that ranks the speech sound transmission capacity of a communications path and predicts speech recognition test performance. The AI is calculated from an acoustical or electroacoustical description of the path that must include an audiogram when considering a listener with hearing impairment. The Fletcher AI calculation [H. Fletcher and R.H. Galt, J. Acoust. Soc. Am. 22, 89-151 (1950)] is uniquely suited for customizing hearing aids and evaluating speech processing algorithms because it produces four metrics (in addition to the AI) that can be used to diagnose the path. These AI component metrics, V, E, F, and H, characterize the contribution to the AI of: (1) speech audibility; (2) loudness discomfort; (3) frequency-response balance; and (4) ear-generated distortions, respectively. The definition and significance of the AI component metrics will be illustrated using data drawn from the published literature.

A19

The relationship between pre-use hearing aid expectations and generic vs. hearing aid-specific measures of outcome

Gabrielle Saunders, National Center for Rehabilitative Auditory Research, and Jeffrey Jutai, University of Western Ontario

As the need to demonstrate efficacy of treatment has become more important, measurement of hearing aid outcomes has become a major issue for audiologists. Among hearing professionals, outcome measurement tends to focus on communication-specific improvement following intervention. Tools for doing so are either hearing-related questionnaires or tests of speech recognition scores. Generic measures of outcome are rarely used because most studies show them to be insensitive to pre-and post-hearing aid use. However, there would be advantages to having a sensitive generic outcome measure. These include the fact that a better understanding of the generic issues underlying auditory rehabilitation could be achieved and that direct comparisons could be made between the impact hearing aids and other assistive devices have upon quality of life.

A20

A psycho-acoustic model of the pleasantness for hearing impaired persons (MCHI)

Georg Schmalfuss and Joerg Haubold, ciAD, Germany

An individually successful hearing aid fitting guarantees a pleasant sound of the hearing instrument beside a good speech intelligibility in everyday life situations. The prediction of the sensed pleasantness by the hearing impaired person with and without an hearing instrument is important for the description of the hearing aid fitting success and also for the development of modern hearing aid algorithms. This estimation can be implemented into a psycho-acoustic model. For normal hearing people, already such models exist. Hearing-impaired persons with hearing instruments are rating the pleasantness of natural sounds however qualitatively differently, than normal hearing people. This is proved by a data collection of over 1,000 hearing impaired subjects. It is inadequately to adapt the models of the pleasantness for normal hearing people to hearing impaired persons under consideration of the affected loudness perception.

A21

Can speech intelligibility in noise explain "unexpected" loudness preferences?

Karolina Smeds and Nils Smeds, Karolinska Institute, KTH, Sweden

Most of the prescriptive methods for WDRC hearing aids are based on the assumption that a hearing-impaired listener should perceive amplified sounds at approximately the same overall loudness as would a normal-hearing listener without amplification. However, people with mild to moderate cochlear hearing loss seem to prefer less than normal overall loudness when loudness is calculated according to a model by Moore and Glasberg (1997). This has been shown in recent research using nonlinear amplification and the research has confirmed some results previously obtained with linear amplification. The results are puzzling, and in the current study two possible explanations are explored.

A22

A real-time normal- and impaired-hearing loundness estimator and data logger for hearing research

Justin Zakis and Hugh McDermott, The University of Melbourne, Australia, Harvey Dillon, National Acoustics Laboratories, Australia, Karolina Smeds, KTH Royal Institute of Technology, Sweden

The restoration of normal loudness sensations is an aim of many hearing-aid amplification strategies. In order to evaluate the merit of this aim, a wearable research hearing aid (known as SHARP) was programmed to estimate and log the sound pressure levels (SPLs) and loudness estimates of real-life sounds, both before and after amplification, as well as the aid user's preferred volume adjustment relative to the prescribed gain. The loudness estimates for the unamplified sounds were computed for normal hearing, while the loudness estimates for the amplified sounds were computed for the audiogram of the aid user. All loudness estimates were computed by a real-time implementation of a loudness model for cochlear hearing loss proposed by Moore and Glasberg. Real-time computation required some approximations, which will be discussed, that introduced some error to the loudness estimates. The error was measured for single-tone, multiple-tone, and broadband noise stimuli at different SPLs. The error was generally between 0 and 3 phon, except at lower SPLs and low stimulus frequencies. The error is sufficiently low for investigations into the loudness that is preferred by normal- and impaired-hearing users of the SHARP in real-life environments, relative to the unaided normal-hearing loudness. Other possible applications of the real-time loudness model include the development of a processing scheme that aims to adaptively amplify sounds to a fixed or varying proportion of normal-hearing loudness. [This work was supported by The Co-Operative Research Centre for Cochlear Implant and Hearing Aid Innovation, Australia].

A23

Design of a vibration membrane of differential electromagnetic transducer of middle ear implant hearing aid for control of vibration characteristics

Byung-Seop Song, Min-Kyu Kim, Young-Ho Yoon, Sang-Heun Lee, and Jin-Ho Cho, Kyungpook National University, Korea

In order to use in the middle ear implant hearing aid for mild to severe hearing loss patients, a differential electromagnetic transducer (DET) that has the vibration frequency response close to that of the normal human middle ear bone was developed.

A24

Open ear fitting using a transcutaneous air conduction hearing aid system (TACHAS)

Mark Winter and Thomas Lenarz, Medical University of Hanover, Germany, Theo Wesendahl, ENT Clinic and Hearing Center Rheine, Germany

For patients suffering from mild to moderate high-frequency sensorineural hearing loss a new category of hearing devices, specified by the FDA, has become available. The transcutaneous air conduction hearing aid system (TACHAS) provides a comfortable alternative to conventional hearing aids. It consists of a titanium tube system, which is implanted into the soft tissue of the outer ear, and a retroauricularly inserted digital hearing module. The open arrangement of this new sound induction procedure solves the key problems (the occlusion effect and the attenuation of lower frequencies) and preserves all outer ear characteristics, especially the ear canal resonance effect.

A25

Predicting speech intelligibility for rapid gain variation systems

William S. Woods and Timothy D. Trine, Starkey Laboratories, Inc.

Given the use of fast time constants in compression and single-channel noise reduction systems, it would be helpful to have an accurate predictor of intelligibility for such systems. With the relatively rapid gain changes that can occur in these systems, however, intelligibility predictions cannot be performed with the basic articulation index (AI) or speech intelligibility index (SII), and the speech transmission index (STI) may also not be accurate (Drullman, 1995; Hohmann and Kollmeier, 1995). Also, the metrics used in many reports wherein a SNR is estimated by comparing the processed noisy signal to the clean speech signal (e.g., Li et al., 2001; Hasan et al., 2004; Hagerman, IHCON ’02), typically yield SNR increases that are not seen to produce concomitant intelligibility increases. Other, auditory-model based methods for such predictions (cf. Holube and Kollmeier, 1996) employ extensive computational algorithms that may not be necessary for accurate predictions. Thus, in this work we report on a computationally simple processing method that uses the underlying ideas of the AI but allows them to be applied in contexts that violate the assumptions of the standardized AI computational method. The method is an extension of one given in Stelmachowicz et al. (1994), applied in, for example, Souza and Turner (1999), in which changes in the speech dynamic range due to processing are incorporated directly into an index computation. The method as applied here essentially compares speech and noise spectrograms with the same duration. From the spectrograms it determines the proportion of the important speech dynamic range that is above the noise level (or absolute threshold, whichever is higher), and converts the proportion to an index value using a frequency-weighting function. The method also allows for the inclusion of spread of masking in frequency and time. When the index is computed this way, reducing noise levels when there is no speech (such as in speech pauses), or changing the speech dynamic range in noise via rapid gain changes (as in fast compression) when the noise controls audibility, will not contribute to a change in the index. This is consistent with the lack of change in speech intelligibility seen for normal listeners with most versions of such processing. Also, any time-invariant, linear processing generates the same index as would be found with a standard AI computation. We will describe the method in detail, and compare predictions of the method to measurements of speech intelligibility for subjects listening in modulated noises that test the novel features of the method.

A26

Microphone port misalignment and directivity of custom directional hearing aids

Tao Zhang, Timothy D Trine, Laurel J Olson, Weili Lin, and Shilpi Banerjee, Starkey Laboratories, Inc.

Directional hearing aids are commonly designed with the assumption that the microphone ports are correctly aligned with the specified direction (i.e. look direction). However, this assumption is often violated in practice. Previous research reported that such misalignment degrades directivity index significantly when the misalignment reaches about 20 degrees based on 2-dimensional polar patterns (Mueller and Wesselkamp, 1999; Ricketts, 2000).

A27

Immersive simulation of hearing loss and auditory prostheses

Patrick Zurek and Joseph Desloge, Sensimetrics Corporation

Simulation of hearing loss is useful for demonstrating the communication challenges facing hearing-impaired people. However, current 'simulations,' most of which are only recordings, do not actually elevate thresholds; i.e., they do not simulate hearing loss, per se. The hearing loss simulator described in this talk is immersive – the user's detection thresholds for ambient sounds are shifted by a prescribed degree. This threshold shift is achieved through a combination of passive attenuation (from muff-type hearing protectors) and additive masking noise (introduced by within-muff earphones). Acoustic signals picked up by microphones near each ear are processed through bandpass AGC channels and delivered via the earphones to complete the simulation of frequency-dependent hearing loss and loudness recruitment. Preliminary results validating the accuracy of specified threshold shift will be presented, along with speech-reception data comparing simulated with actual hearing losses. Subjective reactions of users engaged in one-on-one conversation suggest that strong feelings of communication disability are engendered by even moderate degrees of simulated hearing loss. The system, which is capable of simulating any degree of recruiting hearing loss along with hearing aids or cochlear implants, can provide effective interactive demonstrations of both auditory communication handicap and rehabilitation options. [Work supported by NIDCD].

Posters for Session B should be put up by 8 A.M. Friday, August 27, and taken down after 10 P.M. Friday, August 27 or before 7 A.M. Saturday, August 28. 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

Functioning, disability, and quality of life in the adult hearing impaired

Harvey Abrams and Rachel McArdle, Bay Pines VA Medical Center, Theresa H Chisolm, University of South Florida, Patrick .Doyle, VA Pittsburgh Healthcare System

Despite the well-documented social and economic costs of hearing loss and the predicted increases in its prevalence, the relative burden of the disorder and the relative cost-effectiveness of audiological intervention (i.e., hearing aid use) as compared to those of other chronic health conditions is currently unknown. In order to conduct such comparisons across chronic health conditions the effects of hearing aid intervention need to be measured utilizing a generic test instrument. The objective of this study was to determine the effects of binaural amplification on self-reported functioning, disability, and well being among individuals with hearing impairment as measured by selected individual domain scores and the summary scores of the World Health Organization – Disability Assessment Scale (WHO-DAS II) and of the Medical Outcomes Study –Short Form 36 Veteran's version (MOS-SF36V). Three hundred and eighty-four participants were recruited for this randomized, controlled, pretest/post-test experimental study. All participants received routine comprehensive audiologic evaluations. Participants were randomly assigned to an immediate treatment group or a delayed treatment group. The study treatment consisted of bilateral hearing instruments that were selected and fit according to currently accepted clinical practice. Each participant's self-reported functioning, disability, and well-being was measured by the MOS-SF36V and WHO-DAS II. These instruments were completed by the participants prior to treatment, 8-weeks post treatment, 16-weeks post treatment, and 1-year post treatment. Results will be reported on 367 participants who have completed the 16-week post treatment visit to date. Preliminary results suggest that intervention with hearing aids produces a significant improvement in quality of life as measured by the mental component scale of the MOS-SF36V ($F(1,364) = 7.8$, $MSE = 121121.9$, $p < .05$) and the total score of the WHO-DAS II ($F(1,362) = 31.9$, $MSE = 1452.4$, $p < .001$). Examination of effect sizes suggests that the WHO-DAS II (Cohen's $d = .3$) is more sensitive to changes in quality of life as a result of hearing aid intervention than the mental component subscale of the MOS-SF36V (Cohen's $d = .13$). The results of this study support the WHO-DAS II as a generic measure of the effects of audiologic intervention on self-perceived quality of life. [This work was supported by VA RR&D].

B2

Initial studies of binaural temporal acuity in elderly hearing-impaired patients

Michael Akeroyd, MRC Institute of Hearing Research, UK

Hearing-impaired patients commonly report difficulties in situations involving sounds whose spatial aspects are dynamically changing across time. These difficulties may reflect impairments in the binaural auditory system and, in particular, its ability to track spatial changes. The temporal properties of binaural processing can be characterized by its "temporal acuity" and its "temporal integration"; essentially, the former is "how fast?" (the shortest duration over which a measurement can be made or the degree to which surrounding sounds can be rejected) and the latter is "how long for?" (the longest time over which information can be included to advantage). Here, measurements of binaural temporal acuity are reported for elderly hearing-impaired patients, are related to self-report-questionnaire measurements of their spatial-hearing disabilities [the "SSQ": Gatehouse and Noble, Int. J. Audiology, vol 43, 85-99 (2004)], and also are compared to listeners with normal-hearing. The experimental design is based on that of Kollmeier and Gilkey [J. Acoust. Soc. Am., vol 87, 1709-1715 (1990)], in which the binaural detectability of a short, interaurally-inverted 500-Hz tone is measured as a function of its position in a much-longer broadband noise which has a step transition in binaural configuration at its temporal center. The pattern with which the detectability of the signal varies across the transition in the noise gives an estimate of binaural temporal acuity. Early results indicate that this pattern is slightly shallower for elderly hearing-impaired listeners than for young controls. If further measurements indicate that this result is general, then it would suggest that there is a reduction in binaural temporal acuity in patients, and which may partly underlie some deficits or difficulties in spatial hearing.

B3

Consonant recognition of sine wave modeled speech by normal-hearing and hearing-impaired individuals

Rupa Balachandran, California State University Sacramento, Arlene Neuman and Harry Levitt, City University of New York, Stuart Rosen, University College London, and Patrick Nye, Haskins Laboratories

Sine wave modeling is a parametric tool for representing the speech signal with a limited number of sine waves. It involves replacing the peaks of the speech spectrum with sine waves and discarding the rest of the lower amplitude components during synthesis. It has the potential to be used as a speech enhancement technique for hearing-impaired adults. The present study answers the basic question of how many sine waves are required for consonant recognition by normal-hearing and hearing-impaired adults. VCV syllables representing 20 consonants (/p/, /t/ /k/, /b/, /d/, /g/, /f/, /£c/, /s/, /ƒÃƒ}, /v/, /z/, /tƒÃ/, /ƒÔƒÊ/, /y/, /w/, /r/, /l/, /m/, /n/) in three vowel contexts (/a/,/i/,/u/) were processed with 4, 8, 12 and 16 sine waves. A consonant recognition task was performed in quiet and with cafeteria noise in the background (+10 dB and 0 dB signal-to-noise ratios). Twenty hearing-impaired listeners and five normal-hearing listeners were tested under headphones at their most comfortable listening level.

B4

Sensitivity to delayed auditory feedback in speakers with severe-profound hearing loss

Dragana Barac-Cikoja, Cara Johnson-Adornetto, Linda Kozma-Spytek; Natalye Faison, Gallaudet University

Sensitivity to delayed auditory feedback during speech production was assessed in five speakers with severe-profound hearing loss, using a two-interval, forced-choice (2IFC) adaptive procedure. Subjects repeated a syllable (either PA, TA, or KA) for approximately 2 secs in two successive intervals and listened via insert-phones to the speech feedback that was linearly amplified to each subject's comfortable level. Subjects were required to identify which of the two utterances was delayed relative to the production onset. The length of the delay was changed stepwise depending on the accuracy of the subject's response. The estimates of the minimal delay yielding 71% correct performance were obtained under three experimental conditions. In Condition 1, the low frequency (<500 Hz) band (LFB) was filtered out from the feedback signal, and temporal delays were applied to the spectrum above 500 Hz. In Condition 2, the LFB was separated from the rest of the spectrum by band-pass filtering. This band remained present in the feedback, but was never subjected to a delay. The spectrum above 500 Hz was subjected to delays resulting in varied degrees of spectral asynchrony within the feedback signal. In Condition 3, the LFB was delayed along with the rest of the speech spectrum. The subjects' evaluations of (a) the task difficulty, and (b) the quality of the delayed feedback under each experimental condition were also obtained. Estimates of the delay detection thresholds and their stability will be discussed with reference to the subjects' sensitivity levels for the given feedback spectrum. The relationship between the threshold estimates and the perceived disturbances due to the feedback delay will also be addressed.

B5

Compensating for the hearing aid vent: Is it trivial?

Lars Bramslow, Mie Jørgensen, Peter Lundh, and Lise Obeling, Oticon A/S, Denmark

Using large vents or even open fittings is an option for digital hearing aids with active feedback cancellation. In order to match a specific insertion gain target, the vent must be accounted for in two ways: 1) The sound entering the vent may provide enough gain, but how does it interact with the amplified sound? 2) If considerable target gain is required low-frequency energy is lost out of the vent, and the target is thus not matched. These problems seem simple from an acoustical point of view, i.e. converting the target insertion gain to coupler gain, as presented in the literature. When adjusting the hearing aid response in later fitting, attention must also be paid to these rather large vent effects.

B6

The effect of training on word recognition performance across talkers

Matthew Burk and Larry Humes, Indiana University

Hearing aid users often have difficulty understanding speech within a background noise. One aspect which is often over looked during the rehabilitative process and is independent of the hearing-aid technologies used, is training of the individual user. Of interest here, is the ability to train a hearing-aid user to improve their speech recognition performance. Familiarity with voices in intelligibility task has been shown to facilitate recognition of words. The goal of these studies is to identify whether training on an isolated word list presented by a standardized talker can generalize to everyday speech communication across novel talkers. That is, what would be the optimum words and set length for use in a clinically viable training procedure to improve a hearing-aid users real-world speech understanding.

B7

Comparison of different fitting rationales: Loudness restoration, NAL-Nl1 & DSL i/o

Josef Chalupper, Siemens Audiological Engineering

Loudness normalization which means not only to restore the overall loudness, but also the specific loudness pattern often is referred to as the "original" goal of hearing aid fitting. However, despite many years of research related to loudness normalization, "younger" fitting rationales as DSL i/o and NAL-NL1 are more widespread in clinical use. One reason for this might be that appropriate hearing aid signal processing schemes and prescriptive formulas for loudness normalization have not been available.

B8

Perception of nonlinear distortion by hearing-impaired people

Chin-Tuan Tan and Brian C.J. Moore, University of Cambridge, UK

All transducers and many signal processing systems, including hearing aids, introduce nonlinear distortion. The effects of nonlinear distortion may be described as “harshness” or “roughness” or "noisiness" or in terms of the perception of sounds that were not present in the original signal such as “crackles” or “clicks”. In a previous study (Tan *et al.*, 2003), we reported perceptual experiments in which normally hearing subjects had to rate the perceived quality of speech and music that had been subjected to various forms of nonlinear distortion, including: (1) hard and soft, symmetrical and asymmetrical clipping; (2) center clipping; (3) "full-range" distortion, produced by raising the absolute magnitude of the instantaneous amplitude of the signal to a power ($\neq 1$), while preserving the sign of the amplitude. We have been exploring a number of models that can be used to predict the subjective ratings, and our most recent model is highly accurate (Tan *et al.*, 2004). Here, we report the results of similar experiments for subjects with cochlear hearing loss. Stimuli were subjected to frequency-dependent amplification as prescribed by the "Cambridge formula" (Moore and Glasberg, 1998) before presentation via Sennheiser HD580 earphones. The pattern of the ratings was reasonably consistent across subjects; the correlation of the ratings of individual subjects with the mean ratings across subjects was always greater than 0.8. The general pattern of the results was similar to that for normally hearing subjects. Center clipping and soft clipping had only small effects on the ratings, while hard clipping and the full-range distortions had large effects. The results indicate that hearing-impaired subjects are able to make orderly and consistent ratings of degradations in sound quality introduced by nonlinear distortion. We are currently exploring how to modify our model so as to give accurate predictions of the ratings for hearing-impaired subjects.

B9

Design, implementation and evaluation of a robust multi-microphone noise reduction algorithm for hearing instruments

Simon Doclo, Ann Spriet, , Jan Wouters, and Marc Moonen, Katholieke Universiteit Leuven, Belgium

Noise reduction algorithms in hearing aids and cochlear implants are crucial for hearing impaired persons to improve speech intelligibility in background noise. Multi-microphone systems exploit spatial in addition to spectro-temporal information of the desired and the noise signals and are hence preferred to single-microphone systems. For small-sized microphone arrays such as typically encountered in hearing instruments, multi-microphone noise reduction however goes together with an increased sensitivity to errors in the assumed signal model such as microphone mismatch (gain, phase, position), reverberation, speech detection errors, etc.

In this presentation we discuss the algorithm design, the low-cost implementation and the real-time evaluation of a robust generalised multi-microphone noise reduction scheme, called the *Spatially Pre-processed Speech Distortion Weighted Multi-channel Wiener Filter (SP-SDW-MWF)*. The structure of this scheme strongly resembles the widely used Generalised Sidelobe Canceller (GSC), and consists of two parts: a robust fixed spatial pre-processor, generating speech and noise reference signals, and a robust adaptive Multi-channel Wiener Filter (MWF), reducing the residual noise in the speech reference. This generalised scheme encompasses both the GSC and the MWF as extreme cases and allows for attractive in-between solutions such as the Speech Distortion Regularised GSC (SDR-GSC).

In the standard GSC, both the fixed spatial pre-processor and the adaptive stage strongly rely on a-priori assumptions (e.g. about the microphone characteristics). When these assumptions are not satisfied, both stages give rise to undesired speech distortion and to a reduced noise reduction performance. Robust solutions have been proposed e.g. by calibrating the used microphone array and by using a Quadratic Inequality Constraint (QIC). In the SP-SDW-MWF, *robustness against signal model errors* is achieved by incorporating statistical information about the microphone characteristics (gain, phase, position) into the design procedure of the fixed spatial pre-processor and by taking speech distortion explicitly into account in the optimisation criterion of the MWF.

For the *implementation* of the adaptive MWF, we discuss an efficient stochastic gradient algorithm in the frequency-domain, whose computational complexity and memory usage is comparable to the NLMS-based Scaled Projection Algorithm for implementing the QIC-GSC.

Experiments have been performed using a 3-mic BTE hearing aid, mounted on a dummy head in an office room, where the desired speech source is positioned in front of the head and 5 babble noise sources are positioned at different angles. Simulation results show that the proposed scheme achieves a better noise reduction performance for a given maximum speech distortion level, compared to the widely studied QIC-GSC. The subjective speech enhancement and robustness performance of this generalised multi-microphone noise reduction scheme is currently being evaluated with normal hearing and hearing impaired persons.

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B10

Single-microphone speech enhancement using speech stream segregation

Rong Dong, Jeff Bondy, Ian Bruce, and Simon Haykin, McMaster University

Speech enhancement in multi-speaker babble remains an enormous challenge. While the normal functioning human auditory system is able to segregate and stream separate sounds in the cocktail-party environment, the sensorineural impaired system has a very hard time. One way of explaining human performance is to consider the auditory environment as a complex scene containing multiple objects and to hypothesize that the normal auditory system is capable of grouping these objects into separate perceptual streams based on distinctive features, while the impaired system can not. For future hearing-aid systems, we are developing signal-processing strategies that can help this missing perceptual grouping process of hearing impaired individuals.

The model we use is a three-stage, bottom-up process. First we analyze the input using a multichannel auditory model that estimates important acoustic features. Secondly we bind features across multiple channels as well as track in time. Lastly we emphasize the features to heighten their perceptual relevance. We are attempting to mimic what happens in the human auditory system as it allocates the different components of a sound to a common source based on acoustic cues such as fundamental frequency (F0) or onset time. We fuse multiple grouping cues in parallel and integrate for segregation. The end processing is a stream membership function evolving over a channel in time, that indicates whether that channel belongs to the target stream or not. A selective amplification or suppression is then applied based on this mapping. Subjective listening tests demonstrate that our model has the potential of attenuating babble noise in a substantial manner, while preserving the intelligibility of the target stream.

B11

Vowels in clear and conversational speech: Talker differences in intelligibility for elderly hearing-impaired listeners

Sarah Hargus Ferguson, University of Kansas

Several studies have shown that when a talker is instructed to speak as though talking to a hearing-impaired person, the resulting clear speech is significantly more intelligible than typical conversational speech. A recent study using a database of 41 talkers demonstrated that for normal-hearing listeners identifying vowels in noise, the amount of improvement talkers achieve by speaking clearly varies widely. Acoustic analyses of the vowels produced by 12 of these talkers suggested that the amount of clear speech vowel intelligibility benefit was related to certain clear speech acoustic strategies adopted by the talkers, specifically expansion of the vowel space. However, an earlier paper (Ferguson & Kewley-Port, 2002) suggested that while this clear speech acoustic strategy improves vowel intelligibility for normal-hearing listeners, it may not be beneficial for the group of listeners for whom clear speech is actually intended: listeners with hearing loss. The current project will assess vowel intelligibility in clear and conversational speech for older hearing-impaired adults, using materials from the 12 talkers cited above. Intelligibility data from these listeners, including clear speech effects, will be compared to intelligibility data from young normal-hearing listeners as well as the results of acoustic analysis. [Supported by the University of Kansas Center for Research New Faculty General Research Fund and by NIHDCD-02229.]

B12

Simultaneous dynamic optimization of intelligibility and loudness in hearing instruments

Matthias Fröhlich and Josef Chalupper, Siemens Audiological Engineering

Conventional nonlinear fitting formulas aim at different goals: with DSL[i/o], the range of audibility of hearing impaired subjects is to be restored, while NAL-NL1 maximizes speech intelligibility without exceeding the loudness as perceived by normal hearing subjects. For a given acoustic situation, the intelligibility depends on the frequency dependent difference between speech spectrum and background noise spectrum. NAL-NL1 was developed on the basis of speech intelligibility in quiet, so that it does not capture this dependency on the particular background noise. Furthermore, by its very design as a prescriptive formula, NAL-NL1 can only optimize speech intelligibility in the long-time average even under optimal conditions.

B13

Acclimatization - a comparison of manufacturer fitting proposals with real fittings

Heike Hauermann and Martin Kinkel,
KIND Hoergeraete, Germany

Acclimatization denotes the adaptation to a different hearing perception within the process of hearing aid fitting. Especially after long-term hearing impairments prior to a hearing aid fitting, the immediate full compensation of the hearing loss will often not be tolerated, so that a systematic, successive fine tuning of hearing aid parameters has to be performed over a period of time (“gliding” fitting) in order to find the final settings. To support this process, the fitting modules of many manufacturers offer the possibility to modify target gains based on the level of familiarization ("acclimatization" or "experience" level). In a retrospective study it was examined, to what extent the real fittings match with this picture and the manufacturers proposals.

From the parameter sets of 24 first time users and 29 experienced hearing aid wearers (age 41-88 years, mean 72 years, hearing loss 30-96 dB, mean 55 dB, experience with hearing aids 3.5-57.5 months, 53% bilateral fittings, 45% female), the course of the real adjustments was compared with the manufacturer settings ("First Fit") for different acclimatization levels. Fittings were performed with six current BTE hearing aids from two manufacturers. Target gain was calculated basing on either the NAL NL1-rule (NAL-based) or the DSL [i/o]-rule (loudness based).

Analysis of the data showed that first time users prefer much higher gain values compared to the manufacturer proposals, so that actual acclimatization steps seem to be too conservative. For first time users, loudness-based fittings yielded more stable courses, whereas NAL-based fittings seem to be more appropriate for experienced hearing aid users.

In general, the acclimatization phase was terminated after about 3 months, and first time users do not need more sessions to reach a stable fitting than the experienced users. High frequency losses could be compensated for most easily.

B14

Measurement of the modulation transfer function of nonlinear hearing aids

Inga Holube, Martin Hansen, Stefan Fredelake, and Rainer Blum, Institute for Hearing Technology and Audiology, Germany

A new method for measuring the electro-acoustical performance of modern nonlinear hearing aids is presented. Methods for measuring the electro-acoustical performance of hearing aids are currently defined in standards like, e.g., IEC 118. However, the measurements described there are mainly based on the experience with linear devices or very simple compression hearing aids. The performance of modern hearing aids however, containing multi-channel, multi-microphone, nonlinear, and signal-adaptive processing of the incoming sound, is only very poorly reflected by the measurement results of the current standards.

B15

Can pure tone real ear measurements be used to predict hearing aid gain for speech?

Arne Leijon, KTH – Signals, Sensors and Systems, Sweden

Nonlinear hearing aids adapt their signal processing depending on the input signal. Nevertheless, it has been proposed (Scollie et al., 2002) that hearing-aid gain response for real-life sounds, such as speech, can be predicted sufficiently accurately using pure-tone measurements, if proper correction factors are used to account for the difference between broadband and narrow-band test signals.

B16

Thresholds for brief tones in modulated noise tell a lot about hearing

Ann-Cathrine Lindblad, and Åke Olofsson, Björn Hagerman,Karolinska Institutet, Sweden

The international evaluation of the Modulation Transfer Function for predicting speech intelligibility in technical systems and various environments gave us inspiration for using a similar approach for measurements on the ear, especially the hearing impaired ear. To include the ear in the transmission chain and be able to predict speech recognition from the whole chain, hearing aid included, seemed worth testing. To obtain a Psychoacoustical Modulation Transfer Function we measured thresholds for brief tones at the peaks and in the valleys of 100% intensity-modulated octave band noise at modulation frequencies corresponding to the normal intensity fluctuations of speech. In analogy with the MTF method where a good system does not fill in the silent intervals in the test noise, we expected good speech recognition results from an ear with a large difference between peak and valley thresholds. However, we stumbled on the nonlinearity of the cochlea and obtained different threshold values when the noise level was varied. Instead we decided to test the possibility of fitting hearing aids according to the principle that a level with a large difference between peak and valley thresholds should be optimal for speech recognition. Perhaps the level dependence could serve as an individual level distortion curve for the STI. We made some good attempts in lab tests, but could not continue due to lack of funding.

B17

Outcomes of FM technology use for adults with significant hearing loss

Rachel McArdle, Bay Pines VA Medical Center, Colleen Noe, James Quillen VA Medical Center and Theresa M. Chisolm, University of South Florida

Numerous investigations have demonstrated that individuals with adult onset hearing loss are able to achieve better speech understanding in noise with the use of FM systems than with the use of hearing aids alone. Despite this advantage, there is a general lack of acceptance of FM systems among potential adult users. While reasons postulated for non-use include cost, complexity of device use, and increased attention to the hearing loss due to device visibility, it is also the case that we have no clear-cut criteria for determination of candidacy for FM use in adults, nor do we have a systematic approach for device use training. The objectives for this study were : (1) establish criteria for determination of candidacy for FM use in adults by assessment of residual hearing abilities, auditory demands of daily communication activities; and, specification of the physical and socio-cultural environment in which communication interchanges occur; and (2) develop a systematic approach to FM use training that focuses on identification and resolution of handicapping situations as they occur at specific points in time to promote positive intervention outcomes. Baseline and post-treatment measurements of the Client Oriented Scale of Improvement (COSI), Communication Profile for the Hearing Impaired (CPHI), and Marketrak instruments were collected on 36 subjects. Each subject identified three communication goals and was instructed on the use of the FM system for each goal during three consecutive, two-week trial periods. Results showed that lack of satisfaction with current hearing aids in specific listening environments identified on the Marketrak and self-perceived communication difficulties as measured by the CPHI were good predictors for FM candidacy. Outcomes data indicated improvements in self-perceived communication problems on the CPHI [$F\ (1,30) = 22.60$, $p = 0.00005$, $MSE = 78.00$] following the 6-week trial period with FM technology. Marketrak data showed significant increases in hearing aid satisfaction when the hearing aid was coupled to an FM system in multiple listening situations including environments with competing noise [restaurants (÷2 = 8.92, p = .003)]; reduced visual cues [cars (÷2 = 16.06, p = .000)]; and distance from the speaker [place of worship (÷2 = 25.33, p = .000)]. The results demonstrate positive outcomes from FM use on self-perception of communication performance, improved satisfaction with listening in specific situations, and success in reaching individualized communication goals. [This work was supported by Phonak].

B18

Distorted output from hearing aids because of Electromagnetic Interference (EMI): Some preliminary findings

Kartik Narayanan, Aditee Ashok Khaladkar, Ajith U. Kumar, Ajish K Abraham and M. Jayaram, All India Institute of Speech and Hearing, India

Electromagnetic interference (EMI) produced by a digital cellular telephone and picked up by hearing aids is a major source of frustration for hearing aid users. Such EMI distorts the output of a hearing aid and affects speech perception. The purpose of this study was to (a) document the nature of distortion in hearing aid output as a result of EMI, (b) quantify the reduction in the effect of EMI as result of shielding of the cellphones and, (c) to test the effect of EMI on speech identification scores in a group of individuals with sensorineural hearing loss. Experiments on the nature of distortion and shielding effect employed pure tone signals as the stimuli. Initial experiments carried out indicated that EMI effects on hearing aid performance were negligible when the cellular phone was more than one metre away from the hearing aid. The distortion measurements were made under two conditions; (a) when the hearing aid and cell phone were placed adjacent to each other, and (b) in a second condition when the two were placed one metre away from each other. Experiments were carried out on ear level digital and analogue hearing aids. Three well-known ear level digital hearing aids, one ear level analogue hearing aid and one model of digital cell phone were tested. Performance of body worn analogue hearing aids were also investigated as these hearing aids, rather than digital hearing aids, are more relevant to Indian context (more than 75% hearing aids used in India are body worn). Speech identification tests employed phonetically balanced monosyllables. A two-channel oscilloscope was used to analyse hearing aid output. The hearing aid under test was coupled to the oscilloscope through a HA-2 coupler, microphone, preamplifier and a sound level meter. Preliminary results of analysis have shown some interesting results: (a) EMI introduces frequency, amplitude and phase distortion in the output of the hearing aids;(b) EMI distorts hearing aid output when the hearing aid is in the microphone mode, although it is less when it is in the Telecoil mode; (c) the level of distortion in the hearing aid output was higher in the analogue hearing aids compared to digital hearing aids; (d) shielding in the form of a 2mm thick 10" * 10" copper plate placed adjacent to the mobile phone considerably reduced noise in the output of the digital hearing aids, but had minimal effect on the output of analogue hearing aids; and (e) speech identification scores for speech presented through the digital hearing aid in the presence of cellular phone (on) significantly decreased in a group of persons with moderate sensorineural hearing loss at a distance of 0.8 metres. However, the drop in speech identification scores for the presented speech material presented through analogue hearing aids was more significant.

B19

The challenge of ear scanning for manufacturing hearing instruments

Thomas Powers, Velde Therese, and William Lesiecki, Siemens Hearing Instruments, Inc.

Realizing the dream of eliminating physical ear impressions for hearing instruments is getting closer. With increased miniaturization of digital optics, this dream may become reality soon. While identifying the optimal scanning technology for capturing an image of the ear canal is a significant challenge, the development of the supporting technology and the user interface also pose a formidable challenge. Without supporting technology to transform the ear scan from a set of numbers to a physical shell, the ear scan is useless. The embodiment of an ear scanner has to take into account a number of operator and patient factors. The purpose of this presentation is to discuss the development of the necessary supporting technologies, along with the technical challenges of scanning the ear and also to present a proof of concept for direct ear scanning.

The supporting technology is actually a conglomeration of advanced technologies: three-dimensional modeling software, rapid prototyping technology and information/network technology. All aspects involving these technologies have to be melded together to move efficiently and effectively from scan to hearing instrument shell.

The scan output is a series of numbers defining a point cloud that represents the geometry of the ear. Sophisticated three-dimensional modeling tools take the raw scan and shape it to the final hearing instrument shell. The outside geometry of the raw scan is sculpted to the requested shell size (i.e. hearing instrument style such as CIC or Full-shell). The internal structures are placed inside the sculpted shell; specifically, the vent size, shell wall, and receiver opening are defined in the final digital shell.

A digital shell cannot be produced with "conventional" shell manufacturing processes; therefore, digital manufacturing technologies are implemented to build the shell. Rapid prototype technologies provide a digital platform for such manufacturing. All aspects of these technologies, such as material type and manufacturing tolerances, must be adapted for the application of building a hearing instruments, which represents one of the first applications of digital manufacturing techniques to consumer products.

A direct ear scanner must be easy to operate for the audiologist and must be a "comfortable" experience for the patient. Some of the user and patient considerations that come into play are the length of time to take a scan, stability of the head and scanning apparatus, comfort of the probe, and removal of cerumen and hair.

The anatomy of the ear presents a formidable set of challenges. The ear is a winding tube that ends (or begins, depending on your point of view) with a half sphere with peaks and valleys. The diameter and curvature of the canal limit the type of probe that can be inserted. Viewing in the canal is optically very near field, while viewing the pinna is relatively far field. The entire (or nearly entire) length of the canal must be viewed without any distortion of the canal, e.g. pulling back the ear to straighten the canal for otoscopy.

B20

Toward improving laboratory estimates of real-world directional hearing-aid performance

Lawrence Revit, Revitronix, Michael Valente, and Karen Mispagel, Washington University School of Medicine

According to empirical and theoretical data, studies of directional hearing-aid performance conducted in laboratories generally overestimate performance obtained in field studies. Recently an experimental sound system was designed to present accurately in a laboratory setting the acoustic conditions of an actual noisy restaurant. An earlier validation of that sound system found no significant differences in directional performance on HINT thresholds using the laboratory restaurant simulation for competing noise compared to using the actual restaurant for competing noise, across a broad range of directional microphone types, for normal-hearing subjects.

B21

Localization with linear and wide-band multicompression hearing aidswide dynamic range multichannel compression hearing aids

Helen Simon and Al Lotze, Smith-Kettlewell Eye Research Institute, E. William Yund, and Christina Roup, VA NCHCS

Advances in electronics have made true wide dynamic range multichannel compression (WDRMCC) hearing aids (HAs) available to hearing impaired (HI) patients, but questions remain about the value of WDRMCC signal processing in general as well as the benefit of particular WDRMCC HAs. Many patients have difficulties with conventional HAs, but will they have fewer difficulties or gain greater benefit from advanced WDRMCC HAs in speech perception and/or localization? The primary purpose of our research was to evaluate WDRMCC signal processing for speech perception and localization. The principal goals of the localization research were: 1) to measure localization performance for patients using WDRMCC signal processing; and 2) to study the nature and time course of the new hearing-aid users' adjustment to this type of amplification for sound localization. Throughout the research, localization with WDRMCC hearing aids was compared to localization with the same hearing aid fit as a linear device. Results to date of this pioneering study of the effect of independent binaural WDRMCC hearing aids on sound localization in the laboratory and in the patients' normal auditory environment will be discussed.

B22

QuickSIN soundfield normative data

Carla Sims and Laurel Christensen, GN ReSound

The QuickSIN was developed as a means of measuring a person's ability to understand speech in noise. The test was designed to be a faster version of the original Speech-in-Noise (SIN) test developed by Etymotic Research (1993). The QuickSIN contains lists of sentences presented with four-talker babble noise at various intensity levels. The results give an estimate of signal-to-noise ratio (SNR) loss. SNR loss is defined as the increase in signal-to-noise ratio (in decibels) required by a hearing impaired person to understand speech in noise, as compared to a normal hearing person. Determining SNR loss may not only help clinicians choose appropriate amplification, but may also aid in counseling the patient to set realistic expectations of the hearing aids. During the development of the QuickSIN, actions were taken to ensure reliability and validity of the test measure. Naturally, this included using the test measure with both normal and hearing-impaired subjects. The test was administered to normal hearing subjects under headphones. However, it was not administered to normal hearing subjects in a sound field condition.

B23

Results of subjective benefit measures: The user and the significant other

Tamara Thornton and Laurel Christensen, GN ReSound

The role of the significant other (SO) in the hearing aid selection, fitting, and counseling process has received growing attention in recent years. SO's can often influence the outcome of the hearing aid fitting in several ways through their interactions with and observations of their hearing impaired partner. Although several outcome measures are available to determine the subjective effects of amplification for the hearing aid user, few are specifically targeted for the SO. The Profile of Hearing Aid Benefit for the Significant Other (PHAB-SO; Flamme, 2001) was developed to provide a method for the subjective perspective of the SO to be evaluated and quantified. To date, this new questionnaire has not been analyzed for a large number of subjects or correlated with any other measures of subjective benefit.

B24

Directional hearing using bilateral hearing aids

Tim Van den Bogaert, Marc Moonen, Lies Royackers, and Jan Wouters, Katholieke Universiteit Leuven, Belgium

By using two ears normal hearing people are able to localize sounds correctly in the horizontal plane, and are able to understand speech better in noisy scenarios where noise and target are spatially separated. Although a lot of studies have been done on normal hearing and hearing impaired persons, little research has been done on directional hearing in bilateral hearing aid users.

B25

A measurement of sound level perception when using the bone-anchored hearing aid (BAHA) for trans-cranial stimulation of indivuduals with single-sided deafness

Andrew Vermiglio and Sigfrid D. Soli, House Ear Institute

The Bone Anchored Hearing Aid (BAHA) delivers bone-conducted signals to the cochlea. The goal of this study was to determine the output characteristics of the BAHA Compact and to determine the equivalent loudness in dBHL for tones delivered via direct audio input to the BAHA. The output from the BAHA Compact was determined by measuring the electrical output from a skull simulator. Input/output curves were measured for audiometric frequencies 250 Hz–6000 Hz for both acoustic and direct input. Equal loudness measures across audiometric pure tone frequencies 250-6000 Hz were obtained for 6 subjects with single-sided deafness and normal pure tone thresholds for the better ear. For most subjects and conditions, stimuli were delivered to the direct audio input at levels that would produce essentially the same output level as the maximum output level to acoustic stimuli. Each subject heard the tone from the BAHA Compact first and then acoustically from the soundfield speaker. The subject's task was to report on the level of the second tone in reference to the first tone (louder, softer, or the same). Loudness matching to the same reference levels across subjects revealed a range in loudness perception of 7 to 30 dB across subjects. The loudness matching protocol may be a valuable tool for BAHA evaluations.

B26

Towards a physiological model of human hearing loss

William S. Woods and Timothy D. Trine, Starkey Laboratories, Inc.

Schuknecht (1993) described three basic pathologies visible under the light microscope thought to underlie presbycusis: sensory cell loss, strial atrophy, and ganglion cell loss. He associated each loss with certain audiogram shapes – for instance, strial atrophy with flat losses. Although there exist several animal-model constrained attempts to relate cochlear state to hearing function (e.g., Delgutte, 1996; Heinz et al., 2001; Schmiedt et al., 2002), none has taken the opposite approach of determining what model parameter values (e.g. threshold distribution of auditory nerve fibers) would be required to match human performance. We have taken this approach, incorporating processes, such as peripheral filtering and compression and a population of nerve fibers with varying spontaneous rates and thresholds, considered basic to representation of information on the auditory nerve. Auditory nerve activity is then evaluated using signal detection theory (following Colburn et al., 2003; Heinz, 2000) to examine model performance in detection and discrimination tasks. The method used allows rapid development and evaluation of different model states, and readily permits manipulation of such factors as the health of the stria vascularis, inner and outer hair cells, and spiral ganglion cells. We are applying this model to work such as Schuknecht's, in which audiogram and cochlear state data are both available. This work is an extension of work we presented on "dead regions" at IH-CON '02.

In this presentation we demonstrate that this simple and realistic model can easily (i.e., with few parameters) match normal human auditory tone detection and discrimination performance in quiet. We then compare predicted audiograms of "unhealthy" models to hearing-impaired listeners' audiogram and cytocochleogram data from Schuknecht's and other's work. We also demonstrate the usefulness of such a model through an evaluation of Schmiedt et al.'s (2002) metabolic model for presbycusis. Finally, we provide qualitative comparisons to other models and discuss how further anatomical studies of human temporal bone samples could be used to answer questions concerning the validity of the current model.

B27

The modification of critical-band based frequency compression using cepstral analysis

Keiichi Yasu, Kei Kobayashi, Takayuki Arai, Sophia University, Japan

This study describes the modification of critical-band based frequency compression by Yasu et al. [Acoust. Sci. & Tech., 25(1):61-63, 2004].It is known that the auditory filter shape of hearing impaired patients is wider than that of normal hearing, which causes the loss of frequency selectivity (Grasberg and Moore, 1986). We focused on the characteristics of wider auditory filter of hearing impairment and developed an algorithm, that we called "critical-band based frequency compression," to compensate for the interference of adjacent auditory filter.

Posters for Session C should be put up by 8 A.M. Saturday, August 28, and taken down after 10 P.M. Saturday, August 28 or before 7 A.M. Sunday, August 29. 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

Optimization of a fast method for determining psychophysical tuning curves

José Alcántara, Brian Moore and Karolina Kluk, University of Cambridge, UK, and Alex Sek, University of Poznan.

Psychophysical tuning curves (PTCs) are traditionally measured by determining the level of a narrowband noise required to mask a fixed, low-level tone, for several masker centre frequencies. PTCs are often used to assess the frequency selectivity of the auditory system and they have also been used to detect "dead regions" in the cochlea and to define the limits of the dead regions. However, the traditional method is too time consuming for use in clinical practice. We have evaluated and refined a fast method for determining PTCs, based on the use of a sweeping band of noise. The fixed sinusoidal signal is pulsed on and off regularly and is masked by a band of noise, whose centre frequency sweeps over a range of two octaves during four minutes. A Békésy method is used to adjust the masker level required for threshold; the subject presses a button to indicate that the signal is audible, and releases it when the signal is inaudible, and the masker level is adjusted accordingly by the computer. The fast method was initially evaluated using normally hearing subjects and showed good agreement with the results of the traditional method. The shapes of the PTCs, the slopes on the low- and high-frequency sides, and the positions of the minima were very similar for the fast and traditional methods. The fast PTC method was also evaluated using hearing-impaired subjects with dead regions. Preliminary results also show a good agreement with the traditional method. Additional experiments have been conducted to optimize the rate of change of level of the masker and the masker bandwidth. Also, an objective method has been developed for estimating the tip frequency of the PTC.

C2

Time efficient contrast-enhancing frequency shaping and multiband compression in hearing aids

Shahab Ansari, Harjeet Bajaj, Kamran Mustafa, and Ian Bruce, McMaster University, Canada

Contrast-enhancing frequency shaping (CEFS) has been shown to produce a better representation of formants in the auditory nerve response of an impaired ear than conventional amplification schemes (Miller et al., JASA 1999; Sachs et al., ABME 2002; Bruce et al., IHCON 2002). The aim of CEFS is to compensate for the broadened tuning curves and elevated thresholds of an impaired ear by adjusting the relative amplitudes of the formants without distorting the spectral envelope between the formants. Multiband compression, on the other hand, has been utilized in hearing aids to compensate for the reduced dynamic range of the impaired ear. Bruce (Physiol. Meas. 2003) has shown that multiband compression and CEFS amplification can work together when used in series, without counteracting one another. This is in contrast to spectral enhancement schemes that utilize some form of spectral expansion, which is not compatible with multiband compression (Franck et al., JASA 1999).

C3

Reverberation suppression by transient enhancement

Aslak Bjerkvik, Norwegian University of Science and Technology, Norway

To understand speech under reverberant conditions is one of the biggest problems for people with hearing impairment. Hearing aids that can reduce the effects reverberation has on the speech signal will therefore be very attractive. We propose a fast and efficient method for suppressing the reverberant tails of the signal, with the intention of deploying the method in digital hearing aids. The main feature of the method is to enhance the transients of the signal, where the direct field to reverberant field ratio is maximal. This processing method also gives a desired lift of important modulation frequencies in the signal. The method works well together with a slow acting compression scheme, which is important for the method to be implemented in a hearing aid. The method gives a subjective reduction of the reverberant parts of the signal, and results from listening test with focus on speech intelligibility will be presented.

C4

Virtual environments for standardized validation of amplification benefit: Recording and reproduction

Robert Ghent and Michael Nilsson, Sonic Innovations, Inc.

A sound field test environment, consisting of a five-loudspeaker array, and designed for verification of benefit from hearing aids with advanced technology features, has been previously described. Based on the HINT, the test environment provides a norm-referenced, clinically valid, and repeatable in situ method for obtaining objective performance benefits from individual hearing aid features such as multi-channel compression, noise reduction algorithms, and directional microphones. The test environment also meets the sound field requirements of the HIA guidelines for establishing performance claims for hearing aids. In order to increase the utility and value of the test environment, it is desirable to create a clinically valid, norm-referenced method for assessing subjective benefit and sound quality in real-world listening situations. This paper describes the first steps in on-going research to qualify, record, and reproduce in said sound field, listening environments representative of those idealized in standardized self-report instruments such as the PHAP and APHAB. Future reports will include correlation studies of the laboratory apparatus to real-world data.

C5

Permissable delay in monaurally fit partially occluding hearing aids

Hannes Müsch, Nazanin Nooraei, Brent Edwards, and Chas Pavlovic, Sound ID

This study seeks to determine the longest permissible processing delay in a monaurally fit, partially occluding hearing aid. In this arrangement there exists an across-ear and a within-ear delay component because listeners receive direct sound in both the low-frequency portions of the aided ear and in the contra lateral ear and delayed sound in the high-frequency portions of the aided ear. Using a magnitude estimation task, six normal-hearing listeners judged the “naturalness” of speech at all permutations of 3, 6, 9, 12.4, 15, 30, and 60-ms delay and 0, 5, 10 and 15-dB high-frequency real-ear insertion gain (REIG). After completing the magnitude estimation task, during which listeners built an internal reference scale, listeners reported the magnitude below which the degradation in naturalness was unacceptable. This estimate was used to derive the longest acceptable delay. To assess the effect of loudness imbalance on the judgments of naturalness, the test was repeated with binaural real-ear recordings that were equalized on a 1/3-octave level basis to match the spectrum of the recording with 0-dB REIG. With 0-dB REIG processing delays of 15 ms were acceptable, but with 15-dB REIG processing delays of 6 ms were just unacceptable. Removing loudness imbalance through equalization improved the naturalness and increased the longest acceptable delay to just below 15 ms. When loudness imbalance effects are accounted for, the longest permissible delay of 15 ms found here is similar to that obtained in previously reported studies.

C6

A new method for verifying the audibility of amplified speech with the BAHA

William Hodgetts, University of Alberta, and Sigfrid D. Soli, House Ear Institute

The audibility of conversational speech for a given hearing aid user depends on the relationship between the softest sound a person can hear and the output of the hearing aid. For air conduction hearing aids, this relationship is easily determined through the use of widely available probe microphone technology since both the softest sound a person can hear and the hearing aid output can be measured in dB SPL. For bone conduction hearing aids (e.g. BAHA) the relationship is not as straightforward. Traditional methods use aided soundfield audiometry to estimate the gain of the BAHA. This gain is added to the input SPL to estimate the output of the BAHA and hence the audibility of speech. An alternative to this approach would be to measure directly the hearing aid user's force level thresholds (dB OFL) and the BAHA's output in dB OFL. Two challenges associated with the traditional aided soundfield approach will be addressed in this poster: 1) force level thresholds cannot be accurately estimated from the standard HL audiogram since the relevant force thresholds are those obtained through the percutaneous BAHA abutment, not the skin covered mastoid, and 2) aided soundfield audiometry to assess the gain of the BAHA provides inaccurate and unreliable results. Cases will be presented comparing the audibility of speech using current BAHA fitting methods (aided soundfield audiometry) with an alternative method using a new BAHA audiometry transducer to assess force level thresholds and a skull simulator to measure the output force level of the BAHA.

C7

Factors affecting the benefits of high-frequency speech audibility

Amy Horwitz, Jayne Ahlstrom, and Judy Dubno, Medical University of South Carolina

Although results of numerous studies have demonstrated the benefits of amplification for adult listeners with high-frequency hearing loss, results of several recent experiments indicate that, under some circumstances, providing high-frequency amplification results in a decrease in speech recognition. One possible explanation may involve the underlying cochlear pathology. Specifically, it has been suggested that hearing thresholds greater than about 60 dB HL reflect complete loss of outer hairs cells along with some loss of inner hair cells and diminished afferent input. Under these circumstances, providing amplified speech to the damaged system may be, at best, ineffective and, at worst, deleterious to speech understanding. An alternative explanation that has been proposed is based on the high signal level rather than the high degree of hearing loss. For example, with the addition of amplified high-frequency speech, a decrease in speech understanding may result from elevated lower frequency thresholds and a subsequent decrease in lower frequency speech audibility due to downward spread of masking. This could occur for listeners with normal as well as impaired high-frequency hearing.

C8

Using coherence to compute the speech intelligibility index

James Kates, GN ReSound, and Kathryn Arehart, University of Colorado

Noise and distortion reduce the sound quality in hearing aids, but there is no established procedure for calculating sound quality in these devices. This presentation introduces a new intelligibility and sound-quality calculation procedure based on the Speech Intelligibility Index [ANSI S3.5-1997]. The SII involves measuring the signal-to-noise ratio (SNR) in separate frequency bands, modifying the estimated noise levels to include auditory masking, and computing a weighted sum across frequency of the modified SNR values. In the new procedure, the estimated signal and noise levels are replaced with estimates based on the coherence between the input and output signals of the system under test. Coherence is unaffected by linear transformations of the input signal, but is reduced by nonlinear effects such as additive noise and distortion; the SII calculation is therefore modified to include non-linear distortion as well as additive noise. For additive noise, the coherence calculation gives SII scores identical to those computed using the standard procedure. The SII standard also accommodates the auditory threshold shift resulting from hearing loss. The reduction in speech intelligibility and/or quality for a hearing-impaired listener can be estimated as the difference in the coherence SII values computed for the input signal (adjusted for the hearing-aid amplification) and the noisy or distorted output signal of the system under test. Experiments with normal-hearing and hearing-impaired listeners using additive noise, peak-clipping distortion, and center-clipping distortion are then used to relate the computed coherence SII scores with the subjects' intelligibility and quality ratings. [Work supported by GN ReSound (JMK) and the Whitaker Foundation (KHA).]

C9

Psychophysical tuning curves: The bronze standard for diagnosing a dead region

Karolina Kluk and Brian C.J. Moore, University of Cambridge, UK

A dead region (DR) is a region in the cochlea with no functioning inner hair cells and/or neurons. Hence, in a DR no transduction occurs and no information is transmitted to the brain about the vibration of the basilar membrane. It is important for hearing aid fitting to detect and define precisely the possible extent of a DR, as amplifying high-frequency components more than an octave above the boundary of a high-frequency DR does not improve speech intelligibility and may lead to feedback and distortion. There are two methods of diagnosing a DR: (I) By measuring the masked threshold of sinusoids in threshold equalizing noise (TEN); an elevated threshold at a specific frequency indicates a DR at that frequency; (II) By measuring psychophysical tuning curves PTCs; a shifted tip of the PTC indicates a DR at the signal frequency. One study has shown discrepancies between the TEN test and PTC results (Summers *et al.*, 2003). So far, PTCs have been considered as a "gold standard" for diagnosing and defining a DR in the cochlea. However many factors not connected with frequency selectivity per se can complicate the interpretation of PTCs. Here, we show that factors like detection of beats and simple difference tones (SDTs) resulting from the interaction of signal and masker can influence the shapes of PTCs. As a result, PTCs can have sharp tips at the signal frequency (f_s) even when f_s falls in a DR. Subjects with near-normal or moderate hearing loss at low frequencies and a DR at high frequencies were tested. PTCs for sinusoidal and noise maskers (80-, 160-, 320-Hz wide) were measured in three conditions: (1) with the main masker only; (2) in the presence of low-pass filtered noise designed to mask SDTs; (3) in the presence of a pair of low-frequency (beating) tones designed to interfere with the detection of beats via the phenomenon of modulation detection interference (MDI). In condition (1), when sinusoidal or 80-Hz wide noise maskers were used, a sharp tip of the PTC was often found at f_s. A second broad tip away from f_s was also found, even when f_s fell in a DR (diagnosed previously with the TEN test). The sharp tip at f_s occurred much less often for the wider bandwidths. For subjects with near-normal hearing at low frequencies and a DR at high frequencies, the tip of the PTC at f_s, which occurred in condition (1), was eliminated in condition (2). This suggests that the sharp PTC tip at f_s was caused by the detection of SDTs. For subjects with moderate low-frequency hearing loss, the sharp tip of the PTC at f_s in condition (1) was largely eliminated in condition (3). This suggests that sharp tip at f_s was caused by the lack of beat cue when the masker frequency equalled f_s, but availability of a beat cue for adjacent frequencies. To minimize the influence of beats, we recommend using noise maskers with a bandwidth of 160 or 320 Hz. In cases of near-normal low-frequency hearing we recommend also using low-frequency noise to mask SDTs.

C10

Development of a 24-speaker sound localization test system

Akiko Kusumoto, Gabrielle Saunders, M. Samantha Lewis, National Center for Rehabilitative Auditory Research, and Peter Jacobs, Jacobs Technologies, LLC

Degradation of sound localization ability is a major problem for listeners with hearing loss and for hearing aid users. It causes the difficulty understanding speech in noise because the listener cannot locate the sound source. Ability to localize sound accurately decreases as the number of loudspeakers increases and/or the separation between speakers decreases. We therefore designed a system that allows us to test localization ability for signals originating from up to 24 loudspeakers with 15 degree in the horizontal plane. In this paper the development of the 24-speaker software-controlled system and its hardware setup will be described.

C11

Is there a possibility of using the source separation technique to hearing aids?

Sang Min Lee, Jong Ho Won, See Youn Kwon, In Young Kim and Sun I. Kim, Hanyang University, Korea

Hearing aid users commonly complain of difficulty in understanding speech in the presence of background noise. To date, to solve this problem, many researchers have suggested various methods, such as multiband compression algorithm, adaptive filtering, and single or multi microphone processing. However they didn't think about the probability of application of source separation technique to hearing aids to enhance the speech discrimination. In this research, we estimate the possibility of application of source separation technique such as independent component analysis (ICA) to hearing aids in the loud background noise.

C12

Temporal properties in clear speech perception

Sheng Liu, Elsa Del Rio, Frank Yu, Paul Meneses, and Fan-Gang Zeng, University of California, Irvine, California

Although clear speech has been shown to benefit hearing-impaired listeners under both quiet and noise conditions, it is still unclear what acoustic cues have contributed to this clear speech advantage. Here we conducted three experiments to study the role of temporal properties in clear speech perception, including speaking rate, temporal envelope, and temporal fine structure. Experiment I used uniform time-scaling to digitally modify sentence duration to match the speaking rate between clear and conversational speech.

Experiment II used non-uniform time scaling to decrease the conversational speech rate by proportionally inserting silent gaps between words. Experiment III used "auditory chimeras" to mix clear speech's temporal envelope with conversational speech's fine structure or vice versa to evaluate the relative contribution of temporal envelope and fine-structure to the clear speech advantage. We measured speech intelligibility over a wide range of signal-to-noise ratios in normal-hearing listeners. Consistent with previous studies, we found that the processing artifacts due to uniform time-scaling reduced speech intelligibility. Different from previous studies, the present non-uniform time scaling did not introduce processing artifact and produced a 2 dB positive effect in speech reception threshold, accounting for about 50% of the observed clear speech advantage. Finally, the auditory chimeras result showed that, while any mismatch between temporal envelope and fine structure decreased speech intelligibility, the temporal envelope cue contributed the most to the clear speech advantage. These results have significant implications for hearing aid designs.

C13

Benefits of high-frequency amplification in quiet and noise for threshold-matched ears with and without suspected cochlear dead regions

Carol Mackersie, San Diego State University, Tracy Crocker, Mayo Clinic, and Rebecca Davis, The Hearing Center

The purpose of this study was to compare speech perception benefit from high-frequency amplification for threshold-matched ears with and without suspected cochlear dead regions. The Threshold Equalizing Noise Test (TEN) was used to identify ears with and without suspected cochlear dead regions. Each ear without dead regions was selected to match the thresholds of one ear with suspected dead regions as closely as possible. Listeners were tested while wearing a digital hearing aid that was programmed to approximate DSL[i/o] targets. Consonant identification in nonsense vowel-consonant-vowel combinations was measured in quiet using a forced-choice procedure. Phoneme recognition was measured using the Computer-Assisted Speech Perception Assessment test (CASPA) presented in spectrally matched noise at signal-to-noise ratios ranging from 0 to +15 dB. Performance was measured for unfiltered stimuli and stimuli that were low-passed filtered at the estimated boundary of the suspected dead regions, ½ octave above and 1 octave above the boundary. Filter settings for the ears without suspected dead regions were the same as settings of the threshold-matched counterpart.

C14

Open-jaw vs. close-jaw ear impressions in children

Srikanta Mishra, P. Manjula, and M. Jayaram, All India Institute of Speech & Hearing, India

It has been reported that earmolds prepared from open-jaw ear impressions have many advantages. A more secure fit, less feedback, reduced occlusion effect, higher attenuation of noise, and better performance in noisy environment are some of the reported advantages. However, no studies addressing the differences in Real Ear Insertion Gain (REIG) between earmolds made from closed-jaw and open-jaw ear impressions seem to have been conducted in the pediatric population, in spite of the fact that ear canals of children have a different shape than those of adults. The present study investigated the differences in REIG between earmolds prepared from ear impressions taken with the jaw-open and jaw-closed position in pediatric population. Also, the comfort levels of the two types of earmolds were informally assessed.

Twenty hearing impaired children, in the age range of 4-11 years and using high-power BTE hearing aids participated in this study. Ear impressions of only one ear for each child, with jaw-open and jaw-closed, were taken separately. REIG measurements were done to compare the earmolds made from two types of impressions. The results suggested that earmolds made from open-jaw ear impressions had higher REIG than earmolds made with conventional ear impressions at frequencies 500Hz to 5000Hz. It appears that, in the case of children, earmolds made from ear impressions taken in the jaw-open position provides a greater gain without feedback than earmolds made from impressions taken in the closed-jaw position. The significance of higher REIG for earmolds made from open-jaw ear impressions in hearing aid fitting and other benefits are discussed. Children who participated in this study neither complained of any discomfort, nor do expressed about the comfort when wearing the earmolds made from open-jaw ear impressions.

C15

Combining psychoascoustics and cochlear modeling: Spectro-temporal discrimination ability of CI users

Elisabet Molin, Arne Leijon, Helene Wallsten, Claes Björnemo, and Torbjörn Nillson, KTH, Sweden

The number of cochlear implant users in the world has steadily increased the last couple of years. With the success of these implants the hopes and expectations of people waiting for cochlear implants have been raised. However, it has been shown in several studies that the benefit of cochlear implants, for instance regarding speech understanding, show great individual variance. The topic of this study therefore is to find both practical and theoretical ways to explain and detect the cause of these individual differences in speech recognition results in adult cochlear implant users.

To this end a novel, psychoacoustic, spectro-temporal discrimination test has been developed and tested on a group of CI-users. The results of this test are compared with the individual scores on phonetically based word tests. The intent being to find any correlation between these two test methods and the benefit in speech recognition for the individual CI-user. The spectro-temporal test is constructed using Gaussian white noise shaped as the long-term spectrum of speech. This noise is then filtered to give alternating increased and decreased spectrum levels in a number of frequency bands. At the middle of the stimulus duration the band levels are shifted.

The goal is to measure the required level difference between spectral peaks and valleys in order for the listener to detect the spectro-temporal modulation, for stimuli with 2,4,8,16, and 32 spectral bands. The test procedure is a transformed adaptive 3I3AFC(6) method, converging at 71% correct responses, which corresponds to a detectability index d'=2.2.

The above-mentioned spectro-temporal test can be used to indirectly estimate the limitation in speech recognition of the CI-user. For this purpose, a model has been constructed for the electrical signal transmission and neural excitation in the peripheral hearing system. By observing the neural patterns created by two different sounds a difference index d' between the sounds can be calculated theoretically, using fundamental concepts of pattern recognition. This index can be directly compared to experimental results from the spectro-temporal discrimination test on a CI-user. In this way the electrical model of the peripheral hearing of a CI-user can thus be tuned to perform equally well as the actual CI-user. The tuned model can then be used to estimate the amount of phonetic information in a speech signal presented to the ear that is successfully transferred to the neurons in the auditory nerve. This calculation uses Hidden

Markov Modeling of the neural pattern sequence caused by the speech signal.

Results on the spectro-temporal discrimination ability will be reported for 20-30 CI-users and for a reference group of ten normal hearing listeners. Preliminary results for six CI-users indicate that the ability of understanding speech is connected with the ability of discriminating small spectral changes over time as indicated by the spectro-temporal discrimination test. Some of the tested CI users showed near normal performance for 2 and 4 spectral bands but severely impaired discrimination ability for 8 or more spectral bands. Preliminary simulations using the electrical model of the peripheral hearing show that we can modify the model as to reproduce the results measured in CI-users, for instance by assuming dead regions of neurons in the cochlea.

C16

The noise reduction index: Bench top estimate of SNR changes

Michael A. Nilsson and Victor Bray, Sonic Innovations

A method has been developed that quantifies changes to the signal to noise ratio at the device level for sentences presented in background noise. The measure is independent of device design, and can be used to measure effects of microphone technology as well as signal processing algorithms. A large set of commercially available devices has been evaluated and results will be presented. The method indicates a wide range of performance across device technologies.

C17

The real benefits of bilateral versus unilateral hearing aid fitting

William Noble, University of New England, Australia, and Stuart Gatehouse, Institute of Hearing Research, UK

The vast majority of audiological practitioners would support offering bilateral over unilateral hearing aid fittings, given a universal understanding that the advantages of bilateral fitting outweigh the disadvantages. While there are numerous laboratory demonstrations of the potential benefits of bilateral fittings, robust controlled trials to demonstrate the reductions in disability, handicap and health-related quality of life, with the power to convince clients or funding agencies, are largely absent. We have previously reported a new disability inventory, the Speech, Spatial, and Qualities of Hearing Scale (SSQ), (International Journal of Audiology, 2004, 43(2), 85-99). In addition to items covering a wide range of speech hearing contexts, the SSQ addresses the ability to divide and switch attention, monitor the spatial and temporal dynamics of the auditory world, and aspects of sound segregation, prosody and listening effort. We hypothesise that existing experiments to investigate the benefits of bilateral hearing aid fittings, which have concentrated on traditional, largely stationary, speech-hearing contexts, may be inadequate given the importance of these additional dimensions as drivers of hearing handicap.

The SSQ was applied to three independent clinical groups: 144 people prior to being fitted with amplification; 118 people with at least 6 months experience of unilateral amplification; and 42 people with at least 6 months experience of bilateral amplification. Matching and statistical control ensured similar audiometric, disability and handicap profiles prior to fitting. For most traditional speech hearing contexts (in quiet, in noise, in groups) there was a benefit in fitting one aid, and little further benefit with two. In contrast, speech hearing in demanding contexts (divided or rapidly switching attention) showed further benefit of amplification in both ears versus one. The directional, distance and movement components of spatial hearing showed no benefit from one hearing aid, whereas all components, but especially distance and movement, showed clear advantage of two. Finally, clarity of sounds, and effort needed in conversation, showed bilateral advantage. The conclusion is that the benefit of fitting two hearing aids will not be readily demonstrated so long as traditional speech hearing contexts are relied on to make the case. Having demonstrated the domains within which benefits of bilateral fittings are likely to occur, the challenge is now to construct performance tests that will allow candidature for bilateral versus unilateral fitting to be accurately predicted.

C18

Relation between a measure of nonlinear distortion and the coherence function

Ake Olofsson, Karolinska Institutet, Sweden

In the previous meeting (IHCON 2002) Olofsson & Hansen reported on the relation between objective measurements and subjective perception of nonlinear distortion in simulated hearing aids from broad-banded signals like speech. We used two input measurement signals which were a Hilbert transform pair (90 degrees out of phase) and defined linearity of the aid as that part of the output signals which were still related by the Hilbert transform and the distortion as the part that was not related by this transform. This resulted in a very simple signal processing to obtain the results.

C19

The effect of car noise on output of hearing aids with limiting and with WDRC

Hua Ou, Greg Flamme, and Ruth Bentler, University of Iowa

Compared to other background noise sources, car noise has excessive high energy in the low frequencies. At the lower end of the frequency range (20 Hz) of the intensity often reaches 90 dB SPL. Mixed in with the person’s own speech and the vehicle’s poor acoustic environment, intelligibility can be compromised. It is unclear what impact car noise has on the hearing aid performance. Until now, there have been few studies addressing about the impact of car noise on hearing aids.

C20

Hearing aids and the perception of annoying and aversive sounds

Catherine Palmer, University of Pittsburgh, Ruth Bentler, University of Iowa, H. Gustav Mueller, Vanderbilt University, and Thomas Powers, Siemens Hearing Instruments

Hearing aid users continue to be disappointed in listening in noise (Kochkin, 2002a,b) with reports of difficulty understanding speech, loudness discomfort, and annoyance in background noise. Hearing aids with noise reduction algorithms are promoted as solutions to understanding and comfort in noise. To date, data from studies employing noise reduction do not support an improvement in understanding in noise (e.g., Bentler, 1993; Walden et al., 2000) or a significant improvement in comfort (e.g., Alcantara et al., 2003). The purpose of this investigation was to employ a noise reduction algorithm and measure a group of moderately hearing-impaired subjects’ perceptions of annoyance and aversiveness of noises. These ratings were compared to ratings prior to using the new amplification and to ratings of normal hearing subjects. The results suggest that a properly fit hearing aid will not reduce annoyance or aversiveness or even maintain this perception as compared to pre-fitting results. In this investigation the carefully fit amplification system returned, on average, the individual subject's perception of annoyance and aversiveness to normal. A normal perception would by definition be different to the perception to which the hearing-impaired individual has become accustom. The results of this investigation suggest the need for realistic expectations counseling related to annoyance and aversiveness of sounds at the time of hearing aid fitting.

C21

A comparative evaluation of single-microphone noise reduction algorithms

Gurjit Singh, Vijay Parsa, Karthikeyan Umpathy, Karthikeyan, and Chen Guo, University of Western Ontario, Canada

The presence of background noise has a tremendous impact on the quality and intelligibility of speech and audio. This is especially true for hearing impaired listeners who have increased difficulty in understanding speech and audio in noisy backgrounds. There has been a major thrust in recent years to combat this problem that resulted in the development of digital noise reduction algorithms for speech and audio applications, including hearing aids. While several noise reduction algorithms and techniques have been proposed in the literature, there has not been a systematic comparison where the relative performance of these algorithms is quantified.

C22

A model for prediction of functional hearing abilities in real-world noise environments

Sigfrid Soli, House Ear Institute, Chantal Laroche, Christian Giguere, and Veronique Vaillancourt, University of Ottawa, Canada

Many jobs require functional hearing abilities such as speech communication, sound localization, and sound detection. Tasks utilizing these abilities often must be performed in challenging noisy environments. Individuals whose functional hearing abilities are impaired by hearing loss may constitute safety risks to themselves, fellow employees, and the general public. Identification of impaired functional hearing ability from diagnostic measures of hearing, such as the audiogram, has been both difficult and subject to legal challenges. In addition, diagnostic measures may not adequately assess the potentially beneficial effects of hearing aids on functional hearing ability.

We have developed an alternative approach to this issue based on Plomp's (1986) speech reception threshold (SRT) model of communication handicap that can address these shortcomings. Our approach utilizes the elevation of the individual's SRT in quiet and in noise above the norm, as measured in quiet or in reference noise conditions with the HINT, to predict functional hearing ability in real-world workplace noise environments. A predictive model that includes the statistical characteristics of the real-world noise and the communication task parameters (i.e., communication distance, vocal effort, accuracy of communication, and whether repetition is permitted) has been developed and validated.

Specific noise environments where hearing-critical job tasks are performed have been recreated in the laboratory from calibrated field recordings. Speech intelligibility, sound localization, and sound detection were measured in these environments. In addition, SRTs, sound localization and sound detection were measured in quiet and in reference noise conditions. The correlations between these measures and the measures of functional hearing ability in the real-world noise environments were used to develop the predictive model and to cross-validate its accuracy with both normally hearing and hearing impaired individuals.

The predictive model will be used to screen applicants and employees for the Department of Fisheries and Oceans Canada (DFO Canada). Screening can be based either on percentiles of normal performance observed in each noise environment, or on performance levels specified by job content experts. Comparable SRT tests in English and French have been developed, as required for Canada.

This presentation will summarize the three years of field and laboratory work culminating in the validation and implementation of the model for prediction of functional hearing abilities in real-world noise environments. Applications of the model by DFO Canada will be described. Emphases will be placed on the methods for statistical characterization of noise environments, since these methods, if proven sound, may allow generalization of the model to a wider range of real world noise environments.

[Supported by Department of Fisheries and Oceans Canada]

C23

Temporal masking curves measured in hearing-impaired listeners

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

Iso-response temporal masking curves (TMCs) show the masker level required to forward mask a fixed-level signal as a function of masker-signal interval. Their shapes depend both on cochlear compression and on the decay of the internal effect of the masker. It has been assumed that the decay is similar for all frequencies, and that the variation of TMC shape with masker frequency measured for normally hearing subjects reflects changes in cochlear compression at the signal "place" (Nelson and Freyman, 1987; Nelson *et al*., 2001; Lopez-Poveda *et al.*, 2002). In subjects with moderate to severe cochlear hearing loss, cochlear compression is reduced or absent. Therefore, the TMC shape should mainly reflect the decay of the internal effect of the masker.

To test the assumption that the decay is similar for all frequencies, TMCs were measured for three listeners with mild-to-moderate moderate cochlear hearing loss. Signal frequencies, f_S, were 0.5, 1, 2, 4, and 6 kHz, and masker-signal frequency ratios were 0.5, 0.8, 1, 1.15, and 1.3. For f_S=0.5-kHz, the TMCs were roughly parallel for all masker-signal frequency ratios. For the other signal frequencies, the TMCs were also roughly parallel for the different masker-signal frequency ratios, except for subject AR for f_S= 1 kHz, where the hearing loss was only 40 dB. In that case, the TMC was steepest for the masker-signal frequency ratio of 1, as has been found for normally hearing subjects (Lopez-Poveda *et al.*, 2002). In contrast to what is typically found for normally hearing subjects, the TMCs did not show distinct segments with different slopes (except for subject AR at 1 kHz and subject RL at 0.5 kHz), although they did tend to become shallower for longer masker-signal intervals. Psychophysical tuning curves derived from the TMCs showed broad tuning for all masker-signal delays (longer delays corresponding to higher overall masker levels), and generally varied little in shape with delay (except for AR at 1 kHz). Overall, these results are consistent with the idea that the decay of the internal effect of the masker is similar, but is not exactly the same, for all frequencies.

C24

An adaptive directional microphone system for quiet and noisy conditions

Steele Brenton, Hayley Fiket, and Peter Blamey, Dynamic Hearing, Australia

A new adaptive directional microphone (ADM) has been developed which overcomes problems associated with fixed directional microphones. This new ADM has a flat frequency response, rather than a low frequency roll-off, therefore allowing more accurate fittings to be made. It has the ability to not only move the null, but to automatically switch to omni-directional mode in quiet, thus removing the need for manual intervention or multiple programs in the hearing aid. In addition, it has a particularly simple architecture, meaning that it is suitable for implementation in low power applications and in applications where minimal processing cycles are available.

C25

On the importance of narratives for hearing aid fitting outcome

Thomas Lunner, Oticon A/S, Denmark, and Marie Öberg, Linköping University, Sweden

Any hearing aid fitting process contains a ‘narrative’, being the explanations and justifications given for the particular sequence of actions and their content. Clinical experience indicates that the ‘narrative’ may be important for the outcome of the fitting process.

Two field studies were conducted with experimental wearable hearing aids including 19 and 18 subjects, respectively. Different narratives were given to the test subjects, where they were led to believe that they wore amplification based on (a) self-adjusted hearing aids (b) computer-adjusted hearing aids or (c) amplification based on ‘newly developed fitting methods’.

In the first study, subjects were provided with two acoustically identical fittings accompanied by either the self-adjustment narrative or the computer-adjustment narrative during consecutive test weeks. Subjects showed rather strong preferences for one or the other fitting, indicating that the narrative affected the preference. The results were supported by interview data.

In the second study, subjects were first provided with non-blinded amplification of both self-adjusted amplification and computer-adjusted amplification in consecutive test weeks. Thereafter, the subjects were then given access to both amplifications (identical with the two previously worn amplifications) in a triple blind test design accompanied by a new narrative telling that they wore amplification based on ‘two newly developed amplification strategies’. The results show that ratings of performance were different depending on the narrative.

As expected, it was not possible to show any objective speech recognition in noise differences between the different test conditions with identical amplification, indicating that the narrative effect was mainly subjective.

The results from the two studies indicate that the outcome of a fitting process is affected by the extent to which the client finds the ‘narrative’ appropriate and reasonable. We have also confirmed what practicing dispensers have always known, that a strong narrative can easily convince a willing client, despite contradictory evidence available via listening to the hearing aid in question. The results do not tell us about the effects of the narrative in the long term outcome; nevertheless, it must be important in the typical hearing aid dispensing process.

C26

R-HINT-E: A realistic hearing in noise test environment

Karl Wiklund, Ranil Sonnadara, Laurel Trainor and Simon Haykin, McMaster University, Canada

Owing to developments in microprocessor technology, such as the decreases in cost, size, and power consumption of devices, it has become possible to develop digitally based hearing aids. This means that increasingly sophisticated signal processing algorithms may be incorporated into future designs, ultimately to the benefit of the customer. However, as the processing capabilities become more powerful and capable of a wider variety of tasks, it is essential that the evaluation capabilities of the designer keep up with the progress that has been made. As it stands now, the currently available hearing in noise tests do not take into account the variety of acoustic environments and scenarios that are of interest to the designer.

To remedy this deficiency, we have developed a virtual audio environment that is an integrated acoustic simulator and processing algorithm test platform. This program, called R-HINT-E (Realistic Hearing in Noise Test Environment) incorporates real room and body effects, as well as the possibility of designing multi-talker simulations involving sources at different locations in the virtual acoustic room. Further, users can develop hearing aid processing algorithms to be run in R-HINT-E in order to test their performance under the simulated conditions.

The foundation of the R-HINT-E project is the virtual acoustics system, which provides accurate simulation of a range of scenarios. This system incorporates real acoustic head-related transfer functions and room impulse responses as measured in our laboratory using a KEMAR dummy. These measurements were carried out for three different rooms of varying levels of reverberation, and in each room for 72 different source positions around KEMAR. In these measurements the source azimuth, elevation, and distance from KEMAR are all variable. Users of the R-HINT-E software can select the impulse response for a desired source location, and given a known source signal, simulate the response as recorded from a set of standard in-the-ear hearing aid microphones. By selecting multiple impulse responses and sources, the user can also approximate the cocktail party effect where several spatially distributed signals are active at once.

In addition to this, R-HINT-E is designed to be flexible, allowing users to add in their own acoustic environments very easily. It is also a simple matter to add in new test sounds or processing algorithms as desired. Further flexibility is gained by allowing R-HINT-E to interface with other programs. This means that a user-written program (e.g. for clinical testing) may make use of some of R-HINT-E's functionality in order to carry out specific tasks relating to environmental simulation, while the user's own program can manage the higher level tasks.

The R-HINT-E program is an ongoing research project directed at serving the needs of the engineering and clinical communities. In addition to presenting the current software and its capabilities, we will discuss the future direction of our work to incorporate more real-life acoustic effects, such as individualized user head-related transfer functions and source-receiver motion into R-HINT-E's simulation capabilities. R-HINT-E will also be used to evaluate digital hearing aid algorithms that are currently in use.

C27

Improvements to a physiological model used for hearing aid evaluation and design

Muhammad Zilany and Ian Bruce, McMaster University, Canada

Computational models have been developed by Carney and colleagues to simulate the responses of auditory nerve (AN) fibers in cat (Carney, JASA 1993; Zhang et al., JASA 2001; Tan and Carney, JASA 2003). The most recent version adds a level-independent instantaneous frequency glide in the basilar membrane (BM) filter, as observed in BM and AN data. This model produces realistic responses to simple acoustic stimuli but has yet to be applied to the study of AN responses to speech. The Zhang et al. version of the model has been modified by Bruce and colleagues (JASA 2003) to study the effects of outer and inner hair cell impairment on the AN's representation of speech stimuli. Several further studies have utilized the Bruce et al. model to study the effects of hearing loss on the neural representation of speech (Bandyopadhyay and Young, J. Neurosci. 2004; Bondy et al., NIPS 2003; Bondy et al., 37th Asilomar Conference on Signals, Systems and Computers, 2003) and for the evaluation and design of speech processing schemes for hearing aids (Bondy et al., Asilomar 2003; Bruce, Physiol. Meas. 2004; Bondy et al., Sig. Proc. 2004).

IHCON 2004 Attendees

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Harvey	Abrams	VA Medical Center	USA
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Stefan	Launer	Phonak AG	Switzerland
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Gil	Magilen	Hearing Center	USA
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Stephen	Mandler	Child and Family Clinic	USA
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Carole	Mayer	California State University, Sacramento	USA
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Michael	Stone	University of Cambridge	UK
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Mark	Winter	Medical University of Hannover	Germany
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William	Woods	Starkey Laboratories, Inc.	USA
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Yongfeng	Wu	Kailuan General Hospital	China
Jerry	Yanz	Micro-Tech Hearing Instruments	USA
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