Issues in Advanced Hearing Aid Research

UCLA Lake Arrowhead Conference Center
Lake Arrowhead, California
May 27–31, 1996

Chair: Arthur Boothroyd
Co-Chairs: Claus Elberling and David Fabry
Organizational Co-Chair: Sigfrid Soli

Sponsors

House Ear Institute

Acoustical Society of America
Daily Schedule

Monday, May 27
3:30 PM  Conference registration
5:00 PM  Social
6:30 PM  Dinner
7:40 PM  Introductory comments
7:50 PM  Evening session
10:00 PM  Social

Tuesday–Thursday, May 28–30*
8:00 AM  Breakfast
9:00 AM  Morning session
12:00 PM  Lunch
5:00 PM  Social
6:30 PM  Dinner
7:50 PM  Evening session
10:00 PM  Social

*A short Special Session and Business Meeting will be held Wednesday at 1:30 PM

Friday, May 31
8:00 AM  Breakfast and checkout
9:00 AM  Morning session
12:00 PM  Lunch, buses leave for airports with box lunches for passengers

Notes

1. The Conference Coordinator is Dorothy Cauley. She will be at Lake Arrowhead throughout the week and is available to assist attendees with matters related to the conference.

2. All conference fees and other payments must be settled before the close of the conference.
Program Summary

Monday Evening

SESSION 1: OVERVIEW OF FOCUS TOPICS: OUTCOME EVALUATION, PSYCHOACOUSTIC AND PHYSIOLOGICAL UNDERPINNINGS, ADVANCED SIGNAL PROCESSING

Denis Byrne
The changing face of hearing aid selection
Graham Naylor
Auditory models and their use in hearing aid research
Guido Smoorenburg
Speech processing hearing aids for the profoundly hearing impaired

Tuesday Morning

SESSION 2: SIGNAL PROCESSING IMPLICATIONS

Wouter Dreschler
Fast nonlinear signal processing for speech perception in quiet and in noise
James Kates
An evaluation of hearing-aid array-processing techniques
Patrick Zurek
Microphone-array hearing aids
Patricia Stelmachowicz
Effects of advanced signal processing on speech: implications for fitting hearing aids to young children

Tuesday Evening

SESSION 3: PHYSIOLOGY, PATHOLOGY, EXPERIENCE, AND HEARING AIDS

Robert Harrison
Developmental plasticity of the central auditory system
Stuart Gatehouse
Acclimatization to amplified speech: relationships between changes in audibility, intensity discrimination abilities, and performance on speech identification tasks
Jont Allen
Contribution of outer hair cells to masking and loudness functions

Wednesday Morning

SESSION 4: OUTCOME EVALUATION AT THE CLINICAL LEVEL

Robyn Cox
Relationship between hearing aid benefit and non-auditory variables
Larry Humes
Associations among various measures of hearing aid performance, benefit, use, and satisfaction
Harvey Dillon
Selecting MPO for hearing aids: A theoretical procedure and experimental validation
Louis Braida
Recent progress on aids to speechreading
### Wednesday Afternoon

**SPECIAL SESSION 4a: ADVANCED SIGNAL PROCESSING AND OUTCOME EVALUATION**

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<td>Yo-iti Suzuki</td>
<td>Development and evaluation of a portable hearing aid based on narrow-band loudness compensation</td>
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<td>Business meeting and election of new Chairs</td>
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**Wednesday Evening**

**SESSION 5: ALTERNATIVE APPROACHES TO FITTING AND OUTCOME EVALUATION**

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<td>Christopher Turner</td>
<td>Frequency-weighting functions for speech recognition using a correlational approach</td>
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<td>David Fabry</td>
<td>Effects of using loudness- versus audibility-based fitting procedures on success with digital hearing aids</td>
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<tr>
<td>Christine Rankovic</td>
<td>Nonsense syllable judgment task for comparison of frequency/gain characteristics</td>
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### Thursday Morning

**SESSION 6: MODELING THE DAMAGED EAR: ITS RELEVANCE TO HEARING AID DESIGN, SELECTION, AND FITTING**

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<td>Herbert Bachler</td>
<td>Implications of modeling sensorineural loss to hearing instrument design and fitting strategies</td>
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<td>Arthur Boothroyd</td>
<td>Phoneme recognition as a function of dynamic range in simulated recruitment with and without high frequency emphasis</td>
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<td>Martin Dahlquist</td>
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<td>Brian Moore</td>
<td>A model of loudness perception applied to cochlear hearing loss</td>
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### Thursday Evening

**SESSION 7: CLINICAL TRIALS TO ESTABLISH EFFICACY**

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<td>Brian E. Walden</td>
<td>Issues in the development of clinical trial protocols for evaluating hearing aid performance and benefit</td>
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<td>Thomas Lunner</td>
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<td>Vernon Larson</td>
<td>Implementation of the NIDCD/DVA Hearing Aid Clinical Trial</td>
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### Friday Morning

**SESSION 8: RECONCILING PHYSICAL PERFORMANCE MEASURES AND CLINICAL OUTCOMES**

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<td>Claus Elberling</td>
<td>Evaluation of non-linear signal-processing hearing aids using speech and noise signals</td>
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<td>Carl Ludvigsen</td>
<td>Estimating the optimum input-output characteristics of a self-adjusting hearing aid</td>
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<td>Sigfrid Soli</td>
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<td>Chaslav Pavlovic</td>
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Monday Evening

7:40 PM: Introductory comments

SESSION 1: OVERVIEW OF FOCUS TOPICS: OUTCOME EVALUATION, PSYCHOACOUSTIC AND PHYSIOLOGICAL UNDERPINNINGS, ADVANCED SIGNAL PROCESSING

Moderator: Arthur Boothroyd

7:50 PM: The changing face of hearing aid selection

Denis Byrne
National Acoustic Laboratories, Australia

Modern hearing aids usually employ some form or forms of compression amplification and many also have multiple memories. These features present new challenges for hearing aid fitting. Various new procedures are being suggested although the majority are based on the same general concept, normalizing loudness. This paper raises several fitting issues and, in particular, questions the acceptance of "loudness normalization" as being self-evidently the optimal basis for prescribing compression. There is no compelling evidence that it is important that, in any given situation, speech should have the same loudness for a hearing aid wearer as it would for a normally-hearing person, or that loudness should be normalized across the different frequency bands of speech. Recent data show that, in some situations, normally-hearing people prefer to listen to speech with some high-frequency emphasis. As the proven benefits of compression apply only to low inputs (improved audibility) and high inputs (improved comfort), the use of compression should not substantially alter the amplification required for average inputs. Because of this, loudness normalization is inconsistent with the research supporting linear prescription procedures which are mostly based on equalizing, rather than normalizing, loudness across frequency. Thus, loudness equalization, which requires non-linear amplification if it is to be accomplished for a range of input levels, is an obvious alternative to normalization. Therefore, there are at least two credible fitting rationales that deserve evaluation. A serious and unresolved practical problem is that the prescriptions derived from either rationale will depend on the width of the speech bands being normalized or equalized. Examples illustrate the essential difference between normalization and equalization procedures which is that only the latter include adjustments to "flatten" the long-term average speech spectrum. The examples, which contrast NAL and DSL procedures with the Fig 6, LGOB, and IHAFF procedures, also show that procedures of the same type can lead to very different prescriptions. This illustrates the difficulty of evaluating different rationales independently of "incidental" differences between particular procedures.

Multiple-memory hearing aid studies indicate that at least some people need different amplification in different situations or to provide the preferred quality which may vary from time to time. Other evidence suggests that a single fitting rationale may not be optimal for all people, all situations, or all degrees of hearing loss. Optimal amplification may be a compromise which meets various auditory requirements (e.g. speech recognition, listening
comfort, detection and localization of sounds) in varying degrees. The best compromise may depend on auditory and non-auditory factors.

This paper argues that no fitting rationale should be taken as self-evidently correct, that all plausible rationales should be evaluated, and that, as far as possible, new procedures should use knowledge gained from research, and should build upon existing procedures, rather than starting from "scratch."

8:30 PM: Auditory models and their use in hearing aid research
Graham Naylor and Claus Elberling
Oticon Research Unit, Eriksholm, Denmark

Auditory models of various sorts are becoming more and more widespread in hearing aid research, as understanding of the processes involved in normal and impaired hearing improves and as hearing aid processing schemes of increasing complexity become technologically feasible. This paper attempts to classify models, both existent and as yet non-existent, according to the auditory phenomena they simulate. The features required of adequate models are discussed and compared with those of existing models in respect of their utility in hearing aid research. As a concrete example, the use of a filter bank-based excitation model to visualize the degradation of spatial elevation information with increasing hearing loss is described.

Definition:
An auditory model simulates one or more aspects of auditory function -- on the basis of an input signal and some relevant variables of the auditory system 'under test', the model generates some output which predicts a corresponding physiological, perceptual, or subjective response in the imaginary listener.

Purposes of auditory modeling in hearing aid research:

1. To understand the impaired auditory system, inspiring new ideas for alleviating impairments
2. To predict the effects of potential new hearing aid schemes
3. To itself form part of new hearing aid schemes
4. To cope with the complex interactions in hearing impairment
5. To identify gaps in knowledge and clinical data

Auditory models, whatever phenomena they simulate, can be classified according to whether they model the statistics of an auditory process or the process itself. Statistical models describe (gross) relationships within measured stimulus and response data. As such, they are generally best at predicting the mean response in a population, and cannot address the fact that a given individual's response deviates from that mean. In a process model, such deviations are explained as variations in model parameters, which are (at least in principle) derivable from measurements on the individual subject.

A number of conditions should be met by an auditory model if it is to be useful in future hearing aid research:

1. It should model processes rather than statistics, since the 'higher-order' effects of hearing loss which we wish to tackle show great interindividual variance
2. It should cope with both normal and impaired hearing
3. It should be verifiable against measurable data
4. Its area of faithful operation (within which the assumptions of the model are fulfilled) should at least in principle be identifiable.
9:10 PM: Speech processing hearing aids for the profoundly hearing impaired
Guido F. Smoorenburg
Hospital Utrecht, The Netherlands

The severely and profoundly hearing impaired often derive very limited benefit from conventional acoustic hearing aids. Yet, they do have some hearing which implies that in most clinics they are not considered to be candidates for cochlear implants. In an effort to help these hearing impaired we tried several speech processing schemes based on feature extraction to be implemented in acoustic hearing aids. The features extracted were fundamental frequency, first formant frequency and first plus second formant frequency. The three signals containing these features were modulated with the over-all signal amplitude and complemented with noise bursts representing voiceless consonant information. Sentence tests, including speech reading, showed that these features, presented without training, yielded about the same scores as linearly amplified speech, optimally adjusted per 1/3-octave frequency band to the dynamic range of each individual. Since the subjects were not at all familiar with the speech-like signals based on feature extraction, the present results suggest that feature extraction may help, at least some individuals, after training. The results will be discussed against the poor frequency resolution found in the severely hearing impaired for both sinusoidal stimuli and formants. Moreover, the results will be compared to those found when multi-band compression schemes are used.
Tuesday Morning

SESSION 2: SIGNAL PROCESSING IMPLICATIONS

Moderator: Claus Elberling

9:00 AM: Fast nonlinear signal processing for speech perception in quiet and in noise

Wouter A. Dreschler, Helen E. van Harten-de Bruijn and Sidonne van Kreveld-Bos
Academic Medical Center, The Netherlands

Hans Verschuure
Erasmus University Rotterdam, The Netherlands

The work on fast nonlinear signal processing to be presented is part of the European HEARDIP project, carried out in close cooperation with B. Kollmeier (Oldenburg) and B.C.J. Moore (Cambridge). The aim is to develop fitting procedures and signal processing techniques for digital hearing aids to enhance speech intelligibility in noisy situations.

Reduced dynamic range in the impaired ear is usually compensated by compression. Digital techniques facilitate the use of intelligent combinations of compression and expansion. Our work focuses specifically on the effects of very fast-acting nonlinear systems that are meant to influence the level differences between successive phonemes: phonemic compression and expansion. We used two approaches. In the approach of the Rotterdam group we used a wide-band system with a frequency-dependent compression characteristic. In order to avoid temporal distortions due to the fast compression system, a digital delay proved to be successful. The parameters to be investigated are:

1. The choice of the attack and release times
2. The use of high-pass filtering to reduce the effect of 'upward spread of masking'
3. The use of different control signals for the compression circuit.

In the approach of the Amsterdam group we used a multi-band system in which the frequency-dependence of the nonlinearity can be chosen with greater flexibility. The parameters to be investigated are:

1. The effects of cross-coupling between adjacent frequency bands
2. The use of different combinations of compression and expansion
3. The effects of 4 to 9 frequency channels
4. The effects of using only a 30-dB range of input levels for the nonlinear processing scheme instead of whole-range compression.

A large number of conditions have been tested. For each condition a carefully matched linear condition was used as a reference. The speech and noise levels were chosen at the levels for maximum speech intelligibility in the linear system. This means that possible advantages due to level effects of compression have been eliminated and the effects found can be regarded as the result of the fast nonlinear processing only.

The results show that very fast compression enhances consonant perception in some conditions in quiet and in noise at positive signal-to-noise ratios. The best results in noise were found for background noises with strong temporal fluctuations for listeners with severe sensorineural hearing losses. The most promising schemes will be transferred into a pocket-size wearable unit and will be evaluated in field tests.
9:40 AM: An evaluation of hearing-aid array-processing techniques
James M. Kates, Gabrielle Saunders and Mark R. Weiss
Center for Research in Speech and Hearing Sciences, City University of New York

Microphone arrays have proven effective in improving speech intelligibility in noise for hearing-impaired listeners, and several array-processing techniques have been proposed for hearing aids. Among the signal-processing approaches are classical delay-and-sum beamforming, superdirective arrays, and adaptive arrays. To directly compare the effectiveness of these different processing strategies, a 10-cm long array was built having five uniformly-spaced omnidirectional microphones. This array was used in the endfire orientation to acquire speech and noise signals for a variety of array placements in two representative rooms. Both digital and simulated analog processing techniques were considered, with the array processing implemented in the frequency domain. The physical performance metric was the steady-state array gain weighted to represent the relative importance of the different frequency regions in understanding speech. The processing comparison indicates that the digital systems are more effective than the simulated analog processing, and that both superdirective and adaptive digital array processing can provide more than 9 dB of weighted array gain in typical rooms.

The array evaluation is continuing with speech intelligibility measurements for hearing-impaired subjects listening through the microphone array. These measurements use the same five-microphone array and test rooms as were used for the array gain determination in order to permit a direct comparison between the array gain and the intelligibility measurements. Four microphone systems are being compared in the intelligibility experiments. These systems are a single omnidirectional microphone, a single cardioid microphone, the array processing using delay-and-sum beamforming, and the array processing using optimal superdirective processing. The speech intelligibility metric is the speech reception threshold (SRT) for monaural and for connected discourse. Data from the experiment in progress will be presented.

10:20 AM: Break

10:30 AM: Microphone-array hearing aids
Patrick M. Zurek
Massachusetts Institute of Technology

Amplification of background noise is a persistent source of interference and annoyance for users of hearing aids. Attempts to reduce background noise by processing the signal from a single microphone have achieved very limited success. If it can be assumed that the desired source comes from a specified direction, then processing the signals from multiple microphones should provide substantial noise reduction.

This talk will summarize the noise-reduction performance that can be achieved with an array of microphones worn on the head. Although the constraint that the array be head-worn results in only a few feasible microphone configurations, the variety of array-processing schemes that can be applied makes for a wide range of possible microphone-array hearing aids that vary on the dimensions of size, implementation complexity, preservation of binaural cues, and noise-reduction performance.

The microphone array systems that are the simplest to implement use fixed (non-adaptive) signal processing, which is usually designed to maximize directivity. The directivity that can
be achieved by very simple fixed processing with a 'large' array -- one in which microphones are mounted on eyeglasses or on a headset -- is 7-8 dB (averaged over frequency). This is a significant degree of directionality, but is only about 3 dB more than can be obtained with an ear-level directional microphone.

Adaptive signal processing is designed to cancel the signals from interfering sources, and can potentially provide much greater interference reduction than fixed processing. However, the noise reduction provided by an adaptive microphone-array hearing aid is strongly dependent on environmental acoustic conditions. Under favorable conditions -- a small number of stationary interference sources in an environment with little reverberation -- an adaptive array system can reduce interference by tens of decibels. Departures from these conditions, with the most serious being reverberation, lead to less noise reduction.

Microphone arrays are usually designed to have only a single-channel output. Recent work in our lab has been aimed at developing microphone-array hearing aids with binaural outputs. These systems are designed to enable a trade between the noise reduction achieved by array processing and the natural advantages provided by binaural hearing.

11:10 AM: Effects of advanced signal processing on speech: Implications for fitting hearing aids to young children
Patricia G. Stelmachowicz
Boys Town National Research Hospital

Many of the hearing aid evaluation and fitting techniques used with adults cannot be applied easily to young children. As such, audibility-based hearing aid fitting procedures often are used with this population. However, audibility of the relevant components of speech will depend upon many factors including: speaker characteristics, speaker-listener distance, speaker azimuth, signal-to-noise ratio, ear canal acoustics, and the instantaneous hearing aid frequency/gain characteristics. The situation is particularly complicated by the use of advanced signal processing algorithms. Unfortunately, traditional audibility models, such as the Articulation Index (AI), do not allow for distinctions among most of the signal processing algorithms currently in use.

The purpose of this paper is to describe a theoretical model for the pre-selection and fitting of advanced signal processing hearing aids to children. Preliminary data will be presented that address the following issues:

1. Is the hearing aid gain measured with pure tones, noise, or click stimuli equivalent to the gain measured with real speech as the input?
2. Can a metric such as the AI be modified to be sensitive to changes in the audibility of the low-level components of speech such as might occur with certain types of signal processing?
3. Can a metric such as the AI be modified to distinguish between and account for the perceptual consequences of peak clipping vs. compression limiting?
Tuesday Evening

SESSION 3: PHYSIOLOGY, PATHOLOGY, EXPERIENCE, AND HEARING AIDS
Moderator: David Fabry

7:50 PM: Developmental plasticity of the central auditory system
Robert Harrison
Hospital for Sick Children, Canada

Recent years have seen increased knowledge regarding plasticity of the auditory system, both in the developing system and in its mature state. The evidence that the central auditory brain can reorganize in response to changes in peripheral input creates significant challenges for the design of hearing prostheses. Thus, whilst a hearing loss of cochlear origin may relate to a relatively stable pattern of hair cell degeneration at the level of the organ of Corti, such stability may not exist in the central auditory system, particularly at cortical levels. It may be possible to design hearing aids that will capitalize on such plasticity. Another rather important notion for hearing aid design relates to the developmental plasticity of the auditory system. There is much evidence that the "wiring" and establishment of neural arrays in the central auditory system during early development depends, in part, on the patterns of neural excitation (both spontaneous and driven) arising from the cochlea. In a young child with a (congenital) hearing loss, the activation of the cochlea through a hearing aid (or for a profoundly deaf child, with a cochlear implant) will be influential in the development of central auditory pathways. In other words, a hearing aid provided to a very young child will have a dual role: one of providing hearing sensation, the other of providing activity to systems which will promote its development. There has yet to be serious consideration of this second factor in the design of hearing aids for the young child.

8:30 PM: Acclimatization to amplified speech: relationships between changes in audibility, intensity discrimination abilities, and performance on speech identification tasks
Stuart Gatehouse
Institute of Hearing Research, Scotland

Experiments to study changes in speech identification abilities following the provision of amplification can show variable outcomes. Potentially implicated mediators are the initial loss of audibility, the change in audibility following on the provision of amplification, and the test material and techniques to assess performance. Analysis of speech feature patterns in these experiments have not proved informative, for reasons probably associated with analytical power. More constrained paradigms (such as investigation of intensity discrimination abilities and loudness growth functions) which allow precise control over the frequency and intensity regions of investigation suggest that changes can occur in ways that are specific to the frequency of amplification in the aided ear only. The current investigation aims to integrate measures of speech identification ability, intensity discrimination abilities, and changes in audibility to further investigate the interrelationships.
Five established users of monaural linear amplification with symmetrical sensorineural hearing loss were studied. Measures of audibility were taken from the contribution of specific frequency bands to the articulation index and were derived with and without amplification as used by the listeners for the octave frequencies of 500, 1000, 2000 and 4000 Hz. Intensity discrimination was studied at frequencies of 500, 1000, 2000, 3000 and 4000 Hz and sound pressure levels of 65, 80 and 95 dB in both the aided and normally unaided ears. Speech identification abilities were assessed using the Four Alternative Auditory Feature (FAAF) test which was band-limited in three frequency bands between 0.1 - 1.2 kHz, 1.6 - 2.4 kHz and 3.6 - 6.0 kHz, and presented in quiet at sound pressure levels of 80 and 95 dB.

The results show significant changes in intensity discrimination abilities at high presentation levels (such that difference limens for intensity are improved in the normally aided ear compared to the normally unaided control) which are maximum for a frequency of 2000 Hz and not apparent for frequencies of 500 and 4000 Hz. Speech identification performance also showed significant advantages for the aided versus the normally unaided ear at the high presentation level (95 dB SPL) in the mid-frequency band (1.6 - 2.4 kHz) only. Each of these experimental findings closely tracks the changes in audibility delivered by the hearing aid, which themselves are maximal at 2000 Hz.

Thus the results show a close correspondence between changes in audibility, changes in intensity discrimination abilities and performance on a speech identification task, both across ears and across frequency. The results suggest that the changes that occur in speech identification abilities following the provision of amplification (acclimatization effects) are strongly frequency-specific and related to both audibility and changes in underlying psychoacoustical function. The implications of these findings with regard to underlying mechanisms are discussed.

9:10 PM: Contribution of outer hair cells to masking and loudness functions

Jont B. Allen
AT&T Bell Laboratories

We now know that outer hair cells (OHC) nonlinearly compress the dynamic range of basilar membrane motion (Rhode, JASA 1971; Ruggero, JASA 1990) and the neural response (Yates, Hearing Research 1989), extending the otherwise limited dynamic range of the IHC response. This role of the OHC may be quantified using many objective measures. Loudness, measured by Fletcher and Munson's loudness balance method (Fletcher, JASA, 1933), shows a 1/3 power law compressive function of intensity (i.e., Stevens' Law). This dynamic loudness compression is provided by normal OHC function.

Recruitment results when there is loss of this compression, as when OHCs are damaged (Carver, Handbook of Clinical Audiology, 1978). Masking patterns provide a psychoacoustic measure of cochlear nonlinearity. The OHC compression shows up in the masking growth curves of Wegel and Lane (Phys. Rev. 1924) as the upward spread of masking. The tails of tuning curves are also evident in the masking data as the elevated threshold for the upward spread of masking. These masking data also seem very similar to two-tone suppression. Single frequency otoacoustic emissions and distortion product otoacoustic emissions are an objective measure of the OHC nonlinear compression. Thus the normal function of OHCs play an important and quantifiable role in loudness, sensorineural hearing loss, masking, two-tone suppression, and otoacoustic emissions.
Wednesday Morning

SESSION 4: OUTCOME EVALUATION AT THE CLINICAL LEVEL

Moderator: Sigfrid Soli

9:00 AM: Relationship between hearing aid benefit and non-auditory variables

Robyn M. Cox
University of Memphis

Our previous research has explored the relationship between hearing aid benefit and several auditory variables. Significant relationships have been found between benefit and hearing loss, speech input level, signal-to-babble ratio, mid-frequency audibility change, auditory resolution, and hearing aid in-situ distortion. Despite these relationships, a large proportion of the variance in hearing aid benefit has remained unexplained. In this paper we will describe two experiments that have explored the relationships between hearing aid benefit and several non-auditory variables.

In the first experiment, we explored the hypothesis that aspects of cognitive functioning impact the benefit that can be obtained from any combination of values on the auditory variables. Forty-six elderly hearing aid wearers provided data on hearing aid benefit and the auditory variables listed above. In addition, each subject was tested for four cognitive variables: speed of mental processing, working memory capacity, ability to use contextual constraints, and mental flexibility. The goal of the study was to assess the extent of relationships between benefit and the cognitive variables and to determine whether hearing aid benefit prediction could be improved by including cognitive variables.

Individual personality attributes might contribute to the large between-subject variability in self-assessed hearing aid benefit and other fitting outcome measures such as hearing aid use. If so, it might be possible to use personality variables to help predict the success of amplification. This was addressed in the second experiment. One hundred and fifteen current and former hearing aid wearers provided data for self-assessed hearing aid benefit using the Abbreviated Profile of Hearing Aid Benefit (APHAB). In addition, data were collected on four psychological variables (extroversion, belief in powerful others, internal control, and state anxiety) and self-assessed hearing handicap (using the Hearing Handicap Inventory for the Elderly). This investigation assessed the extent to which the four personality variables and handicap are predictive of two outcome variables (hearing aid use and self-assessed benefit) when gender, hearing loss, and age are controlled. (Supported by Department of Veterans Affairs, RR&D Service)

9:40 AM: Associations among various measures of hearing aid performance, benefit, use, and satisfaction

Larry Humes
Indiana University

Results of two clinical investigations will be described in this presentation. In the first study, a small group (N=20) of hearing-aid wearers were followed over a 6-month period following binaural fitting with two-channel, nonlinear ITE hearing aids. Multiple outcome measures, including objective and subjective measures of hearing-aid use, benefit and performance and subjective measures of satisfaction, were obtained from all subjects at several intervals during
the 6-month study. In the second study, 110 new hearing aid users were followed over a 3-month period following binaural fitting with single-channel, linear ITC hearing aids. Objective and subjective measures of hearing-aid benefit and performance and subjective estimates of hearing-aid use were obtained from all subjects in the latter study. For both studies, factor analyses of and correlations among the various hearing-aid outcome measures will be presented. In addition, the stability of the various outcome measures over the duration of the studies will be examined.

10:20 AM: Break

10:30 AM: Selecting MPO for hearing aids: A theoretical procedure and experimental validation

Harvey Dillon, Lydia Storey, and Hugh McDermott
National Acoustic Laboratories, Australia

Despite the need to individually select hearing aid Maximum Power Output (MPO), or Saturation Sound Pressure Level (SSPL), there are few prescriptive procedures for this, and of those that exist, none have been experimentally evaluated. MPO should be low enough to avoid the hearing aid causing loudness discomfort to the aid wearer, and low enough to minimize the chance of the hearing aid causing further damage to hearing. Conversely, MPO should be sufficiently high to enable the aid wearer to achieve an adequate sensation level and to prevent the hearing aid from excessively saturating.

A new procedure for selecting MPO was derived by separately estimating the highest MPO that would avoid discomfort, and the lowest MPO that would achieve adequate sensation level and avoid excessive saturation. The optimum MPO is then estimated to be the point midway between these two limits. The upper limit was derived from several published studies that related pure tone Loudness Discomfort Level (LDL) to pure tone hearing threshold level (HTL). The lower limit was derived by calculating what MPO would be necessary to avoid saturation when continuous discourse at a long term rms level of 75 dB SPL entered the hearing aid. It was assumed that an aid wearer would use the amount of gain prescribed by the NAL-RP gain and frequency response selection procedure. A hearing aid was considered to be saturated when limiting caused the output long term rms level to be reduced by 2 dB relative to the level that would occur in the absence of limiting. The theoretical prescription formula, based entirely on three frequency average HTL, indicates that the acceptable range of MPOs is very wide for people with mild losses, but is vanishingly small for people with profound losses.

An experiment aimed at evaluating the new procedure contained a laboratory and a field component. In both, we aimed to determine the upper limit and lower limit of acceptable MPO settings for each of 34 subjects with losses ranging from mild to profound. In the laboratory component, subjects adjusted MPO upwards until loudness discomfort occurred, and downwards until the quality of the stimuli deteriorated in some way. In the field experiment, subjects wore programmable, multi-memory hearing aids in which the two programs were identical except for the MPO adjustment. After each listening period of a week or more, subjects reported which program they preferred and why. The MPO settings were varied adaptively by the experimenter for each successive listening period. Both the laboratory and field studies confirmed that MPO selection becomes more critical as hearing loss increases, and that the theoretical procedure provided good estimates of the optimum MPO for most subjects.

The importance of MPO selection for the severely and profoundly impaired suggests that the shape of the MPO curve might also be important for these subjects. An experimental portable
digital signal processing hearing aid incorporated an output limiting compressor, designed to shape MPO to match the individual subject's LDL contour. The processing appears to allow increased listening comfort, relative to conventional AGC, without any degradation in intelligibility.

This research is supported by the Cooperative Research Centre for Cochlear Implants, Speech and Hearing Research.

11:10 AM: Recent progress on aids to speechreading
L.D. Braida, M. Bratakos, P. Duchnowski, D.S. Lum, P.M. Nadeau and M. Sexton
Massachusetts Institute of Technology

We describe recent work at M.I.T. on the development of two types of aids to speechreading: aids based on visual presentation of Cued Speech and aids based on acoustic presentation of the envelopes of bands of speech.

We are studying aids that use Automatic Speech Recognition (ASR) to derive discrete visual symbolic supplements similar to those used in the Manual Cued Speech system. In this system the talker uses discrete hand positions and shapes to provide distinctions between consonants and vowels that are often confused in speechreading. Highly trained receivers of Manual Cued Speech can achieve nearly perfect reception of everyday connected speech materials at normal speaking rates through the visual sense alone. To understand the benefits that might be derived from automatically generated cues, we evaluated the reception of videotaped sentences that were dubbed with hand shapes corresponding to the phonemes identified by several simulated speech recognizers. In these simulations, the hand shapes were displayed at discrete positions about the face, without articulated movement from one position to the next. The intelligibility of perfectly cued sentences, corresponding to an ideal recognizer, was only slightly lower than that of manually cued sentences. When cues were derived from imperfect recognizers, intelligibility declined, but substantial benefits to speechreading were observed when realistic recognizers were simulated. We will discuss the implications of this research for the design of real-time cueing systems based on ASR technology.

Grant et al. (Q. J. Exp. Psych., 43A 621-645) showed that the amplitude envelopes of one or more spectral bands of the speech waveform can be effective speechreading supplements for listeners with normal hearing. We are studying aids that present such supplements acoustically. Since listeners with very severe hearing impairments are likely to be the primary beneficiaries of such aids, the envelopes are presented by modulating low-frequency tones. Recent field trials of wearable aids that provide such signals indicate that some adults with long-standing hearing impairment find the aids preferable to conventional hearing aids. We will discuss the implications of these findings for future research on this type of supplement.
Wednesday Afternoon

SPECIAL SESSION 4a: ADVANCED SIGNAL PROCESSING AND OUTCOME EVALUATION

Moderator: Arthur Boothroyd

1:30 PM: Development and evaluation of a portable hearing aid based on narrow-band loudness compensation

Y. Suzuki¹, K. Ozawa¹, T. Sone¹, F. Asano², M. Ohashi³, H. Hidaka⁴, S. Takahashi⁴, T. Kawase⁴ and T. Takasaka⁴

¹Research Institute of Electrical Communication, Tohoku University, ²Electro-technical Laboratories, Tsukuba, ³Ono Sokki Co. Ltd., Yokohama, ⁴Dept. of Otolaryngology, Tohoku University School of Medicine

Persons with sensorineural hearing loss have a narrow audible range, causing loudness recruitment and an abnormal loudness function. Moreover, the distortion of the loudness function depends on frequency. To deal with this problem we have built a fully digital hearing aid with frequency-dependent compression. Compression characteristics are chosen so that the output of the aid becomes equal in loudness to that perceived by normal listeners for each 1/1 octave band centered from 250 Hz to 4 kHz. The compression characteristics for the five frequency bands are defined by Loudness Compensation Functions (LCF), which describe the relationship between the loudness functions of an impaired listener and an average loudness function of normal listeners. The input signal is first AD-converted then short term frequency spectra are calculated using FFT. The spectra are band-averaged for each 1/1 octave band and the band levels are input to the LCFs. The outputs of LCFs yield gains which make the instantaneous output signal for each frequency band as loud for the impaired listener as it would be for an average normal listener. The gains are interpolated over frequency and realized by a single digital filter; the frequency sampling digital filter. The algorithm is called CLAIDHA (Compensating Loudness by Analyzing the Input Digital Hearing Aid).

The newest model, the fourth model, was completed in the beginning of 1995. This was approved for sale by the Japanese Ministry of Health and Public Welfare in July 1995, and has been sold under the name of Cleartone since September 1995. The aid uses a Motorola DSP-56166 processor and can be kept operating for over 20 hours with four standard LR6 dry batteries. The size is 70 x 130 x 24 mm.

It is quite important for a good fitting of the aid to measure loudness functions of a specific impaired listener effectively and correctly. Thus a PC-based system for measuring loudness functions by the category subdivision scaling method was developed. The measured loudness functions are converted to LCFs and transferred to the aid using a serial interface.

The aid has been fit to more than one hundred impaired listeners during the last six months at a clinic which was established for the fitting of the aid in Sendai, Japan. Performance has been evaluated by a Japanese monosyllabic speech test and a questionnaire survey. The results show a relative improvement of speech intelligibility compared with that for a linear system. The questionnaire survey shows that some 60% of the listeners are quite satisfied with the performance of the aid. Moreover, a preliminary experiment to examine the benefits of binaural use of the aid is being conducted. Some results of the experiment will be presented at the symposium.
Wednesday Evening

SESSION 5: ALTERNATIVE APPROACHES TO FITTING AND OUTCOME EVALUATION

Moderator: Robyn Cox

7:50 PM: Frequency-weighting functions for speech recognition using a correlational approach

Christopher W. Turner and Karen A. Doherty
Syracuse University

In these experiments, we use the Correlational Method to estimate how listeners "weight" (or use) the information contained in various frequency bands of speech. Several questions are addressed: 1) Can this method be used to determine reliable weighting functions for speech recognition? 2) Does this method yield different results than are obtained in traditional (and tedious) filtering experiments (as in Articulation Index procedures)? and 3) What weights do hearing-impaired listeners place upon speech information in regions of normal and impaired hearing, and are their weighting functions different from those of normal hearing listeners?

Speech stimuli were filtered into 4 frequency bands and then each band was degraded by mixing it with a randomly chosen level of a corresponding frequency band of noise prior to presentation to the listener. From the trial-by-trial data record, a point biserial correlation was computed between the listener's response and the noise degradation within each frequency band. The stronger the correlation, the greater the influence that given frequency band had upon the listener's recognition performance. Results to date indicate that the correlational method provides a very efficient procedure for determining frequency weighting functions for speech recognition. This procedure has attractive possibilities for evaluating the effectiveness of hearing aids. We also found that the frequency weighting functions obtained by this new method for normal-hearing listeners are different from those obtained with traditional filtering experiments. We propose that this discrepancy arises from the fact that filtering experiments measure how much information is "available" within a restricted frequency band of speech, whereas the correlational approach measures how much the listener actually "uses" this available information in a broadband listening situation. Results from hearing-impaired listeners will also be presented.

8:30 PM: Effects of using loudness- versus audibility-based fitting procedures on success with digital hearing aids

David Fabry
Mayo Clinic

Much debate has been focused recently on the benefits of using loudness growth measures for clinical selection and evaluation of hearing aids. One limitation, however, has been that most conventional and digitally-programmable analog hearing aids do not permit compression thresholds and ratios to be adjusted by the clinician to permit a close "match" to fitting goals established by various procedures. Another issue has been that the ability for frequency-response shaping has been rather limited with these devices, particularly for individuals with precipitous hearing losses. The re-introduction of wearable digital hearing aids offers greater flexibility for evaluating differences between hearing aid fitting strategies in a clinical setting. This talk will focus on ongoing experiments with several digital hearing aids, comparing
loudness-based and audibility-based fitting models. Preliminary data suggest that word recognition is preserved as well or better with a maximum audibility model.

9:10 PM: Nonsense syllable judgment task for comparison of frequency/gain characteristics
Christine M. Rankovic
Northeastern University

An intelligibility judgment task comprising connected nonsense syllables is under development. Concatenated consonant-vowel-consonant syllables are presented along with a visual display on a computer monitor that identifies each syllable to the listener. The listener is asked to estimate the percent-correct consonant score. This test allows for rapid assessment of a broad range of listening conditions that vary along certain parameters of interest, such as, for example, amount of high-frequency emphasis or overall gain. Responses from the listener will be useful for identifying conditions that may be of interest for further investigation, as well as for eliminating conditions that are not of particular interest. Moreover, the powerful framework offered by articulation theory is still applicable: Articulation theory is not directly applicable when speech testing is conducted with contextually-loaded materials such as words or sentences. An evaluation of the application of Fletcher's (1953) version of articulation theory to hearing aid research will be presented. A comparison of intelligibility judgments with scores obtained by standard scoring procedures for the same nonsense syllable test will also be included. [Work supported by NIH].
Thursday Morning

SESSION 6: MODELING THE DAMAGED EAR: ITS RELEVANCE TO HEARING AID DESIGN, SELECTION, AND FITTING

Moderator: Brian Walden

9:00 AM: Implications of modeling sensorineural loss to hearing instrument design and fitting strategies

Herbert Bächler
Phonak, Switzerland

Recent research has provided increasing evidence that restoring loudness perception might improve the performance on other psychoacoustic tasks, such as speech perception.

Auditory processing for normal hearing and the perceptual consequences of cochlear damage will be discussed. Those with sensorineural hearing impairment experience elevated hearing thresholds, steeper growth of loudness, reduced frequency selectivity and reduced loudness summation. These problems seldom occur separately and thus reflect a common underlying mechanism, which can be explained and modeled.

The loudness model of Zwicker has been modified recently in two different ways (Launer & Moore) to account for sensorineural hearing impairment. However, the difference between the models is not very significant when applied to hearing instrument design and fitting.

The implementation of a loudness model in a digital hearing instrument allows one to restore loudness of narrowband as well as broadband signals correctly, which is supported by clinical data.

Furthermore the loudness model has been applied to analyze the underlying assumptions of current fitting targets, such as NAL or restoring normal equal loudness contours.

9:40 AM: Phoneme recognition as a function of dynamic range in simulated recruitment with and without high frequency emphasis

Arthur Boothroyd
City University of New York

The psychoacoustic consequences of sensorineural hearing loss include, but are probably not limited to, elevated threshold, reduced dynamic range, abnormal relationship of loudness to intensity, reduced frequency resolution, and reduced temporal resolution. All may be frequency-dependent and the last two may be amplitude-dependent. These psychoacoustic consequences are believed to contribute to the two most serious difficulties experienced by the hearing-impaired person -- reduced ability to perceive speech, and increased susceptibility to the interfering effects of noise. It is, however, difficult, in a clinical population, to determine the relative importance of these psychoacoustic factors, or to delineate the exact influence of each one on speech perception in quiet and noise. There are several reasons for this difficulty. The factors do not occur in isolation. They probably interact. Their effects and their interactions are likely to be non-linear. And, in a clinical population, the relative magnitudes of the different factors, and their interactions, are likely to vary from
individual-to-individual. One solution to this problem is to simulate the psychoacoustic consequences of sensorineural damage by processing the acoustic speech signal and measuring the effects in normally hearing subjects. If the simulations are valid (and that is always a big [I]), then we have an experimental model in which the psychoacoustic factors can be studied both separately and in interaction. This paper describes a study of phoneme recognition under conditions of simulated limited dynamic range with recruitment. The amplitude envelope (integrated over 22 msec) was extracted from the speech signal. A threshold was set at (say) 10 dB below the peak amplitude and any portion of the signal falling below this threshold was silenced. The remaining 10 dB was expanded (linearly in the pressure domain) to cover the range from threshold to comfort. This was done separately in 3 frequency bands (0-900 Hz, 900-2500 Hz, and 2500-7000 Hz) and the bands were recombined before presentation to the subjects. Phoneme recognition was measured as a function of dB dynamic range both without and with pre-emphasis of the high frequencies in the speech signal (6 dB and 9 dB in the middle and upper bands, respectively). Important findings were:

1. Even with a 30 dB dynamic range, performance was measurably poorer than with no recruitment,
2. The performance/dynamic range function was almost identical to that found when threshold loss is simulated with masking,
3. High frequency emphasis gave dramatic benefits when the simulated dynamic range was below 15 dB.

These data support two conclusions. The first is that the principal problem for the recruiting ear is loss of audibility of portions of the signal -- not the nature of the loudness versus intensity function for those sounds that are audible. The second is that many of the speech perception difficulties of the hearing-impaired arise because of the inherent weakness of much of the high frequency information in the speech signal.

**10: 20 AM: Break**

**10:30 AM: Simulated loudness recruitment**

Martin Dahlquist and Arne Leijon  
KTH - Speech, Music and Hearing, Sweden

First we will review some of the theoretical principles underlying various implementations of simulated loudness recruitment and discuss possible problems and advantages with each method. There are two main signal-processing techniques to achieve frequency-dependent non-linear level expansion of the signal envelope. 1) Time-domain processing using filter bank analysis, and 2) Frequency-domain block-wise processing using windowed FFT analysis and overlap-add synthesis. These two approaches are equivalent in principle but may differ considerably in the practical implementations.

Our own implementation uses the filter-bank approach with non-linear envelope estimation and non-linear transformation of the envelope to simulate the abnormal loudness function. This implementation runs in real time and thus allows interactive audiovisual presentation. The signal processing runs on a DSP board which is programmed using a graphical object-oriented design platform.

The shape of the loudness growth function applied in the present implementation is currently being evaluated with unilaterally impaired persons. We compare the real impairment in the affected ear with a simulated impairment in the healthy ear. Preliminary results will be presented from binaural loudness balance tests, speech reception tests and subjective judgments of sound quality.
11:10 AM: A model of loudness perception applied to cochlear hearing loss

Brian C.J. Moore and Brian R. Glasberg
University of Cambridge, England

A model developed to account for loudness perception in normally hearing people (Moore and Glasberg, 1995) is extended to deal with cochlear hearing loss. The model has four stages: (1) A fixed filter to account for transmission through the outer and middle ear; (2) Calculation of an excitation pattern; (3) Transformation of the excitation pattern to a specific loudness pattern; and (4) Calculation of the area under the specific loudness pattern, which is assumed to be proportional to perceived loudness.

It is assumed that the hearing loss (the elevation in absolute threshold) at each audiometric frequency can be partitioned into a loss due to damage to outer hair cells (OHCs) and a loss due to damage to inner hair cells (IHCs) and/or neurons. The former affects primarily the active mechanism that amplifies the basilar membrane (BM) response to weak sounds. It is modeled by increasing the "internal" threshold in the model, which results in a steeper growth of specific loudness with increasing excitation level; this mimics the reduction in or loss of the compressive nonlinearity in the input-output function of the BM, which is mainly associated with OHC damage. Loss of frequency selectivity, which results in broader excitation patterns, is also assumed to be directly related to the OHC loss. IHC damage is modeled by an attenuation of the calculated excitation level at each frequency. The model also allows for the possibility of complete loss of IHCs or functional neurons at certain places within the cochlea. The predictions of the model are compared with empirical data obtained in our laboratory using subjects with unilateral cochlear hearing loss who were required to make loudness matches between tones presented alternately to the two ears. The predictions are also compared to data in the literature on loudness matches between narrowband and broadband sounds. Generally, the predictions fit the empirical data reasonably well. The model is currently being extended to account for the perception of dynamically varying sounds.

Reference

Thursday Evening

SESSION 7: CLINICAL TRIALS TO ESTABLISH EFFICACY

Moderator: Christopher Turner

7:50 PM: Issues in the development of clinical trial protocols for evaluating hearing aid performance and benefit

Brian E. Walden
Walter Reed Army Medical Center

The current climate of healthcare reform, with its emphasis on managed care, has focused attention on the efficacy of hearing healthcare services, especially the development and validation of outcome measures. Although the need to document the efficacy of hearing aid evaluation and fitting procedures has existed for many years, recent actions of the Federal Trade Commission and the Food and Drug Administration provide a major impetus for documenting hearing aid performance and benefit. Government regulations require that manufacturers of medical devices, including hearing aids, substantiate marketing claims through clinical studies. In August, 1994, the FDA issued guidelines to hearing aid manufacturers for designing clinical trials to substantiate user-benefit claims. Some hearing scientists and clinical researchers are already engaged in manufacturer-sponsored clinical trials, and many others are likely to become involved in such research studies in the future. Although the FDA guidelines provide general guidance for conducting clinical trials to substantiate hearing aid performance and benefit, they do not dictate specific dependent measures or test conditions. Instead, it is left to individual manufacturers and/or researchers to select appropriate measures of hearing aid performance and to assess performance under appropriate test conditions. Agreement among manufacturers and researchers on a general-purpose clinical trial protocol would facilitate meaningful comparisons among user-benefit claims for competing products by dispensers and consumers. This presentation will discuss some basic issues in the development of clinical trial protocols to assess hearing aid performance and benefit, including the choice of test materials, control test conditions, and patient population. In addition, the rationale for and elements of a general-purpose protocol to assess hearing aid performance and benefit will be described. Data from a clinical trial of a multiband full dynamic range compression hearing aid system will be used to illustrate certain principles. A complete description of this clinical trial will be provided in an accompanying poster presentation.

This work received support from the ReSound Corporation, Redwood City, CA, under a Cooperative Research and Development Agreement with the Clinical Investigation Regulatory Office, U.S. Army Medical Command, Fort Sam Houston, TX; and from the Department of Clinical Investigation, Walter Reed Army Medical Center.

8:30 PM: Clinical trials with a digital filter-bank hearing aid

Thomas Lunner, Stig Arlinger and Johan Hellgren
Linkoping University, Sweden

Objective:

In a series of field tests with an experimental wearable binaural digital hearing aid, two hearing aid processors were compared. Both processors provided individual frequency
shaping via a seven band filter-bank with compression limiting in the high-frequency channels. They differed in the processing of the low-frequency channels, using dynamic range compression for one (DynEar) and compression limiting for the other (LinEar). In a pilot field test we found that LinEar/DynEar preference could be predicted from auditory dynamic range data. For the subjects who preferred the DynEar processor the mean dynamic range was broader for low frequencies and narrower for high frequencies, as compared to the LinEar preference subjects. This was tested in a main field test.

Design:

The main study included 26 experienced subjects with sloping or inverse sloping symmetrical sensorineural losses. The processors were compared in a one month long blind field test. A data logger function was included for objective recording of the total time used and how the volume controls were used. The preference was based on the time used for each processor and from subjective statements. To verify the preferences we tested the S/N threshold for speech and obtained sound quality ratings through a questionnaire. We also tested the S/N thresholds for the subjects' conventional (own) aids.

Results:

The preference was correctly predicted on 12 out of 15 new cases. S/N thresholds in noise were lower for the preferred fittings compared to the non-preferred fittings and to the subjects' own aids. In the questionnaire the preferred fittings were rated significantly higher concerning overall impression and clearness. The way the DynEar-preference subjects used the DynEar volume control was interpreted as indicating that upward spread of masking was an important factor.

Conclusions:

Preference for DynEar versus LinEar depends on the audiometric configuration of the listener. Both DynEar and LinEar present substantial benefit over conventional aids.

Keywords: Digital signal processing, filter-bank, multichannel, dynamic range compression, compression limiting, field test

9:10 PM: Implementation of the NIDCD/DVA Hearing Aid Clinical Trial

Vern Larson¹, Linda Barrett², William G. Henderson³, Lynn Huerta³, Jillyn Roxberg⁴, Lucille B. Beck⁵, Gene W. Bratt⁶, George B. Haskell⁷, Douglas Nofsinger⁸, Gerald Schuchman⁹, James A. Henry⁹, Stephen A. Fausti⁹ and Mia Rosenfield⁹

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The National Institute on Deafness and Other Communication Disorders and the Department of Veterans Affairs sponsored the development of a protocol to conduct a clinical trial of hearing aids. Acting on the recommendations of an advisory panel, the protocol was designed (a) to determine the efficacy of three hearing aid circuits (linear amplification with peak clipping, linear amplification with compression limiting, and single channel, wide-dynamic range compression) by examining the performance of adults with sensorineural hearing loss on three categories of outcome measures (speech intelligibility, quality, and self-rating of hearing aid benefit), and (b) to determine if there are characteristics that are useful for...
predicting performance with the three circuits (e.g., communicative need, degree and configuration of hearing loss, loudness discomfort level).

The trial is being conducted using a three-treatment, three-period crossover design. Patients are tested at baseline in the unaided condition and then use each of the three hearing aid circuits for ninety days (order randomized for each patient). At the end of each period, the patients are tested both in the aided and unaided condition. Neither the audiologist who administers the battery of outcome tests, nor the patient, is aware of the circuit under test. The focus of this presentation will be on the implementation of the clinical trial at eight audiology clinics located in VA Medical Centers.

A Power Macintosh (Model 7100) equipped with a board (Orange Micro, Inc.) that runs both Windows and DOS applications, controls a HI-PRO interface (Madsen Electronics), a three-channel audiometer peripheral (Virtual Inc., Model 322) and a hearing aid test system (Virtual Inc., Model 340). The output of each of the three channels is directed to a designated loudspeaker in sound-field.

Programmable hearing aids (Phonak, Dyna P2) are fitted, binaurally. They are programmed at the start of each ninety-day trial to function as one of the three circuit types. All electroacoustic characteristics, except those associated with the method of output limiting, remain constant throughout the trial. Frequency response is programmed to approximate an amplification target established using the NAL-R method. NOAH software (Hearing Instruments Manufacturers Software Association) is used to access hearing aid programming software. The user toggles between the programming software and the Model 340 real-ear measurement software to alternately fine-tune hearing aid characteristics and verify in-situ amplification targets.

For intelligibility testing, the materials from the Connected Speech Test (CST), developed, recorded and digitized by Cox and colleagues (Cox et al., 1988; Cox, 1994) and digitized recordings of the NU-6 test were installed on the computer’s hard disk. Software was written to control the presentation of stimuli through one channel and to present competing signals through the other two channels. CST materials are also accessed for use in a Quality Rating Test. In both intelligibility testing and quality judgment testing, a remote monitor alerts the subject to listen. Following each stimulus, a remote monitor prompts the patient to enter a number on a ten-point scale to rate one of three perceptive dimensions: overall quality, loudness and noisiness. In both tests, the computer records and stores patient responses for subsequent retrieval and analyses. Details of the protocol will be illustrated and discussed during this presentation. (Supported by the National Institute on Deafness and Other Communication Disorders and the Cooperative Studies Program, Medical Research Service, Department of Veterans Affairs.)
Friday Morning

SESSION 8: RECONCILING PHYSICAL PERFORMANCE MEASURES AND CLINICAL OUTCOMES

Moderator: Patricia Stelmachowicz

9:00 AM: Evaluation of nonlinear signal-processing hearing aids using speech and noise signals

Claus Elberling and Graham Naylor
Oticon Research Unit, Eriksholm, Denmark

With the appearance of nonlinear and complex signal processing hearing aids it is getting more and more difficult to present technical data which are relevant for the hearing aids in real life environmental conditions. Also some of the more complex proposed fitting methods (e.g. the IHAFF procedure - Cox, 1995) are based on hearing aid data which are normally not available or cannot be obtained with approved standardized methods.

The transfer properties of a hearing aid are normally obtained with the application of pure-tones, pure-tone sweeps or other 'stationary' signals. However, real life signals can rarely be described by stationary processes and especially speech, music and noise signals have quite other properties than those of stationary signals. The signal algorithms in complex signal processing hearing aids are not 'memory-less;' i.e. the output is not dependent only on the present input signal but also on the past input 'history' i.e. characteristics of previous segments of the input signal. Therefore in these hearing aids, the signal processing and the input signal interact. In order to get meaningful descriptions of the hearing aid the measurements should therefore be made with the proper input signals.

At the Lake Arrowhead Conference in 1994 we presented the first of a series of attempts to use speech signals to measure and evaluate nonlinear hearing aids. A specific method was developed which gave results which clearly demonstrated the 'stochastic' nature of the transfer properties in nonlinear hearing aids. In other words, the 'static' input-output function should not blindly be applied in the fitting process.

These and other results using input speech and noise sequences are acoustic in nature and therefore do not describe how the hearing aids may handle important phonemic elements. Boothroyd et al. (1994), described an approach which assesses 'phonemic' related properties of input speech signals. We have extended this idea by using speech signals from a 'labeled' speech data base to identify specific phonemic tokens and to evaluate how these tokens are processed embedded in running speech which is used as input to a nonlinear hearing aid. Further, by means of an auditory model simulating hearing loss the dimensions in the input-output domain could be changed from physical 'sound pressure level' to psychophysical 'loudness'.
9:40 AM: Estimating the optimum input-output characteristics of a self-adjusting hearing aid

Carl Ludvigsen
Widex Corporation

Fitting methods for nonlinear hearing aids typically aim at restoring normal loudness perception by means of amplifying input sounds within each frequency band according to a set of loudness functions for a particular user. Since threshold-based estimation of individual loudness perception has proven to be imprecise, most existing fitting strategies incorporate the measurement of individual loudness growth functions within relevant frequency bands. We examined various problematic aspects of this fitting rationale. First, we questioned the value of basing the fitting strategy on a precise restoration of a set of loudness functions obtained using stationary narrow band signals. Secondly, we could easily demonstrate that the dynamic properties of a hearing aid have a substantial influence on the loudness of its output signal. Therefore, we find it unlikely that a single set of stationary I/O curves could prove to be optimal for various hearing aid models incorporating different dynamic properties. Moreover, we uncovered a number of physical and methodological issues which must be considered before any fitting strategy can be precisely implemented. Finally, experience with linear amplification suggests that the optimum frequency response of the instrument should adapt to the various acoustic environments. We feel that a realistic fitting strategy should take these issues into account and we therefore formulated a time-efficient fitting strategy based on in-situ threshold data which allows for subsequent fine tuning as a means of compensating for individual variations in loudness perception.

10:20 AM: Break

10:30 AM: Algorithms for fitting and for signal processing with binaural hearing aids

Sigfrid Soli
House Ear Institute

Binaural directional hearing (BDH) is the ability of a listener to “tune out” noise from one direction and listen selectively to a signal from another direction. In noisy, non-reverberant environments BDH can improve speech intelligibility by 60% or more for normal hearing individuals when the signal and noise sources are spatially separated, relative to conditions without spatial separation of signal and noise. BDH ability is present in individuals with sensorineural hearing impairment, but may be reduced as a result of hearing impairment.

The use of hearing aid algorithms and fitting methods that attempt to preserve and optimize binaural hearing requires a means for determining the residual capacity for BDH of the hearing impaired individual. We have developed a headphone test system for this purpose. The system uses simulated head-related transfer functions plus amplification to maximize the Articulation Index as a means of eliminating the effects of audibility on BDH while preserving binaural cues. Reception Thresholds for Sentences (RTSs) are measured with this system in conditions with and without spatial separation of the speech and a spectrally-matched noise. We will report the results for 25 hearing impaired individuals tested with this system. These results reveal that BDH ability often fell within the range of normal-hearing listeners when audibility was controlled. The implications of these results for binaural hearing aid fittings will be discussed.
We will also describe algorithms and procedures that have been implemented with a wearable prototype digital hearing aid processor, and that may enable the hearing aid wearer to utilize fully their BDH ability. Binaural fitting of the processor is accomplished in two steps: Hearing Aid Equalization (HAE) and Hearing Loss Compensation (HLC). HAE equalizes the amplitude and phase insertion effects of the ear modules and maintains the binaural cues with the modules in place. The HAE digital filters for each ear are obtained from in situ probe tube measurements of aided and unaided test signals based on calculations performed by the PC. HLC for each ear is also achieved with digital filter and associated gain. The target response for each HLC filter is determined from measures of auditory thresholds and soundfield reference signals, and is based on an adaptation of Articulation Theory for hearing aids. The response of the HLC filter can be modified to prevent acoustic feedback from occurring. The HAE and HLC filters are combined to produce a single filter. A description of the processor, fitting algorithms, methods and accuracy of fittings, as determined from in situ probe tube measurements, and the hardware and software comprising the binaural hearing aid will be given. We will also report results from field trials with the portable processors.

11:10 AM: Issues in fitting nonlinear hearing aids
Chaslav Pavlovic
ReSound Corporation

Successful fitting of nonlinear hearing aids depends not only on the quality of the sound processing, but also on: (1) a precise characterization of the suprathreshold sound processing abilities of the hearing-impaired ear; (2) a precise electro-acoustic characterization of the hearing aid when it is inserted in the ear canal; and (3) a reasonable test efficiency with regard to the required test time.

While various psychoacoustical measures have been developed and used for describing suprathreshold problems (e.g. loudness scaling for recruitment), no methodology exists which relates these psychoacoustical descriptors to the speech signal processed by nonlinear devices and measured at the tympanic membrane. In this paper: (1) the above rationale will be considered; (2) some existing methodology will be analyzed; and (3) a new time-efficient protocol for fitting nonlinear devices will be elaborated on and analyzed in the light of patient data obtained at different test sites.

This technique (Real Ear Loudness Mapping or RELM) results in an auditory map of the patient's unaided dynamic range measured in real ear sound pressure level. Further, this auditory map is used as a prescriptive template against which the real ear aided response of multi-level speech signals are evaluated. The RELM measurements can then be used to fit, fine-tune, or validate the appropriateness of the hearing device.
Poster Abstracts

The 24 posters have been divided into two groups of equal size. The posters in the first group can be displayed on the walls of the second floor of the Arrowhead Conference Center during the first half of the conference, Monday evening through Wednesday noon. The first group of posters should be taken down Wednesday after lunch. The second group posters can be displayed from Wednesday afternoon through Friday noon.

POSTER GROUP A: Monday evening–Wednesday noon

The perception of noise in hearing aids by normal-hearing listeners

Jeremy Agnew and Michael Block
Starkey Laboratories, Inc.

Internal circuit noise in hearing aids is distracting to the listener and, if loud enough, may interfere with intelligibility, either by direct masking of weak components of speech or through the generation of undesired intermodulation products. The characteristics of noise may be measured objectively; however, wearers often differ in their subjective reporting of the perceived pitch and loudness of hearing aid noise. This poster reports the results of the pitch and amplitude matching of four normal-hearing subjects to the noise generated by a series of test hearing aids. Results showed that the subjects primarily matched the perceived pitch of the noise to the frequency of their most sensitive hearing, and approximately matched the perceived level of the noise to the total SPL noise level as measured by a sound level meter.

Implementation of adaptive loudness scaling procedure in compression parameter setting of a 3-band WDRC hearing aid system

Christopher Schweitzer, HEAR 4U International
Ole Dyrlund, GN Danavox

An adaptive loudness scaling procedure, based on work by Kiessling et al (1993, 1995), was studied in conjunction with the fitting of a 3-band hearing aid with adjustable low threshold compression. Ten sensorineural hearing-impaired subjects were fit bilaterally with Danavox Danasound 163 hearing aids using the ‘ScalAdapt’ procedure. This method utilizes sound field presentations of narrow band noises at levels regulated by a control microphone hung on the pinna. Using a 7-category scale, the subjects manipulated aid parameters by registering loudness responses on a handheld control device. This interactive, adaptive procedure is done for sound field levels corresponding to ‘Soft’ and ‘Loud’ responses for normal hearing subjects at frequencies centered within the 3 bands of the hearing aid frequency response. Adjustments of the compression parameters are accomplished automatically according to user loudness perceptions in a heuristic solution approach that will be described in greater detail in this poster. Findings of particular interest will be the number of adjustments required per measurement needed to finalize a parameter setting. The test/re-test reliability properties of
this method on untrained listeners will be also discussed. Expanded work on a larger number of subjects is in preparation.

**Spectral-temporal factors and the real ear compression ratio**

Todd Fortune
Argosy Electronics

An investigation was conducted to determine the effects of stimulus characteristics and venting diameter on the real ear compression ratio (CR). Real ear input/output (I/O) functions generated in response to syllables and speech noise revealed that the real ear CR is often less than the nominal value measured on the coupler, but only for CRs greater than 2:1. Pooled across vents, CRs were found to be associated with the duration and crest factor of individual phonemes, and the interval of time between successive phonemes. Highest CRs were associated with long-duration, low crest factor phonemes, while lowest CRs were associated with phonemes of short duration and high crest factor. Results were also influenced by an adaptive release time, which may have affected the compression of vowels and consonants differently.

**Perceived lateral position of narrow band noise in hearing-impaired and normal-hearing listeners under conditions of equal sensation level and sound pressure level**

Helen J. Simon and Inna Aleksandrovsky
Smith-Kettlewell Eye Research Institute

The current study is part of a project designed to increase the knowledge base regarding localization and binaural hearing in hearing-impaired individuals. The purpose of this paper is to report the results of a study that systematically investigated the relation between lateralization performance in subjects with normal hearing and sensorineural hearing loss (SNHL). Narrow-band-noise (NBN) signals with interaural intensity differences (IID) were presented with equal sensation level (EqSL) or equal sound pressure level (EqSPL) at the two ears. Using a graphic pointer paradigm, the perceived lateral position of the NBN at 11 center frequencies was studied by maintaining consistent intensity levels for EqSL and EqSPL within the same subject. The IIWs of the stimuli were varied randomly in 4-dB steps over a range of ±20 dB at each of 11 frequencies. The subjects indicated where they heard the stimulus by pointing a mouse-controlled cursor to the position on a schematic of the front of a head depicted on the computer screen. The results indicated that for EqSL (UnEqSPL), lateral position is essentially linearly dependent on the degree of asymmetry in asymmetric normal-hearing and hearing-impaired listeners. Equalizing by SPL showed no such dependency. EqSPL also produced more accurate lateralization performance than did EqSL. These results will be discussed in terms of the fitting of binaural hearing aids.

**Robust processing gains provided by a calibrated array of microphones**

Michael W. Hoffman, Xiao Feng Lu and Zhao Li
University of Nebraska-Lincoln

Previous simulation and calibration work has suggested that modest but significant and robust processing gains are available for arrays of conventional omnidirectional hearing aid microphones. Processing gain results (reported as changes in intelligibility averaged gain) are
presented for a variety of simple 3 microphone arrays. In addition, robust adaptive processing is demonstrated under conditions of missteer for an actual array of microphones. There are a number of interesting results. The robust processing gains are shown to be relatively insensitive to the number of taps. In addition, for this small aperture array (8 cm), adaptive processing significantly outperforms the fixed processing. Little gain is provided at lower frequencies – this places a limit on the achievable intelligibility averaged gain. Finally, intelligibility averaged gain is approximately constant for different input signal filtering, whereas change in the broadband signal to noise ratio is quite varied.

*This work has been supported by the Whitaker Foundation and by the National Science Foundation (IRI-9405286).

Excitation-pattern model for cochlear hearing losses applied to loudness and level discrimination

Mary Florentine and Soren Buus
Northeastern University

The main contention of this presentation is that excitation patterns provide a general framework for quantitative modeling of intensity perception in cochlear hearing losses. The loudness of a sound can be estimated from excitation patterns based on masking patterns when listeners’ reduced frequency selectivity and recruitment are taken into account [Florentine & Zwicker, Hear. Res. 1, 121-132 (1979); Florentine, Buus, & Hellman, Conf. On Modeling Sensorineural Hearing Loss, Boys Town Institute (1995)]. Likewise, the ability to discriminate level differences can be predicted when the listeners’ audiometric configuration and frequency selectivity are taken into account [Florentine & Buus, Proc. Biennial Hearing-Aid Research and Development Conf., NIDCD/VA, 1, 8 (1995); Florentine & Buus, J. Acoust. Soc. Am. 70, 1646-1654 (1981)]. To test the model, predictions were made for published data on growth of loudness in listeners with high-frequency losses and for data from our laboratories on level discrimination in listeners with various configurations of hearing loss. Model predictions agree well with the data. Applications to hearing-aid fitting will be discussed. [Supported by NIH/NIDCD]

Objective and subjective measures of hearing aid sound quality

Vijay Parsa, Donald G. Jamieson, Eva Chiu, Leonard Cornelisse and R. Wong
University of Western Ontario

Conventional measures of hearing aid distortion using pure tone signals (THD, IMD) correlate poorly with listeners’ subjective assessments of the severity of distortion. This is a particular concern because the perceived quality of hearing aid processed speech is an important determinant of listener acceptance of and satisfaction with a hearing aid. The use of broadband stimuli which have certain speech-like characteristics, such as an equivalent long-term average speech spectrum and crest factor may offer some advantages over measures based on pure tone signals, but such broadband stimuli still lack the important dynamic characteristics of speech. To address these concerns, we are exploring several objective measures of distortion, derived by testing hearing aids using real speech signals. These objective measures are being evaluated in relation to an extensive corpus of subjective quality data collected using both rating scale and paired comparison data, with a sample of 30 hearing aid distortion conditions, with both normal and hearing-impaired subjects. Correlations of electroacoustic measures and behavioral data demonstrate that certain speech-based objective measures may help to predict listeners’ sensitivity to various types of hearing aid distortion.
Differences between real ear aided responses of a hearing aid using its omnidirectional microphone mode and its directional microphone mode: Loudspeaker azimuth, reference microphone location, and signal type effects

Robert F. Zeliski and Robert J. Nozza
University of Georgia

The effects of two loudspeaker azimuths, two reference microphone locations, and two signal types on the differences in real ear aided responses for a hearing aid using its omnidirectional microphone mode and the same hearing aid using its multi-microphone directional mode were assessed. Results showed the over-the-ear reference microphone location provided more stable results than the cheek location, and the 45° loudspeaker azimuth produced smaller mean differences by microphone type than the 0° azimuth when using the complex noise stimulus. The results indicate that probe microphone measures of directional microphone hearing aids do not require unique protocols.

Predicting noise-masked thresholds of plosive bursts

Jim Hant, Brian Strope and Abeer Alwan
University of California, Los Angeles

The goal of this work is to develop an auditory model that can predict the masking of plosives in the presence of background noise. Since plosive bursts are brief, are generated by a noise source, and have varying spectral shapes, the modeling approach must account for duration, signal type (noise versus tone), center frequency, and signal bandwidth. In the first part of the study, we report on a series of psychoacoustic experiments that are aimed at quantifying the relationship between masked thresholds of noise signals and the signals’ center frequency, duration and bandwidth. We examined center frequencies ranging between .4-4 kHz, durations between 10-300 ms, and bandwidths between 1-8 critical bands. Signals were presented to four subjects within a flat noise masker with a spectrum level of 36 dB SPL/Hz. Adaptive 2AFC procedures were used to determine masked thresholds. Based on the experimental results, we derive a model of frequency- and duration-dependent auditory filters. The model assumes that roex auditory filters reach steady-state shapes when the signal duration is 300 ms or longer (Patterson et al., 1982). At shorter durations, both the effective filter bandwidth and the signal to noise ratio at threshold are frequency- and duration-dependent. We then conducted a set of perceptual experiments using synthetic and natural plosive bursts, in both front and back vocalic contexts and with varying durations. The noise masker was perceptually flat. Our proposed model predicted well listeners’ masked thresholds of these bursts in noise.

A description of our experimental results and the derived model’s parameters will be presented. [Work supported by NIH-NIDCD Grant No. 1 R29 DC 02033-01A1 and the Whitaker Foundation].
Measuring the dynamic acousto-electric path transfer function in hearing aids
Rod Speece, Marcio Siqueira and Abeer Alwan
University of California, Los Angeles
Sigfrid D. Soli and Shawn Gao
House Ear Institute

Acoustic feedback is a problem in hearing aids that contain a substantial amount of gain, hearing aids that are used in conjunction with vented or open molds, and in-the-ear hearing aids (Neuman, 1989). Acoustic feedback is both annoying and reduces the maximum usable gain of hearing aid devices. Thus, it is highly desirable to maximally attenuate the acoustic feedback signal to improve the performance of hearing aids. The acoustic path transfer function can vary significantly depending on the acoustic environment. Hence, effective acoustic feedback cancellers must be adaptive. We have measured the dynamic acousto-electric path transfer function (AEPTF) in a variety of acoustic environments and conditions for KEMAR and for two human subjects fitted with special in-the-ear (ITE) hearing aid modules in which the forward signal path through the hearing aid was interrupted by disconnection of the microphone and receiver. Wideband noise was delivered to the receiver and recorded on one channel of a DAT. The output of the ITE microphone was recorded simultaneously on the other channel. Recordings were made during hand movements near the ear simulating hearing aid adjustments, and during jaw movements simulating chewing.

The Steiglitz-McBride algorithm, a pole-zero estimator, was used to estimate the AEPTF, and the estimate was updated every 20 ms. Our analyses show that the KEMAR and human AEPTF measurements differ by about 8 dB for the same acoustic conditions. In addition, jaw and hand movements affect the AEPTF by as much as 18 dB, especially at frequencies above 1 kHz. These changes typically occur over .5 sec intervals. We will report detailed analyses of the dynamic AEPTF. These analyses are also being used as reference data for simulation and evaluation of adaptive feedback cancellers [Work supported by NIH].

Classroom experiments with FM amplification systems
Arthur Boothroyd1,2 and Frank Iglehart1,2,3
1Lexington Center, 2City University of New York, 3Clarke School for the Deaf

Two issues were addressed:

1. the implications of shifting from body-worn to behind-the-ear FM systems, and
2. the implications of shifting from an “equal gain” to an “equal output” criterion when adjusting relative acoustic gains in the FM and local microphone channels.

Thirteen orally-trained teenagers, with prelingual severe or profound hearing loss, were presented with CVC words, spoken lie at a distance of 10 feet, and asked to write down what they heard. Subjects listened with body-worn (Phonic Ear 471) and behind-the-ear (Phonic Ear FreEar) hearing aid/FM receivers, in noise (5dB and 20dB s/n ratio at the student’s ear and the talker’s chest, respectively) and in quiet, with and without the FM microphone turned on (i.e., a 3-factor design with two levels for each factor). Analysis of phoneme recognition scores showed:

1. a marked group FM advantage, in quiet and noise, for both systems
2. individual differences of FM benefit (tending to favor Ss with better hearing)
3. small but significant superiority of the body-worn system with the FM turned on
4. equal noise penalty (in percentage points) with and without the FM.

Finding 3 was expected and is believed to reflect the higher SSPL of the body-worn system. Finding 4 was unexpected and is believed to reflect the presence of amplitude compression in the microphone transmitter. In a subsidiary experiment, the gain via the FM microphone was reduced by 15 dB to meet the “equal output” criterion recommended by ASHA (1994). Predictably, the FM benefit was eliminated. These findings highlight the need for careful selection and adjustment of the many interacting parameters involved in successful use of FM amplification. Specific implications are:

1. peak clipping should be available as an alternative to compression-limiting for the profoundly deaf (in the interests of higher maximum rms output),
2. compression-limiting in the FM transmitter should be used only to limit unusually loud speech,
3. attempts to increase gain via the student microphone, relative to that via the FM microphone, must be accompanied by either automatic gain control or a microphone-priority circuit in order to preserve the FM benefit.

(Research funded by NIDRR)

**Response time and word recognition scores in a forced-choice and monitoring task**

Carol Mackersie, Arlene Neuman and Harry Levitt
City University of New York

This study explored the feasibility of incorporating response time measurements into tests of word recognition as a means of improving test sensitivity. The Modified Rhyme Test was administered to 12 hearing adults using both the traditional forced-choice format and a word monitoring task. The monitoring task required subjects to listen to series of words and push a button when they heard the target word that appeared on the monitor. Both response times and word recognition scores were measured for six signal-to-noise ratios (-3, 0, +3, +6, +9, +12 dB). Word recognition scores on the forced-choice task were approximately 10 percentage points better than on the monitoring task. Performance-intensity functions were parallel. Multiple regression analyses were performed for each subject to examine how well percent correct scores, response times and the combination of the two measures predicted signal-to-noise condition. Percent correct scores were better predictors than response time for both tasks. Predictions improved when percent correct scores were supplemented with response time data for the monitoring task, but not the forced-choice task. Results suggest that word recognition test sensitivity can be improved by combining percent correct results with response time measures obtained with a monitoring task.
A wearable real time digital hearing aid incorporating noise suppression
Neeraj Magotra, Sudheer Siravara and Bill Swartz
University of New Mexico

This paper describes the development of a binaural wearable digital hearing aid. The device, referred to as the Customized Universal Digital Listening System (CUDLS), uses a TMS320C3X floating-point digital signal processing chip. It provides for frequency shaping from dc to 16000 Hz using variable number (1 to 50, typically 14) of Finite Impulse Response (FIR) filters. Each one of these FIR filters has 50 - 200 taps. These filters allow for precision frequency shaping. Linear phase is maintained across the entire frequency bandwidth, and greater than 80 dB band isolation is achievable. CUDLS also incorporates noise suppression as an integral part of the hearing aid. The current noise suppression algorithm is the Real-time Adaptive Correlation Enhancer (RACE). RACE is essentially an open-loop adaptive FIR filter. RACE estimates a short-term autocorrelation function of the input signal and uses it to update the adaptive FIR coefficients. The algorithm works with dual (right ear and left ear) single-input single-output channels. In addition to frequency shaping and noise suppression, CUDLS also permits real time implementation of multiband amplitude compression, frequency dependent interaural time delay algorithms and also other applications in dealing with autism and attention deficit disorders. We have developed our own customized hardware and software to implement CUDLS in real time with zero processing delay. The hardware consists of a DSP card that installs on a PC to enable the audiologist to program the hearing aid and a portable battery operated unit approximately half the size of a walkman. The software package designed to operate CUDLS in the PC environment makes it easy for the audiologist to modify system features. Once the audiologist is satisfied with the patient's performance on the PC based system the portable unit can be programmed for the individual being tested.

We have recently been awarded a patent for CUDLS.

A clinical trial of full dynamic range compression amplification
B.E. Walden, R.K. Surr and M.T. Cord
Walter Reed Army Medical Center

This poster presents the results of a clinical trial of the ReSound BT2 Personal Hearing System (PHS), following a general-purpose protocol described in another presentation at this conference (Walden, "Issues in the development of clinical trial protocols for evaluating hearing aid performance and benefit"). The BT2 PHS consists of binaural, behind-the-ear, hearing instruments that incorporate multiband full dynamic range compression amplification. The speech recognition ability of forty adults with sensorineural hearing loss was evaluated unaided and aided with the BT2PHS. All subjects had used binaural linear hearing aids for at least one year prior to their enrollment in the protocol. The Continuous Speech Test (CST) was administered under the following laboratory conditions; (a) at 50dBA with +10 speech-to-babble ratio (S/B), (b) at 60 dBA with 0.78 sec. reverberation, (c) at 60 dBA with +5 S/B, and (d) at 70dBA with +2 S/B. In addition, the Profile of Hearing Aid Benefit (PHAB) was administered to the subjects. The CST scores revealed that the subjects obtained significantly better speech recognition when wearing the BT2 PHS in comparison to unaided performance under all four laboratory test conditions. Similarly, the subjects reported significantly better
performance with the BT2 PHS in comparison to unaided performance on the PHAB. Despite the significant improvement in speech recognition ability provided by the B2 PHS, when compared to the CST scores of normal-hearing listeners at the same presentation levels and S/B’s, hearing-impaired subjects performed significantly worse. However, the hearing-impaired subjects reported performance that approximated normal on several scales and subscales of the PHAB. Finally, hearing-impaired subjects reported significantly better performance with the BT2 PHS compared to their own linear hearing aids on each of the scales and subscales of the PHAB.

Programmable hearing aid fitting procedures in the context of a clinical trial
R.K. Surr, M.T. Cord and B.E. Walden
Walter Reed Army Medical Center

It is essential in a hearing aid clinical trial that the procedures for fitting the instruments are sufficiently explicit that different clinicians would provide similar fittings for a given patient. A companion poster (Walden, Surr, and Cord, "A clinical trial of full dynamic range compression amplification") reports the results of a clinical trial of the ReSound BT2 Personal Hearing System (PHS). This poster presents the hearing aid fitting goals, guidelines, and procedures used in that study, and illustrates the complexities of standardizing fitting procedures for technologically advanced hearing instruments. The fittings were accomplished in three clinic visits. During Visit 1, audiometric thresholds and loudness growth of 1/2-octave bands of noise centered at .5, 1.2, and 4 kHz (LOGB test) were obtained. These data were used to derive the initial programming parameters for the BT2 instruments. The initial settings were then adjusted according to specific gain criteria relative to the subject's audiometric thresholds. During Visit 2, the instruments were fit to the patient and the fitting parameters readjusted to achieve appropriate aided loudness perception for speech. During Visit 3, which followed a 2-week familiarization period, further adjustments to the fitting parameters were made on the basis of subjective experiences with the BT2 PHS in daily use. The average change in the gain parameters between the initial and the final programming of the instruments was less than 4 dB. Similarly, there were minimal average changes in the cross-over frequency and in the compression ratio for either the low- or the high-frequency band. Examination of individual fitting data indicated that the gain parameters were increased for most subjects. This increase in gain was disproportionately higher for the subjects who elected to purchase the BT2 PHS at the conclusion of the clinical trial compared to those who chose to continue to wear their own linear instruments.

A new portable digital hearing aid incorporating time and intensity
Helen J. Simon¹, Neeraj Magotra², Brennan McBride¹
¹Smith-Kettlewell Eye Research Institute, ²University of New Mexico

Under the auspices of the Smith-Kettlewell Eye Research Institute (SKERI) and its Hearing Research Unit, the Electrical and Computer Engineering Department at the University of New Mexico (ECED) has specifically designed a prototype digital, multi-band binaural hearing aid for manipulating frequency dependent time delay. This hearing aid, based on Texas Instruments' floating point TMS320C3X digital signal processing (DSP) chip is referred to as the Customized Universal Digital Listening System (CUDLS). The customized hardware and software is designed to permit real time implementation of frequency shaping, background noise reduction, amplitude compression, as well as the frequency dependent interaural time delay algorithms to be used for this particular application. The hardware consists of a DSP
Listener performance on the telephone: A comparison of conventional hearing aids, hearing aids with programmable telecoils, and amplified handsets

Stephanie A. Davidson
The Ohio State University

When listeners with hearing loss complain of difficulty understanding speech on the telephone, hearing aids equipped with a telecoil are often suggested. Unfortunately, many individuals remain dissatisfied with their ability to communicate on the phone. Although many factors may contribute to the hearing aid user’s dissatisfaction, one explanation may be the telecoil response. It has been well documented that the electroacoustic characteristics of a hearing aid can change substantially when telecoil input rather than microphone input is utilized. In many cases the changes that occur with telecoil input are undesirable (e.g., a more restricted frequency response, lower gain and output levels, or prominent peaks in the response), and it may be that these changes are contributing to the hearing aid users’ dissatisfaction. To determine if programmable telecoil responses could be used to improve listener performance on the telephone, eighteen experienced hearing aid users were tested with their own hearing aids. Two different telecoil responses were obtained using a commercially available programmable hearing aid (a response programmed using the manufacturer’s suggested algorithm for telephone listening, and a response programmed to match the NAL-R target).

Listener performance using an amplified handset was also evaluated. Speech understanding ability (CUNY Nonsense Syllable Test, R-SPIN, and the Telecoil Evaluation Procedure) was best using the amplified handset and poorest using an unmodified version of the manufacturer’s suggested algorithm. Subjective ratings of the loudness, intelligibility, and quality of each amplification system were highly variable across subjects and may be related to the configuration of the hearing loss.
Supra-threshold directional hearing with single talker maskers

Michael J. Nilsson, Sigfrid D. Soli and Andrew J. Vermiglio
House Ear Institute

Reception thresholds for sentences (RTS) were measured in 14 normal hearing listeners in three supra-threshold conditions (noise at 0°, 90°, or 270°) with three alternative 65 dB(A) maskers: spectrally matched noise, forward single talker speech, and backward single talker speech. The single talker speech maskers include envelope cues not found in the spectrally matched noise, while the forward speech contains linguistic information not found in the backward speech. The largest masker effect was improved directional hearing (performance measured when the target and masker are spatially separated) with either single talker masker versus the spectrally matched noise (F(2,81)=60.07, p<.01; RTS = -9.8 dB S/N, -15.9 dB S/N, and -16.1 dB S/N for the noise, forward speech, and backward speech maskers, respectively). Linguistic content (forward versus backward speech) was found to have a detrimental effect when the speech and masker were presented at the same azimuth (t(25)=2.4, p<.01; RTS = -2.1 dB S/N and -4.5 dB S/N for the forward and backward speech maskers, respectively), but not when presented from different azimuths (t(53)=0.3, p=.78; RTS = -15.9 dB S/N and -16.1 dB S/N for the forward and backward speech maskers, respectively). The large directional hearing advantage found with the single talker maskers can only be attributed to acoustic properties of the masker since there was no difference between the forward and backward speech conditions. The data support the hypothesis that the linguistic processing of speech is not obligatory when the speech is used as a spatially separated masker.

A portable hearing aid based on narrow-band-loudness compensation and its clinical evaluation

Y. Suzuki, K. Ozawa, T. Sone
Research Institute Of Electrical Communication

F. Asano
Electro-Technical Laboratories

M. Ohashi, Hidaka, S. Takahashi, T. Kawase and T. Takasaka
Tohoku University School of Medicine

Persons with sensorineural hearing loss have narrow audible range causing loudness recruitment. This means that the loudness function of the persons is deformed. Moreover, this deformation is usually a function of frequency. Therefore, a full digital hearing aid with frequency-dependent compression has been studied. Compression characteristic is so determined that the output of the aid becomes equally loud to that perceived by normal listeners for each 1/1 octave band from 250 Hz to 4 kHz. The compression characteristics for the five frequency bands are defined by Loudness Compensation Functions (LCF), which describe the relation between the loudness functions of an impaired listener and an average loudness function of normal listeners. The input signal is first AD-converted then short term frequency spectra are calculated with FFT. The spectra are band averaged for each 1/1 octave band and the band levels are input to the LCFs. The outputs of LCFs yield gains which make the output signal for each frequency band at this moment be perceived by the impaired listener as loudly as perceived by an average normal listener. The gains are interpolated over frequency and realized by a single digital filter; the frequency sampling digital filter. The
algorithm is called CLAIDHA (Compensating Loudness by Analyzing the Input, Digital Hearing Aid).

The newest model, the fourth model, had been completed in the beginning of 1995. This was approved to sell by the Ministry of Health and Public Welfare, Japan in July 1995, and has been sold by the name of Cleartone since September 1995. The aid uses Motorola DSP-56166 and can be kept operating for over 20 hours with four standard LR6 dry batteries. The size is 70 x 130 x 24 mm. It is quite important for a good fitting of the aid to measure loudness functions of a specific impaired listener effectively and correctly. Thus, a system to measure loudness functions by the category subdivision scaling method was built on PC. The measured loudness functions are converted to LCF’s and transferred to the aid using a serial interface. The aid has been fit to more than one hundred impaired listeners during the last six months at a clinic which was established for the fitting of the aid in Sendai, Japan. The performance has been evaluated by a Japanese mono-syllabic speech test and a questionnaire survey. The results show a relative improvement of speech intelligibility compared with those for the linear system. The questionnaire survey shows that some 60% of the listeners are quite satisfied with the performance of the aid. Moreover, a preliminary experiment to examine benefits of binaural use of the aid is being conducted. Some results of the experiment will be presented at the symposium.

Field evaluations of a DSP processed algorithm and CIC hearing aids

Chiquita Mayhugh and Jeremy Agnew
Starkey Laboratories, Inc.

The performance of subjects using a new DSP hearing aid algorithm developed by the House Ear Institute Hearing Aid Research Department was compared to their performance wearing binaural CIC hearing aids. The DSP algorithm in a full concha ITE electrically corrects for both the loss of unaided magnitude and phase effects. The position of the microphone of the CIC and the exposed pinna ridges are thought to maintain phase effects acoustically. Ten subjects participated in clinical evaluations which included measures of sound quality, HINT, localization, real ear measurements and electroacoustic analysis of both systems. The results show that performance with the DSP algorithm is equivalent to performance with CIC hearing aids in terms of HINT, SPIN and localization. Subjective responses show preference for the DSP algorithm for sound quality and localization.

Detection of complex modulation

Curt Southworth
University of California, Irvine

A modulation detection procedure was used to estimate thresholds for complex sinusoidal amplitude modulation of broadband noise. Modulators with rates from 4 to 1024 Hz were individually paired with a 16-Hz modulator, and yoked in amplitude. Detection thresholds for the 16-Hz pairings as a function of second modulator rate reveal small performance enhancements in the neighborhood of 16 Hz for one listener, and at all rates for a second listener. However, the improvements lie below predictions based on the sum of modulator amplitudes, and overall modulation depth does not appear important for detection. A comparison of additive and multiplicative modulator combination methods reveals lower thresholds in multiplicative conditions for most of the modulator pairings examined. The finding of an advantage for multiplicative combination is inconsistent with a “listening in the valleys” strategy for complex modulation detection. Weights for modulation components
estimated by the methods of conditional-on-a-single-stimulus (COSS) analysis show positive and negative interactions that may be correlated with results from selective adaptation studies. Interactions between modulation components represent departures from models that assume simple linear combination of modulators for auditory processing.

[Work supported by ONR.]

**High frequency audibility: Benefits for hearing-impaired listeners**

Cynthia A. Hogan and Christopher W. Turner
Syracuse University

The present study was a controlled investigation of the benefit of providing hearing-impaired listeners with various amounts of high-frequency speech information when it was presented at audible levels. The results of the study have implications for the fitting of hearing aids, since clinicians need to know the expected benefits of various amplification prescriptions. Previous research on this topic has not been conducted under well-controlled conditions of audibility. Five normal-hearing and nine high-frequency hearing-impaired listeners identified nonsense syllables presented at levels that insured audibility. The syllables were low-pass filtered at a number of cut-off frequencies and the benefit of increasing cut-off frequency, thereby increasing audible high-frequency information, were assessed in terms of phoneme identification. Articulation Index (AI) was calculated for each condition for each listener; this served as a means of quantifying audibility for each condition. An increase in AI indicated that the change in cut-off frequency resulted in increased audibility, i.e., that additional high-frequency information was audible to the listener. A new measure of how well hearing-impaired listeners used information within specific frequency bands called “efficiency” was devised. This measure compared the benefit of increasing speech audibility to a hearing-impaired listener to that observed in normal-hearing listeners.

Results showed that the normal-hearing listeners were able to improve their speech recognition as high-frequency information was increased. Most hearing-impaired listeners demonstrated an improvement in speech recognition as audible high-frequency information was provided. The listeners with mild high-frequency hearing loss performed in a manner similar to the normal-hearing listeners. While listeners with moderate to severe hearing impairment did improve speech recognition performance to a certain extent with increases in audibility, they did not receive the same benefit as did normals. In some cases for severely impaired listeners, increasing the audibility of high-frequency speech information resulted in decreases in speech recognition. There was a clear pattern in the results suggesting that as the degree of hearing loss increased beyond 55 dB HL, the efficacy of providing additional audibility was diminished, especially when this degree of hearing loss was present at frequencies of 4000 Hz and above.

**Temporal properties of background interference affect choice of compression ratio**

Michael A. Stone, Emma Gudgin, Brian C. J. Moore and Magda Wojtczak
University of Cambridge, England

Extensive testing of dynamic range compression circuits in hearing aids has shown that the choice of compression parameters is a compromise among several factors. These include the need to understand speech both in quiet and in background noise, the need to be aware of
environmental sounds, the need to prevent an acoustic signal from becoming uncomfortable and the need for 'hands-off' use of the aid, other than to make fine adjustments.

In laboratory-based speech-comprehension tests of aids, it has been common to use an interfering continuous noise with the same frequency spectrum as the long term average of the speech to be detected. In real life, interferences have temporal fluctuations. Recent literature reports a greater disparity in levels of performance between normal and hearing-impaired listeners when tested in modulated noise, compared to the disparity seen when using continuous noise.

Does a fast-acting compression system permit a hearing-impaired user to make use of the spectro-temporal gaps found in a modulated masker?

This poster describes the effect on the SRT of using either a full dynamic range compressor or a limiting-compressor when the interference is either a speech-shaped noise or a single competing talker. Three types of speech material were used: a male or a female speaking short sentences or a male speaking isolated CVC words. Paradoxical results have emerged consistently: compared to a full dynamic range compressor, compression-limiting is a better choice when temporal fluctuations are present (i.e. for a single interfering talker), but is worse when the interference lacks fluctuations. Compared to a linear aid, compression appears to offer small advantages in SRT when the masker is fluctuating; this benefit is reversed when the noise is continuous.

**A method for analyzing rating data**

Harry Levitt and Christopher Oden

City University of New York

Rating techniques have several useful advantages in hearing aid evaluation. These include speed and ease of administration as well as providing a direct assessment, from the user's perspective, of how well the hearing aid functions. The analysis of rating data, however, presents a problem in that the usual assumptions of linearity and additivity do not apply. A method for analyzing rating data is presented. The technique has many useful properties, including a transformation that converts rating data to an interval scale, thereby allowing for the use of conventional methods of statistical analysis. Only two assumptions are made: (i) the rating scale is ordinal, or (ii) test-retest variability is accounted for by random variations in the ratings about a fixed underlying distribution, i.e., repeated administrations of a test condition will yield a distribution of ratings that converge on the underlying distribution for that test condition.
# Conference Attendees

Names marked with an asterisk are recipients of student scholarships to attend the conference.

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<th>Name</th>
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<td>Starkey Laboratories, Inc.</td>
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<td>Jont Allen</td>
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<td>Mark Dobkin</td>
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*Issues in Advanced Hearing Aid Research*

*May 27-31, 1996*
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