ISSUES IN ADVANCED HEARING AID RESEARCH

FIFTH BIENNIAL CONFERENCE
MAY 25 - 29, 1998

UCLA Lake Arrowhead Conference Center
Lake Arrowhead, California
ISSUES IN ADVANCED HEARING AID RESEARCH

Sponsored by

House Ear Institute

&

Acoustical Society of America
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Welcome Letter

The House Ear Institute is pleased to present to you the fifth in a series of conferences held biennially to foster an open and creative exchange of information and ideas among researchers in the diverse field of hearing aid research. The content of the Conference Issues in Advanced Hearing Aid Research will embrace both basic and applied research, and will allow in-depth presentations and discussions in advanced hearing aid research.

Sigfrid D. Soli, Ph.D.
Director, Hearing Aid Research
House Ear Institute
ISSUES IN ADVANCED HEARING AID RESEARCH

PLANNING COMMITTEE

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DAILY SCHEDULE

MONDAY, MAY 25

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 26-28

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

FRIDAY, MAY 29

8:00 AM    Breakfast and checkout
9:00 AM    Morning session
12:00 PM    Adjournment (buses leave for airports with box lunches for passengers)
PROGRAM SUMMARY

MONDAY, MAY 25

SESSION ONE
7:50PM-10:00PM

EFFICACY OF AMPLIFICATION

Moderator: Christopher Turner

Ruth Bentler  Hearing aids through the ages
Mead Killion  Predicting “missing AI dots” from SNR loss; It all seems to hang together
Robyn Cox  Listener satisfaction with hearing aids

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TUESDAY, MAY 26

SESSION TWO
9:15AM-12:00NOON

THEORETICAL ASPECTS OF SPEECH PERCEPTION AND HEARING LOSS

Moderator: Jerry Agnew

Jont Allen  Information theory and the articulation index in human speech recognition
Bob Shannon  Speech recognition with degraded and shifted tonotopic cues: implications for hearing aids and cochlear implants
Gerald Studebaker  Intensity level effects on speech recognition
TUESDAY, MAY 26
SESSION THREE
7:50PM-10:00PM

MORE THEORETICAL ASPECTS OF SPEECH PERCEPTION AND HEARING LOSS

Moderator: Harvey Dillon

Tammo Houtgast  Modeling speech reception by the hearing impaired
Christopher Turner  Efficacy of high-frequency audibility in sensorineural hearing loss
Van Summers  Towards a behavioral metric of active-mechanism status in listeners with sensorineural hearing loss

WEDNESDAY, MAY 27
SESSION FOUR
9:00AM-12:00NOON

MODULATION AND COMPRESSION

Moderator: Roger Miller

Hans Verschuure  The concept of compression in relation to speech scores and effective modulation suppression
William Cole  Using modulated signals to test hearing aids
Rene van der Horst  Modulation-masking and filtering in a speech recognition task for hearing impaired and normal hearing subjects
Brian Moore  Comparison of different forms of compression using wearable digital hearing aids
WEDNESDAY, MAY 27

SESSION FIVE

7:50PM-10:00PM

NON-LINEAR HEARING AID TECHNOLOGY

Moderator: Chas Pavlovic

William Johnson  Compressing the dynamic range with minimum ill effect
Steve Armstrong  A new way of viewing the dynamic gain of WDRC circuits
Soren Westermann, Carl Ludvigsen  Technical and clinical assessment of non-linear hearing aids

THURSDAY, MAY 28

SESSION SIX

9:00AM-12:00NOON

ASSESSMENT OF HEARING TECHNOLOGY

Moderator: Claus Elberling

Graham Naylor  Ear level DSP hearing aids as research tools: How can their potential best be exploited?
Wouter Dreschler  Some problems in the objective evaluation of advanced signal processing in hearing aids
Pat Zurek  Laboratory assessment of advanced signal processing algorithms for hearing aids
Pat Stelmachowicz  The relationship between stimulus context, speech audibility and perception for children and adults
THURSDAY, MAY 28

SESSION SEVEN

7:50PM-10:00PM

ASSESSMENT OF NEW TECHNOLOGY

Moderator: Leslie Collins

Sig Soli  Measuring, preventing, and canceling acoustic feedback in hearing aids
Dave Preves  Some developments in directional hearing aids
Birger Kollmeier  Signal processing for hearing aids based on auditory models

FRIDAY, MAY 29

SESSION EIGHT

9:00AM-12:00NOON

MULTIPLE EARS, MULTIPLE MICROPHONES

Moderator: Arne Leijon

Dave Fabry  Laboratory and real-world evaluation of multiple microphones in hearing aids
Dianne Van Tasell  Binaural speech intelligibility in noise with and without hearing aids
Denis Byrne  Optimizing sound localization by people wearing hearing aids
Stewart Gatehouse  Localization, SNHL, and amplification

*Issues in Advanced Hearing Aid Research*  
May 25-29, 1998
SESSION ONE

EFFICACY OF AMPLIFICATION

Moderator: Christopher Turner

7:40 PM  INTRODUCTORY COMMENTS

7:50 PM  HEARING AIDS THROUGH THE AGES
Ruth Bentler
University of Iowa

Whether the setting is a clinic performing a hearing aid evaluation for a potential candidate, or an independent laboratory performing a FDA clinical trials investigation, some outcome measures of hearing aid performance are obtained. Based on these outcome measures, claims of performance or efficacy are made. Data will be presented from several laboratory and field investigations wherein outcome measures were obtained across a number of “high” and “low” tech circuitries. These circuits included compression systems that were phonemic, syllabic, and/or slow acting in nature. Regardless of the technology used, audibility of the signal was the best predictor of performance across the different measures used, once distortion and bandwidth were accounted for. With these typical outcome measures, the various signal processing strategies could not be differentiated as to their efficacy for the hearing impaired listener. A discussion of the limitations of the measures vs. the limitations of the processors currently available will be presented.

8:30 PM  PREDICTING “MISSING AI DOTS” FROM SNR LOSS: IT ALL SEEMS TO HANG TOGETHER
Mead Killion
Etymic Research

Several groups, notably Olsen and Tillman (1968), Plomp and his colleagues (1978 and subsequently), and Dirks and his colleagues (1982) have measured the signal-to-noise ratio (SNR) required by normal and hearing-impaired listeners for 50% correct scores. The resulting SNRs varied because of the different intelligibility in noise of Spondees, complete sentences, or CNC words, and because of the different characteristics of the noise maskers. The latter ranged from continuous speech-spectrum noise (SSN) to single-talker competition to several-talker babble, to so-many-talker babble that no individual voice could be
identified (almost a complete circle back to SSN). In 1993, Selda Fikret-Pasa and I chose four-talker babble as the appropriate interference to simulate the “cocktail party problem.” We later found it was possible to follow any one of five talkers (including the target) at 0 dB SNR, reflecting the common experience at such gatherings that it is possible to selectively attend to one of several speakers (Broadbent, 1958). With half credit for partially-correct words, the statistical characteristic of each 25-word subset of the resulting speech-in-noise (SIN) test (Killian & Villchur, 1933) is similar to a full-word scoring 25-word test (10% standard deviation for scores near 50%), and the SNR estimate (derived from two such scores) has a standard deviation of 0.7 dB. Selectively filtering out the low- or high-frequency speech and noise cues of the SIN test provides tests offering refined estimates of the proportion of low- and high-frequency speech cues that appear to be lost somewhere between the eardrum and the brain, even when all or most speech cues are audible to someone with sensorineural hearing loss. Attempts to correlate these “missing dots” results with N1 (ABR) amplitude and histological inner-hair-cell counts are under way. The application of these results to hearing aid design will be defended.

9:10 PM

LISTENER SATISFACTION WITH HEARING AIDS
Robyn Cox and Genevieve Alexander
University of Memphis

In the prevailing atmosphere of zeal to improve services and to demonstrate their efficacy, hearing aid practitioners have embraced the collection of self-assessment data with considerable enthusiasm. Service providers have found that self-reported outcome data yield valuable insights into the impact of impairment on everyday life and promote planning and execution of a rehabilitative strategy that effectively addresses the needs of the hearing-impaired person. Further such data can be used to document the merit of the treatment program and can point to areas that are meeting expectations as well as those that are in need of improvement.

Most of the standardized inventories proposed to date to measure amplification effectiveness have concentrated on quantifying Benefit by measuring reduction in communication disability. However, there is a substantial body of literature examining experienced hearing aid consumers which establishes that improved speech communication, while essential, is only one of several elements needed for a fully adequate hearing aid fitting. Satisfaction is the outcome variable that appears to encompass the full constellation of factors needed for a positive fitting result. Although the importance of satisfaction has been appreciated from a marketing point of view for many years, it has not received much attention from researchers. We propose that when the overall outcome of amplification provision from the patient’s point of view is the variable of interest, Satisfaction is perhaps more important than Benefit.

We have developed the Satisfaction with Amplification in Daily Life (SADL) Scale to address the need for a standardized, clinically-useful approach to
quantifying Satisfaction. The scale is composed of 15 items and generates four sub-scale scores and an overall score. This talk will describe the development and final form of the scale, and will present preliminary information about the relationships between scores and several demographic variables such as age, gender, type of clinic, hearing aid experience, etc. [Supported by the US Department of Veterans Affairs, RR&D Service]
SESSION TWO

THEORETICAL ASPECTS OF SPEECH PERCEPTION AND HEARING LOSS

Moderator: Jerry Agnew

9:15 AM INFORMATION THEORY AND THE ARTICULATION INDEX IN HUMAN SPEECH RECOGNITION
Jon E. Allen
AT&T Bell Laboratories

Humans have been attempting to understand their unique ability to communicate through the spoken word, probably since they first became aware of their unusual intelligence. We stand to gain tremendously from a scientific understanding of the basic principles behind human speech recognition (HSR). Unfortunately this "holy grail" problem has not yet cracked to research investigation. In part this is because of the complexity of language, and in part because of the complex transformation required to get from the acoustic waveform to a phonemic string.

This problem was studied exhaustively at Bell Labs between the years of 1918 and 1950 by Harvey Fletcher and his colleagues. The motivation for these studies was to quantify telephony speech to improve both intelligibility and preference. To do this he and his group studied the effects of filtering and noise on speech recognition accuracy for nonsense consonant-vowel-consonant (CVC) syllables, words, and sentences. Fletcher used the term \textit{articulation} as the probability of correct recognition for \textit{nonsense} speech sounds, and \textit{intelligibility} as the probability of correct recognition for words (sounds having meaning).

In 1919 Fletcher sought a nonlinear transformation of the articulation data for filtered and unfiltered speech defined to give an additive articulation density function $S(f)$ over frequency. The area under $S(f)$ he called the \textit{Articulation Index} (AI). Fletcher then went on to find relationships between the recognition errors for the nonsense speech sounds, words, and sentences, and showed that these probabilities may be predicted from his AI measure. This work has been reviewed, replicated, and extended by Boothroyd and by Bronkhorst {at et al}. Taken as a whole, these studies tell us a great deal about how humans process and recognize speech sounds.

This work lead the more accessible and widely publicized, but greatly simplified, version of the AI model by French and Steinberg. This model was then reduced to the approximate ANSI standardized AI method. The ANSI scheme, which was
not intended to give insight into the perception of speech, should not be confused with Fletcher and Galt's much earlier and extensive investigations.

I shall review this work, and descriptively (no math) blend it with a modern information-theoretic approach on how we understand speech.

10:20 AM BREAK

10:30 AM Speech recognition with degraded and shifted tonotopic cues: Implications for hearing aids and cochlear implants
Bob Shannon and Qian-Jie Fu, House Ear Institute
Angela Jensvold, University of Southern California

Recognition of spectrally degraded and frequency-compressed or expanded vowels was measured in both acoustic and electric hearing. Spectral resolution was reduced by extracting envelope cues from 4, 8, or 16 spectral bands and using those envelope waveforms to modulate 4, 8, or 16 noise bands. Spectral shifting was accomplished by altering the tonotopic location and of the noise bands relative to the analysis bands. This manipulation not only degraded the spectral information by limiting the number of bands, but also shifted the relative tonotopic representation of speech envelope information. Results from five normal-hearing subjects showed that vowel recognition was sensitive to both the number of bands and frequency shifting. The effect of a frequency shift did not interact with the number of bands, suggesting that spectral resolution and spectral shifting are orthogonal manipulations in terms of intelligibility. High vowel recognition scores were observed for as few as 4 bands. Regardless of the number of bands, no significant performance drop was observed for tonotopic shifts less than 3 mm, equivalent to frequency compression/expansion of up to 60%. Similar results were obtained from five cochlear implant listeners, when electrode locations were fixed and the spectral location of the analysis filters was shifted. Implant and acoustic functions were similar in terms of the relative location of electrodes rather than the absolute location of electrodes, indicating that cochlear implant users may at least partly accommodate to the new patterns of speech sounds after long experience with their normal speech processor. A model of speech recognition was developed by using the present results to modify the articulation index (AI). Independent correction factors were determined for spectral resolution and tonotopic shifting. The modified AI can be used to predict speech recognition in patients with reduced spectral resolution and altered tonotopic location. Limited frequency compression (up to 60%) may improve performance in hearing-aid listeners with high-frequency hearing loss by shifting inaudible speech information into a usable frequency range. The effect of spectral degradation may help refine models of speech recognition for use in hearing aid design. [Work supported by NIDCD]
INTENSITY LEVEL EFFECTS ON SPEECH RECOGNITION
Gerald Studebaker
University of Memphis

There has long been evidence that the speech recognition performance of normal-hearing listeners is worse when speech is transmitted at high (>90 dB) intensity levels. (Fletcher, 1922; French and Steinberg, 1947; Fletcher and Galt, 1950; Kryter, 1946; Pickett and Pollack, 1958; Pollack and Pickett, 1958; and others). However, the literature offers contradictory evidence about whether this effect occurs at lower speech levels. For example, studies by Hawkins and Stevens (1950), Hirsh and Bowman (1953), Hirsh et al. (1954) and Duquesnoy and Plomp (1983) all indicate that the S/N ratio needed to maintain a threshold level of performance remains constant at speech levels up through at least 90 dB SPL. But various other studies suggest that performance over intensity level at a constant S/N ratio begins to decrease at or near normal conversational speech levels (Speaks et al., 1967; Chung and Mack, 1979; Dirks et al., 1982; Hagerman, 1982; Beattie, 1989; Goshorn and Studebaker, 1994).

People with hearing losses often must listen to higher than normal levels of speech and noise when they wear amplification devices. In theory, most of these people should benefit from greater amplification because it raises the sensation level of the speech signal, which increases audibility, and, in turn, intelligibility. However, when amplifier gain controls are turned up, speech and noise spectral levels usually increase together with the result that the S/N ratio remains constant within a given frequency band. Thus, audibility may increase less than expected and if the value of what is audible diminishes, as some studies of normal-hearing subjects suggest that it does, intelligibility may increase substantially less with increased gain than a simple model would suggest. The goals for this paper are to review these issues and to present some new data that support the hypothesis that above normal listening levels have an effect on speech recognition that is the net result of any increase in audibility that occurs and a decrease in the value of what is audible, a decrease that may be the same for normal-hearing and hearing-impaired subjects.
SESSION THREE
MORE THEORETICAL ASPECTS OF SPEECH PERCEPTION AND HEARING LOSS
Moderator: Harvey Dillon

7:50 PM  MODELING SPEECH RECEPTION BY THE HEARING IMPAIRED
Tammo Houtgast
Free University Hospital, The Netherlands

Considering current ideas on the role of the inner and outer hair cells in cochlear pathology, one may expect various types of functional deficits in the signal transduction processes in the cochlea. This may include, for instance, a loss of sensitivity, a reduced dynamic range or frequency resolution, or an increased noisiness in the neural coding process. Each of these deficits may have a specific effect on speech reception, and identifying the exact type of deficit (or mix of deficits) for an individual hearing impaired may be helpful in defining appropriate signal processing strategies. I will report on a series of experiments on speech reception by the hearing impaired, designed to differentiate between various types of functional deficits underlying poor speech reception performance. The experimental conditions include manipulation of the speech signal in terms of signal-to-noise ratio, bandwidth reduction and envelope fluctuations. The individual results lead to a classification of the hearing impaired in a few groups, which may be related to the functional deficits as indicated above.

8:30 PM  EFFICACY OF HIGH-FREQUENCY AUDIBILITY IN SENSORINEURAL HEARING LOSS
Christopher Turner
University of Iowa

Today there is a better understanding of the benefits and limitations of amplification for patients with sensorineural hearing loss. Several lines of research indicate that when the degree of hearing loss exceeds approximately 60 dB HL in frequency regions above 2500 Hz, making speech audible provides little or no benefit to the patient. In other words, some patients with a high-frequency loss have a limited usable bandwidth for amplification. Experiments with hearing-impaired listeners using filtered speech, 2) a correlational frequency-weighting method, and also 3) audiometric thresholds are used to identify (or predict) frequency regions where amplification is ineffective.
For those patients who have a limited usable bandwidth for amplification due to the degree of hearing loss in the high frequencies, we have been testing a frequency compression scheme to improve their speech recognition. The frequency compression scheme we use preserves the natural frequency ratios present in normal speech, as well as preserving the normal temporal cues of speech. Normal-hearing listeners show no deficits in speech recognition for frequency compression factors as low as 0.6. This frequency compression scheme provides consistent benefit for those patients who have limited usable amplification bandwidths, with each patient showing maximum benefit at a specific compression factor.

9:10 PM  
TOWARDS A BEHAVIORAL METRIC OF ACTIVE-MECHANISM STATUS IN LISTENERS WITH SENSORINEURAL HEARING LOSS  
Van Summers  
Walter Reed Army Medical Center

Outer hair cells within the cochlea are thought to be the physiological source of an active mechanism that alters the internal representation of an input signal in important ways. With damage to outer hair cells, reduced influence of the active mechanism may contribute significantly to the deficits that hearing-impaired listeners experience in processing complex sounds like speech in competing sound environments. However, the extent of outer hair cell damage and the underlying status of the active mechanism may vary greatly across listeners with similar losses in absolute hearing sensitivity. These differences in active mechanism status may contribute to the varying degrees of real-world deficit experienced by listeners with similar audiometric thresholds.

The present research examines a possible psychoacoustic means of evaluating the status of the active mechanism in individual listeners. Tone detection and sentence recognition were measured for normally-hearing and hearing-impaired listeners using maskers consisting of equal-amplitude harmonic components summed in positive or negative Schroeder phase. The maskers had identical long-term spectra but their temporal waveforms were time reversals of one another. For listeners with normal hearing, positive Schroeder-phase complexes masked tones and sentences less than negative Schroeder-phase maskers. The two maskers were more nearly equal in effectiveness in the presence of cochlear damage. The findings support an interpretation that involves differences in the shape of the basilar-membrane waveform generated by each masker, and active cochlear processing which enhances the internal signal-to-masker ratio for signals presented in the positive Schroeder masker. Schroeder-phase complexes and related stimuli may provide a valuable tool for gaining a more complete understanding of how compromise of the active mechanism may impact speech processing in competing sounds.
The concept of compression in relation to speech scores and effective modulation suppression
Hans Verschuure, André Goedegebure, Ronald Maas, Wouter Dreschler
Erasmus Medical Center Rotterdam and Academic Medical Center Amsterdam

Compression systems are often used in modern hearing aids. The rationale behind its use is the compensation for the reduced dynamic range in an impaired auditory system. Early research in compression has shown two major effects of compression:

- Speech is intelligible over a wider range of presentation levels (comfort)
- The maximum speech score that is achieved at the optimum presentation level is often somewhat lower with a compression system than with a linear system (intelligibility)

In modern concepts the trade-off between comfort and intelligibility seems to be decided in favor of more comfort, probably because of the better quality of modern hearing aids, because less measurements of speech intelligibility are done and because loudness compensation or loudness normalization seem acceptable goals in itself. Add to this that questionnaires probe mainly comfort and the choice for compression systems is clear.

In using compression systems we should distinguish between different goals like protection against loud sounds, slow level compensation to extend the range of levels over which speech can be understood and fast compression to reduce the range of levels within running speech. The differences between the goals are mainly determined by the time constants. These time constants should be so low for the slow systems that changes in voice effort are compensated without affecting the speech signal itself. At the other hand, protection against loud sounds and modulation reduction within running speech require the use of very fast time constants.

We analyzed the effectiveness of compression systems in a technical measure using a modulated sine wave as the measuring signal and in a speech distribution measure using running speech as the measuring signal. The analysis showed that a
fast single-channel compressor does not affect the level distribution of speech very much if analyzed in narrow bands; this is due to spectral and temporal interactions in the speech signal. The goal of reducing the level distributions within frequency bands will not be reached.

Fast single-channel compression can only serve as a speech enhancement system by reducing level differences between speech components like vowels and consonants. The concept of speech enhancement by fast compression will be presented with data to support its effectiveness for certain groups of hearing-impaired persons. Next, results of field tests with portable, pocket-book sized prototype hearing aids will be presented. We conclude that subjective ratings of users differ from speech intelligibility results. Furthermore, learning (or acclimatization) effects are clearly present over the test period of six weeks.

Researchers should be aware of these effects which question the interpretation of certain date and the use of some well accepted procedures.

9:40 AM

**USING MODULATED SIGNALS TO TEST HEARING AIDS**

William Cole
Etymotic Design, Inc., Canada

Traditional steady-state signals, either broad or narrow-band, provide incorrect estimates of hearing aid output for speech signals. This presentation will describe two new amplitude-modulated test signals and will compare results obtained for these signals with those obtained for real speech. It will also propose a new way of presenting hearing aid performance data obtained with speech-like test signals.

10:20 AM

**BREAK**

10:30 AM

**MODULATION – MASKING AND FILTERING IN A SPEECH RECOGNITION TASK FOR HEARING IMPAIRED AND NORMAL HEARING SUBJECTS**

Rene van der Horst, and Wouter Dreschler
University of Amsterdam, The Netherlands


An experiment was designed to verify whether detection of the modulations in a speech signal is essential for consonant recognition. The objective was to determine whether stochastic modulations can mask the modulations in a speech signal.
signal and whether this masking disturbs speech recognition as effectively as filtering out speech modulations (Drullman et al., 1994).

The envelope of the signal was extracted from every critical band of the speech and was used to modulate a set of narrow-band noise carriers. Certain regions of modulation frequencies between 0 and 20 Hz of the temporal envelopes were filtered out or masked with narrow-band stochastic maskers in order to assess the relative importance of these frequency regions for the intelligibility of medial consonants.

The results of six young sensorineurally hearing impaired subjects and six normal hearing controls were analyzed in terms of recognition scores. There appears to be no consistent effect of the maskers in different modulation frequency regions. But for relatively small masker bandwidths (below 1.5 octave) there is an effect of masker correlation. The scores in the conditions with uncorrelated maskers are consistently higher than those with the correlated maskers. The difference is largest in the 0.74-octave condition where it amounts to 12 percent in recognition score. Consonant recognition appears very robust for the masking or filtering out of speech modulation frequencies. [This research was supported by a grant of the Heinsius Houbolt Fund]

11:10 AM

COMPARISON OF DIFFERENT FORMS OF COMPRESSION USING DIGITAL HEARING AIDS
Brian Moore, Michael Stone, Joseph Alcantara, and Brian Glasberg
University of Cambridge, England

Four different compression algorithms were implemented in wearable digital hearing aids: (a) The slow-acting dual-front-end AGC system [Moore et al., Brit. J. Audio. 25, 171-182 (1991)], combined with appropriate frequency response equalization, with a compression threshold of 63 dB SPL and with a compression ratio of 30 above this (DUAL-HI). (b) The dual-front-end AGC alone combined with appropriate frequency response equalization, with a compression threshold of 55 dB and with a compression ratio of 3 above this. This was intended to give a more accurate impression of the loudness of sounds in the environment (DUAL-LO). (c) Fast-acting full dynamic range compression in four bands. The compression was designed to minimize envelope distortion due to overshoots and undershoots (4-FULL). (d) A combination of (b) and (c) above, where each applies less compression than when used alone; the compression ratio for the dual front-end AGC was reduced to 1.7 (DUAL-4).

Initial prescriptive fitting was based on the concept of giving a flat specific-loudness pattern for a 65-dB SPL speech-shaped noise input [Moore and Glasberg, (1998) Brit. J. Audio. (in press)], and this was followed by fine tuning using an adaptive procedure with speech stimuli [Moore et al., (1998) Brit. J. Audio. (in press)]. Eight subjects with moderate cochlear hearing loss were tested in a counter-balanced design. Subjects had at least two weeks experience
with each system in everyday life before evaluation using the APHAB test and measures of speech intelligibility in quiet (AB word lists at 50 and 80 dB SPL) and noise (ASL sentence lists in speech-shaped noise, or that same noise amplitude modulated with the envelope of speech from a single talker).

Testing is not quite complete at the time of writing, but results so far indicate the following: The APHAB scores did not indicate clear differences between the four systems, although a few individual subjects did show preference for one or the other system. Scores for the AB words in quiet were high for all four systems: at 50 dB SPL, mean scores across subjects ranged from 90% for DUAL-HI to 96% for DUAL-4; at 80 dB SPL, scores ranged from 95% for DUAL-LO and DUAL-HI to 99% for 4-FULL. The speech-to-noise ratios required for 50% intelligibility were very low (indicating good performance) in all conditions; mean ratios across subjects ranged from -6 to -7.5 dB for the speech-shaped noise background at 60 or 75 dB SPL, and from -7.5 to -10 dB for the modulated noise, also at 60 or 75 dB SPL. Performance was similar for the four systems, but there was a slight trend for poorer performance (about 0.6 dB higher SRT) with the 4-FULL system than with the other systems. [This work was supported by ReSound, Audiologic and Danavox, and by the EU (SPACE project), and the MRC (UK)].
SESSION FIVE

NON-LINEAR HEARING AID TECHNOLOGY

Moderator: Chas Pavlovic

7:50 PM  COMPRESSIONG THE DYNAMIC RANGE WITH MINIMUM ILL EFFECT
William Johnson
Threepenny Electronics Corporation

I will discuss ways in which the compression systems in hearing aids are less than optimum. I attempt to associate specific defective mechanisms with user complaints. I then attempt to generalize about potential future improvements.

8:30 PM  A NEW WAY OF VIEWING THE DYNAMIC GAIN OF WDRC CIRCUITS
Steve Armstrong
Gennum Corp, Canada

Because the gain of a compression amplifier is constantly changing, and the effective compression ratio is seldomly as high as the static I/O curve predicts, there exists a need to visualize the behaviour of these automatic systems. Early work on a potential method to simultaneously view the gain by frequency by time of any complex algorithm will be presented.

9:10 PM  TECHNICAL AND CLINICAL ASSESSMENT OF NON-LINEAR HEARING AIDS
Soren Westermann and Carl Ludvigsen
WIDEX Aps, Denmark

Assessment of hearing aids can be divided into: A) coupler measurement, B) real-ear measurement, C) psychoacoustic measurements and D) self-assessment of benefit and drawbacks. Coupler measurements are technical, objective measurements that mainly aims at quality control. Real-ear measurements are technical, objective measurements that are used for verification of hearing aid performance. Psychoacoustic measurements are typically speech discrimination tests, aided threshold, loudness scaling and attempt to provide an objective measure of the hearing aid benefit. Finally, self-assessment tests are subjective measures of hearing aid benefit and drawbacks.

There has always been a desire for objective tests and measurements as these can be standardized, have a high reproducibility and can be performed in a rational way. The drawback of objective measurements is that they tend to represent only
a very specific and often unrealistic listening situation. The advantage of subjective measurements is that - carefully performed - the outcome is the closest we can get to the true benefit/drawback of a hearing aid. The drawbacks are: a large individual spread, difficulties in designing and standardizing the tests and considerable time and resource requirements.

With linear hearing aids objective and subjective measurements have worked pretty well with good agreement between the two, but especially with digital hearing aids using an increasing number of non-linear processes, escalating inconsistencies between the objective and subjective tests are observed. The conflict seems to stem from the highly artificial nature of objective tests combined with the more "intelligent" behavior of the digital hearing aids. As the hearing aids get capable of distinguishing between different listening situations and optimizing their performance accordingly, the outcome of traditional, objective measurements becomes less and less predictable and meaningful.

Either the objective measurements should be downgraded (given up) or a new set of "objective" tests representing various, important listening situations should be developed. One attempt to begin the design of new tests is the ICRA recommendation of new speech-simulating noises for measuring hearing aids and their performance in babble noise. It will be demonstrated how traditional technical measurements of digital aids can generate both unstable and meaningless results. Also, it is demonstrated that it is possible - e.g. by using the ICRA noise - to obtain stable and meaningful performance data from digital aids.

A number of clinical test results with digital aids will be discussed including speech discrimination tests and self assessment tests. The obvious differences between the objective and subjective data will be discussed. In particular, detailed analyses of the expected behavior of the digital aid during each of the objective tests will be presented in order to better understand and explain the differences.
Thursday, May 28

Session Six

Assessment of Hearing Technology

Moderator: Claus Elberling

9:00 AM

Ear-Level DSP Hearing Aids as Research Tools: How Can Their Potential Best Be Exploited?
Graham Naylor
Oticon A/S, Denmark

There are now several ear-level DSP hearing aids available from major manufacturers, and more are on the way. These devices are typified by high sound quality and flexibility enormously beyond that of previous ear-level hearing aids. Such features make the new devices attractive as research tools, allowing many studies to 'come out of the laboratory' and meet the challenges of real-life use. This in turn encourages studies, which are relevant to the true needs of the hearing impaired. These new devices could, if desired, facilitate a new thrust in hearing aid research with the potential to provide new knowledge in areas previously covered rather poorly. It is the contention of this paper that such research is best carried on in an open fashion - that is, researchers should not be fettered by manufacturers' proprietary ideas about what is worth studying, nor prevented from communicating their work. Experiences and results from a scheme of this sort will be presented, and it will be argued that such an open approach is also the best one for the manufacturers' long-term interests.

9:40 AM

Some Problems in the Objective Evaluation of Advanced Signal Processing in Hearing Aids
Wouter Dreschler and M. Boymans
Academisch Medisch Centrum, The Netherlands
J. Verschuure
University of Rotterdam

In this paper the results of two clinical studies with full-digital hearing aids are presented. The first study was a field test with the Widex Senso. This study involved 27 sensorineurally hearing-impaired subjects, following a cross-over design in which the subjects used successively digital hearing aids and newly fitted analogue reference aids in a randomized order. On average, the subjective data are more positive than the objective data. In the end, 20 out of 27 subjects had an overall preference for the digital hearing aid. However, objective data do not support this strong subjective preference. A possible explanation is that the method of analysis (short sentences in a short-duration background noise) is not suited for
the digital hearing aid; the noise-reduction algorithm requires another testing procedure to adapt to the background noise.

In the second study we evaluated the effectiveness of different noise-reduction strategies, by means of a Siemens 4-channel prototype hearing aid (H150) in a laboratory set-up. The results of a paired-comparison test show that strong preferences exist for the use of noise reduction and noise reduction results in more favorable speech perception in noise. However, both results show that different noise-reduction strategies are preferred in different noise conditions.

From these studies some general problems emerge:

- In the subjective evaluation one should realize that the results can easily be biased by the large amount of publicity that was devoted to the new “digitals”. So, it is imperative to rely on robust and reproducible objective evaluation techniques.
- The technical evaluation is very complex, and new techniques should be developed. The specification of the dynamic behavior of active signal processing in hearing aids needs to be standardized. It will be suggested to use well-described noises for the measurement of the dynamic behavior of hearing aids.
- The objective evaluation in noise is complicated by the long adaptation times used for the noise-reduction processing. The type of noise that is chosen appears to be a determinant factor for the final outcome. One way to come across is again to put more effort in the standardization of background noises.

10:20 AM  Break

10:30 AM  Laboratory Assessment of Advanced Signal Processing Algorithms for Hearing Aids

Patrick Zurek, Julie Greenburg, Merry Brantley, Andrew Brughera, and William Rabinowitz
Massachusetts Institute of Technology

The goal of this project is to evaluate advanced signal processing algorithms for hearing aids under realistic conditions. This evaluation will be performed through field studies of hearing-aid users equipped with wearable signal processors that will implement promising algorithms. The field study will be performed at two sites - M.I.T. and the University of Memphis. Prior to the field studies, laboratory studies are being conducted to select the algorithms to be implemented. In addition, wearable digital signal processor units capable of implementing those algorithms are being designed and built. Algorithm selection is being done at M.I.T. with users of linear binaural hearing aids. Design and construction of the wearable processor is being done at Sensometrics Corporation.
Two areas of signal processing -- microphone array processing and feedback reduction -- have been chosen for investigation because these promise the greatest benefit to the largest range of hearing aid users. In order to explore alternative microphone array strategies, two different configurations of four microphones are being used. One of these configurations has two omnidirectional microphones on each of two behind-the-ear hearing aid shells and the other has four directional microphones on a headband. For both microphone configurations the signals are delivered to the ears by in-the-ear modules containing only receivers.

Signal processing algorithms are being tested and refined in the laboratory phase of the project. Comparisons are made among algorithms within each of the two signal processing areas and with subjects' personal hearing aids. An extensive fitting protocol has been developed to find the binaural frequency-gain characteristics that represent a good compromise between their personal hearing aid characteristics and those prescribed by R-NAL, while also factoring in the feedback path. Evaluations are directed at assessing the extent to which algorithms: 1) improves speech intelligibility in noisy environments; 2) prevent annoyance from loud sounds; and 3) affect sound localization ability. Current results of these laboratory evaluations will be presented. [Work supported by NIDCD.]

11:10 AM

THE RELATIONSHIP BETWEEN STIMULUS CONTEXT, SPEECH AUDIBILITY, AND PERCEPTION FOR CHILDREN AND ADULTS
Pat Stelmachowicz, B M. Hoover, D.E. Lewis, and R W.L. Kortekaas
Boys Town National Research Hospital

A prescriptive hearing-aid fitting procedure typically is used to fit hearing aids to infants and young children who cannot actively participate in the evaluation. The majority of available prescriptive procedures were developed from data obtained from hearing-impaired adults (e.g., preferred gain, use gain, loudness measures). In a previous study with hearing-impaired adults [Stelmachowicz et al., Ear & Hearing, 19, (1998)], we assessed the audibility of speech associated with a 2-channel WDRC hearing aid fitting that was based on loudness growth measures. Results suggested that, under many conditions, a loudness-based algorithm provides adequate audibility of speech only if one assumes that the listener can take full advantage of linguistic cues. The purpose of the present study was to investigate the relation between stimulus context, audibility, language competence, and performance on a speech recognition task. Study design included 20 normal-hearing adults, 30 normal-hearing children (5-7 yrs.), and a group of hearing-impaired children (5-13 yrs.). An estimate of receptive language age was obtained for both groups of children. Performance-intensity (PI) functions were obtained for 60 semantically correct and 60 semantically anomalous sentences designated as high and low predictability stimuli, respectively. Logistic functions were fitted to rau-transformed PC values to estimate the slope of the PI functions and the levels required to achieve 50% performance. An audibility index also was calculated at each presentation level.
8:30 PM  SOME DEVELOPMENTS IN DIRECTIONAL HEARING AIDS
Dave Preves
Micro-Tech Hearing Instruments

The use of directional microphones is one of the few methods available in hearing
aids to increase SNR. A few dB improvement in SNR can produce a relatively
large improvement in word recognition in noise. The advantage of directional
microphone usage in BTE hearing aids has been well documented.

Many hearing health care professionals believe that until recently directional
microphones have not been available in ITE hearing aids. Directional
microphones have also been used for many years in ITE hearing aids. However,
since their introduction over 20 years ago, directional ITE hearing aids virtually
disappeared from the marketplace. In fact, popularity of all types of directional
hearing aids waned considerably since their introduction in the 1970’s. Smaller
microphones now available permit easier incorporation of directionality into ITE
hearing aids.

This discussion explores why early ITE directional hearing aids were not
generally successful in the marketplace and why a switched directional-
omnidirectional ITE hearing aid may be more viable. Results are reported from a
study evaluating the effectiveness of such a switched directional/omnidirectional
ITE instrument. Data for a Bi-CROS version of a switched
directional/omnidirectional ITE instrument is also reported.

Testing directional hearing aids and predicting the amount of directionality they
may provide in real life requires special consideration to reduce artifacts.

9:10 PM  SIGNAL PROCESSING FOR HEARING AIDS BASED ON AUDITORY
MODELS
Birger Kollmeier
Universit Oldenburg, Germany

Models of the normal and impaired hearing process can be used to design
"intelligent" digital hearing aids for various reasons. One reason is that our
knowledge about the impaired processing in sensorineural hearing-impaired
patients directly influences our goals with respect to the algorithms to be
incorporated into hearing aids. At least the following components play a
significant role:

- The "attenuation" component (characterized, e.g., by the audiogram or the
  MCL across frequency) plays a dominant role and can be compensated by
  appropriate (linear, frequency-dependent) amplification.

- The "loss of compression" component (characterized, e.g., by the slope of
the loudness growth curve or the loss in outer hair cells) influences various secondary variables (i.e., frequency resolution and temporal resolution, [Kollmeier, B., Derleth, R.-P., and Dau T., Modeling the "effective" auditory signal processing for hearing-impaired listeners, in Psychophysical and Physiological Advances in Hearing, Palmer, A.R., et al., Editors. 1998, Whurr Publishers: London. p. 482-490] and can be compensated by an appropriate nonlinear, frequency-, level-, and bandwidth-dependent dynamic compression.


- A "central" component of the hearing loss remains which can be compensated by appropriate signal enhancement and noise reduction techniques. In addition, appropriate control algorithms have to be incorporated that adapt the appropriate algorithm to the respective acoustic situation [Woods, W.S., et al., Using multiple cues for sound source separation in Psychoacoustics, Speech and Hearing Aids., B. Kollmeier, Editor. 1996, World Scientific: Singapore. p. 253-258].

A review will be given on these different factors in sensorineural hearing loss and their respective implications with regard to hearing aid algorithms. Also, results will be presented on various attempts to incorporate auditory models into laboratory-based (desktop) DSP hearing aids.
SESSION EIGHT

MULTIPLE EARS, MULTIPLE MICROPHONES

Moderator: Arne Leijon

9:00 AM  LABORATORY AND REAL-WORLD EVALUATION OF MULTIPLE MICROPHONES IN HEARING AIDS
Dave Fabry
Mayo Clinic

Directional microphones are enjoying a resurgence in popularity for use in hearing aids, due in part to their availability in in-the-ear styles, and to the popularity of several recent digitally programmable hearing aids that use electrically coupled microphone arrays. In addition, several beamforming microphone arrays will be available commercially in the U.S. within the next year.

Clinical evaluation of these devices has often focused on favorable conditions, where speech and noise are spatially separated, with noise sources located a "null" points for the microphone arrays in low reverberation test rooms. Although these conditions provide an opportunity to showcase the benefit of these devices, they do not represent actual conditions that patients will encounter during daily use. The focus of this presentation will be describe experiments underway that attempt to relate performance by normal and hard-of-hearing patients under these booth conditions to those in "real-world" test environments, where digital recordings were made under several listening conditions and replayed to subjects for a sentence recognition task (Speech-in-Noise Test). Results will be discussed in terms of the predictive capability of booth tests for "real world" conditions encountered by patients wearing these devices.

9:40 AM  BINAURAL SPEECH INTELLIGIBILITY IN NOISE WITH AND WITHOUT HEARING AIDS
Dianne Van Tasell
Starkey Laboratories, Inc

Normal hearing listeners enjoy an advantage of binaural over monaural listening when speech and noise originate from different azimuths. There are two components of this binaural advantage: Head shadow is the monaural improvement in intelligibility at the ear on the side of the head opposite the noise source, resulting from the acoustic "shadow" cast by the head for the noise frequencies above about 1000 Hz. "Squelch" is the intelligibility improvement...
over the monaural performance of the shadowed ear resulting from the addition of binaural information from the other ear. In the work to be reported, unaided binaural listening advantage was measured in both normal-hearing and hearing-impaired listeners. For the hearing impaired listeners, aided binaural listening was also assessed. Results were analyzed using a model based on the one described by Zurek [P.M. Zurek, in G.S. Studebaker & I. Hochberg, Acoustical Factors Affecting Hearing Aid Performance] of directional effects in speech intelligibility. When standard AI calculations and binaural interaction assumptions were used, the model predictions were accurate for both normal and hearing-impaired listeners in the unaided condition; although aided performance was consistently overpredicted. As predicted by the model, access to head shadow is limited by peripheral hearing loss above 1 kHz, as well as the amount of insertion gain the hearing aid can provide. Binaural advantage for speech is shown to be a complex effect of interactions between hearing loss and hearing aid gain characteristics. Potential usefulness of the model for evaluation of binaural hearing aid fittings will be discussed.

10:20 AM  Break

10:30 AM  Optimizing sound localization by people wearing hearing aids
Denis Byrne
National Acoustic Laboratories

Hearing impairment is almost always accompanied by reduced ability to locate sounds and, more generally, a disrupted sense of spatial orientation. This leads to a variety of problems that are likely to result in reduced satisfaction with and acceptance of hearing aids. Research permits several recommendations about how hearing aids should be fitted to optimize sound localization and promote a natural sensation of spatial orientation. For localization, bilateral fittings are usually advantageous for people with moderate or severe hearing losses. Such fittings, when using occluding earmoulds, do not always provide better localization for people with mild hearing losses. Conductive/mixed hearing losses are associated with poorer localization than similar degrees of sensorineural hearing loss. Aiding often results in improved localization, sometimes greatly so and sometimes more so with some types of earmoulds than with others. People with sensorineural hearing loss and good low frequency hearing may suffer a decrease in aided, compared with unaided, localization when bilaterally fitted with occluding (closed) earmoulds. This decrement, apparently resulting from phase distortions caused by a mixture of amplified and unamplified sound, can be avoided by using open earmould fittings. Open earmould fittings can also be beneficial for people with low and mid frequency hearing losses but good hearing above 4000 Hz. Such earmoulds, especially a new type designated the “sleeve” earmould, permit good vertical plane localization by leaving the pinna unobstructed and permitting unamplified high frequency hearing. Participants in a field trial consistently preferred the sleeve earmould. They reported advantages
related to a more natural sound quality, better externalization of sound and more comfortable listening. A series of clinical recommendations will be presented for considering sound localization and related difficulties and for fitting hearing aids in ways that will minimize such problems.

11:10 AM  LOCALIZATION, SNHL, AND AMPLIFICATION
Stuart Gatehouse
MRC Institute of Hearing Research, Scotland

Everyday listening often involves integration of auditory information arriving at the two ears, and specific binaural effects include the improvement in speech identification abilities over and above monaural listening and processes which can underpin the localization of sounds. Motivation for the current experiments arises from a program of work to develop and evaluate procedures for the simultaneous assessment of speech identification abilities and the extent to which binaural listening can allow listeners to capitalize on the spatial separation of speech and noise sources, simultaneous with the ability of listeners to identify the source of sound. The resultant program investigates the psychophysical bases of these phenomena and the extent to which conductive and sensorineural hearing loss compromise these abilities, in addition to the efficacy of management for these conditions.

A preliminary step has been the development of localization paradigms, and current experiments concerning localization in the frontal horizontal plane in the presence of a background noise demonstrate the extent to which interaural time delay (ITD) and interaural level difference (ILD) cues are compromised by the presence of noise for normal hearing listeners, and the extent to which listeners can combine the two types of cues. Further experiments on listeners with sensorineural hearing loss document the additional decrements in performance associated with impairment and the extent to which they are related to the availability of ITD and ILD cues.

A third experiment investigates the relationship between low frequency sensorineural hearing loss and the use of ITD cues to achieve sound source identification for both unaided conditions involving amplification. The results show that localization abilities based upon ITD cues are degraded in sensorineural hearing loss in systematic ways which are not a function of simple loss of audibility, and furthermore are not restored to those enjoyed by normal hearing listeners by the provision of amplification.

Further steps have been taken to develop the combined speech identification and localization paradigm and this contribution will set out the series of experiments to assess the impact of conductive and sensorineural hearing loss on binaural abilities and will outline the ways in which specific psychoacoustical tests of binaural function related to the use of interaural time, level and decorrelation cues
can be systematically evaluated using psychophysical headphone experiments and free field paradigms.
Poster Abstracts

Contributed posters have been divided into two groups of equal size. The posters in the first group will 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 of posters will be displayed from Wednesday afternoon through Friday noon.

POSTER GROUP A
Monday evening–Wednesday noon

PA1
Hearing aid outcome measures after six months of hearing aid use
Nancy Nelson Barlow and Larry Humes
Indiana University

Data on hearing aid outcome measures will be presented for 21 geriatric (ages 59 - 89 years) hearing-impaired subjects. Data were collected at one month and six months following hearing aid fitting. The following outcome measures were obtained: 1) Hearing Aid Performance Inventory; 2) Hearing Aid Satisfaction Survey; 3) Hearing Handicap Inventory for the Elderly; 4) CUNY-NST at 65 dB SPL in noise (+8 SNR); 5) Judgments of Sound Quality for speech and music; and 6) average hearing aid usage (average number of hours per day) from hearing aid "diaries".

The discussion will address changes in outcome measures that occur during the first six months of hearing aid use. In general, the group showed significant hearing aid benefit. Hearing aids were judged to be helpful and subjects described themselves as "satisfied" or "very satisfied".

The average number of hours of daily use was between eight and nine. Outcome measures did not change significantly over time. There were, however, individual differences among the group and these differences will be discussed in the poster. [This work was supported by a grant from NIH]

PA2
Speech recognition as a function of spectral resolution in hearing impaired listeners
Siu-Ling Chi and Christopher W. Turner
University of Iowa

It has been hypothesized that poor spectral resolution is one of the contributing factors to the difficulty that hearing-impaired listeners have in understanding speech. In this study consonant recognition was measured as a function of the number of frequency channels (or degree of spectral resolution) in both normal-hearing and hearing-impaired listeners. Speech was divided into various numbers of frequency bands (1, 2, 4, and 8 bands) and the amplitude of each speech band was used to modulate a corresponding noise band carrier. The bands were then recombined. An unprocessed (Full Speech) condition was also included. High-pass amplification was provided to ensure audibility for the hearing impaired listeners.

Our working hypothesis was that consonant recognition would be equivalent between the two groups of listeners for conditions of a small number of channels (where the stimulus itself served to limit spectral resolution). On the other hand, for higher numbers of channels, where the spectral resolution of the impaired auditory system becomes the limiting factor, the hearing-impaired listeners should show a deficit in performance compared to the normals.
Results were consistent with previous studies, in that for 1-channel speech the two groups performed comparably (approx. 20% correct, Tuner et al. 1995). However, impaired listeners performed poorer than the normal listeners for all other conditions, including the 2-channel condition. This result is similar to that found by Fishman et al. (1997) for poorer-performing cochlear implant patients. Thus, even though audibility was controlled, temporal resolution was measured to be equivalent between the two groups, and spectral resolution was presumed to be limited by the stimulus itself, the hearing-impaired listeners still showed a deficit. It is suggested that there are additional deficits that cause the poorer speech recognition in hearing-impaired listeners.

**PA3**

**Derivation of the NAL-NL1 prescription procedure for nonlinear hearing aids**

Harvey Dillon, Denis Byrne, Richard Katsch, Gitte Keidser, and Teresa Ching
National Acoustic Laboratories, Australia

The NAL-NL1 procedure prescribes insertion gain as a function of input level for non-linear hearing aids. The rationale behind the procedure is that for any input level, predicted speech intelligibility should be maximized subject to the constraint that the overall loudness of speech is equal to or less than that perceived by normal hearing people when listening to the same input signal. Speech intelligibility has been predicted using a modified version of the Speech Intelligibility Index method (also known as the Articulation Index). The modifications allow for the reduced ability of hearing impaired people to extract useful information from speech even when it is audible. These modifications are substantial for any frequency region where hearing loss is severe. Loudness is calculated using a published loudness model that allows for the effects of hearing loss. The gain-frequency response prescribed by the model is different from that which would be needed to normalize loudness independently at all frequencies. For average input levels, the gain-frequency response is, however, similar to that prescribed by the established NAL-RP formula. Also, a consequence of the mathematical optimizations is that loudness is held approximately constant across a wide range of frequencies, which is the key rationale behind earlier NAL procedures.

**PA4**

**Fitting high frequency hearing loss using digital technology**

Ole Dyrlund
GN Dananox, Denmark

Two digital signal processing algorithms, which significantly increase the fitting possibilities for particular clients with high frequency losses and associated large increase in recruitment over frequency, are presented: A digital WDRC compression system with high frequency resolution. This system has 13 overlapping frequency bands, which makes it possible to provide differences in effective compression ratio from 1 to 3 over about one octave. In addition this architecture with overlapping bands eliminates signal-processing artifacts known from traditional multi-channel compression systems. A new active acoustical feedback suppression system with no audible sound probe during normal operation of the hearing aid and providing about 10 dB extra real ear gain in almost all listening situations. These systems will be described and results of provisional clinical testing will be presented.
Variance in loudness scaling data versus fine-tuning of the hearing aid
Claus Elberling
Oticon Research, Denmark

Over the years much effort has been made to develop adequate methods to measure loudness related to hearing aid fitting and a series of different loudness scaling procedures have been processed. Analysis of published loudness scaling data from subjects with normal hearing and with hearing impairment demonstrates both differences and similarities between the data obtained with the different procedures. A unified presentation of the data enables the conclusion that only in about 25-30% of the hearing impaired will the measurement of individual loudness functions lead to hearing aid settings beyond what would be covered by a ‘normal’ fine tuning. Further, if the variance of the target setting also is incorporated this number approaches zero.

Aided growth of masking for speech and non-speech signals
Todd Fortune
Telex Communications

Oxenham and Plack (1997), recently described a behavioral measure to indirectly estimate basilar membrane nonlinearity by comparing the slopes of on and off-frequency growth of masking (GOM) functions. In the current investigation on and off-frequency GOM functions for speech and non-speech signals were obtained in normal and impaired ears, with an emphasis on aided listening conditions. Ten hearing-impaired listeners with moderate, sloping, high-frequency hearing losses participated. Five normal hearing listeners provided baseline data. GOM functions were obtained unaided and aided for three sets of test signals. Signal set 1 consisted of a 5ms, 4000 Hz tone pip masked by a 100ms 1/3 octave band of noise centered at either 4000 Hz (on frequency conditions) or at 2500 Hz (off frequency condition). Signal set 2 consisted of the fricative /f/ masked by the stop consonant /t/ (on-frequency condition). Signal set 3 consisted of the fricative /s/ masked by the vowel /e/ (off frequency condition). GOM functions were obtained using a two-interval forced choice forward masking paradigm. Masked thresholds for each probe signal were obtained at masker levels of 50-90 dB SPL. All testing was performed monaurally in the sound field, using real ITE WDRC hearing aids fit to individual subjects using the DSL/io method.

Results to date indicate that under non-speech conditions, normal GOM functions show evidence of nonlinearity, similar to that reported by Oxenham and Plack (1997) and by Nelson and Schroder (1997). This nonlinearity is indicated by GOM slopes that differ between on and off frequency conditions. In general, impaired listeners fail to show such nonlinearity unaided, and in fact reveal lower than normal slopes for both maskers. Under aided conditions, GOM slopes increase for both conditions due to lower masked thresholds at the lowest masker levels. On-frequency, the GOM slope becomes essentially normal, but the off-frequency slope remains abnormally low. Preliminary findings with speech signals suggest a similar pattern, although GOM functions are shallow under all conditions for both normal and impaired listeners. Results will be discussed in terms of circuit algorithm, time constants, fitting method, and psychoacoustic factors.


**PA7**

**Improved method for equalization of magnitude and phase insertion effects in hearing aid fitting**

Shawn Gao and Sigfrid Soli
House Ear Institute

The method of equalizing a hearing aid's magnitude and phase insertion effects, as described in "Method for Fitting Binaural Hearing Aids" (Shawn Gao et al., Issues in Advanced Hearing Aid Research, 1994), requires three steps: a) design a Hearing Aid Equalization (HAE) filter to equalize the magnitude and phase insertion effects of the hearing aid; b) design a linear phase Hearing Loss Compensation (HLC) filter to compensate for the hearing loss; and c) convolve the HAE and HLC filters to produce an FIR filter that can be programmed in the hearing aid.

There are several potential limitations of this method of magnitude and phase equalization. First, it is difficult to find an optimal allocation of filter coefficients for the HAE and HLC filters. Second, the linear phase HLC filter necessarily introduces a group delay of half of its length which can have undesirable effects on acoustic feedback. Third, because of the symmetric property of linear phase FIR filter coefficients, only half of the filter coefficients carry unique information. All of the limitations become more problematic with the limitations in filter length and computational complexity imposed by the power and size constraints of the hearing aid circuit.

We have developed a new method that combines the three filter design steps into a single step by using an optimal Wiener filter design technique. This method requires acquisition of the unaided signal and the aided signal with a probe microphone placed in the ear canal. The unaided signal is preprocess with the desired HLC filter, which is designed to match the chosen insertion gain target. This preprocessed signal defines the target aided response. Both the acquired aided response and target aided response are used to compute an FIR filter with the Wiener optimal filter design technique that, when applied to the acquired aided response, will produce a response that matches the target aided response. The FIR filter derived in this manner not only equalizes the magnitude and phase insertion of the hearing aid, but also provides appropriate frequency shaping, as specified by the insertion gain target, to compensate for the hearing loss. In addition, the acquired aided response and the preprocessed target aided response can be aligned in the time domain so that the optimal filter response has minimum delay. A detailed description of proposed filter design procedure will be provided together with examples of its use with KEMAR and with hearing impaired individuals.

**PA8**

**Development of a test of sound localization: The Source Azimuth Identification in Noise Test (SAINT)**

Shawn Gao and Sigfrid Soli
House Ear Institute

Pure-tone thresholds are not strongly predictive of auditory performance in many noisy and supra-threshold listening situations. We are developing a measure of Source Azimuth Identification (SAINT) in noise as a means of evaluating sound localization ability in the soundfield and under headphones. Soundfield and
headphone data will be presented for normal and hearing impaired subjects. Validity of the headphone SAINT will be described using correlational analyses that compare soundfield and headphone performances.

A signal detection theory analysis of auditory processing: phase and frequency uncertainty
Lisa Gresham and Leslie Collins
Duke University

Two common difficulties experienced by individuals with hearing impairment are a loss of sensitivity to sound and an increase in difficulty discriminating between sounds with similar frequency content. Traditional hearing aids, which generally provide some frequency-dependent gain, often do not improve an individual's ability to comprehend complex acoustic signals such as speech in a noisy environment. One hypothesis is that this occurs because the signal processing strategy employed by these hearing aids only addresses the issue of reduced sensitivity. In order to design better hearing aids, new strategies must be developed which both capitalize on the signal processing functioning which still remains in the damaged ear and compensates for the signal processing which has been lost. To realize this goal, a better understanding of auditory processing must be obtained.

Auditory processing has been studied using both bottom up methods which begin with the acoustic stimulus and work through the peripheral auditory system towards the central nervous system, and top down approaches which attempt to infer the underlying mechanisms of the auditory system from behavioral responses. The eventual goal is to be able to link acoustical signals, which can be manipulated by hearing aids, to the perception of sounds. However, this task has proven difficult.

Typical bottom up approaches include the use of signal detection theory to fit Receiver Operating Characteristic curves to neural data and the use of computational auditory models to predict neural firing patterns. Although both of these approaches are valuable, neither addresses an issue critical to the design of improved remediation devices, namely signal "detectability". Other techniques based on signal detection theory precepts have been designed to predict detection performance, or behavioral responses, for specific psychophysical tasks, but often lack physiological details. The theoretical performance predicted by such techniques often exceeds experimental performance on a task such as the detection of a tone in noise. The observed discrepancy is commonly attributed to additive "internal noise", the variance of which is adjusted until matching results are obtained.

Our previous work has shown that incorporating signal detection theory into the framework of a computational auditory model yields more accurate predictions of detection performance than traditional methods [Gresham and Collins, J. Acoust. Soc. Am., May 1998]. However, some discrepancy still exists between theoretical and experimental results. The work presented here investigates the role frequency and phase uncertainty might play in the detection of a tone in noise. It is observed that phase uncertainty does not affect detection performance for this task. This result agrees with psychophysical data which indicate that phase does not affect an individual's perception of a tone. In contrast, frequency uncertainty appears to have a greater affect on detection performance which may help explain one way the auditory system does not behave
Nordic minimum requirements for clinical test of hearing aids
Bjorn Hagerman
Karolinska Institutet, Sweden

In the Nordic countries we have common requirement specifications for hearing aids. For example, there are limits on allowable internal noise levels and on distortion levels. However, some new digital hearing aids with adaptive signal processing may not be possible to measure according to the international standard methods (IEC 60118). They may reduce the gain for signals that are not modulated. Then our requirement specifications may not be applied since they rely on measurements based on these standardized methods. Therefore the interest to use clinical tests is rapidly increasing. However, since such tests are very expensive compared to ordinary electroacoustical measurements, it is important to find a common method, the result of which can be used in several countries.

Within the Nordic Cooperation on Disability we have a group preparing documents for harmonization of requirements on aids for hearing impaired people. We have now prepared the document. Nordic minimum requirements for clinical test of hearing aids, which hopefully will be issued this summer. Some important points:

- Minimum requirements mean that tests and questions may be added.
- The manufacturer of the hearing aid shall specify the fitting procedure.
- The test hearing aid shall be compared with a reference hearing aid, which may not be the user’s own one.
- A questionnaire is included in the

Comparison of some fitting methods for non-linear (WDRC) hearing aids
Carolina Johansson and Arne Leijon
Kungl Tekniska Hogskolan, Sweden

Some commercially available threshold-based prescriptive methods for non-linear hearing instruments are compared for three common listening situations. Two of the selected fitting methods are intended to be manufacturer-independent: Fig 6 and DSL. The other methods are implemented in the NOAH system specifically for the manufacturers’ hearing aids. The NAL-R procedure (Byrne & Dillon, 1986) as well as two theoretical prescriptions have been implemented and used as references for the comparison: Optimization of SII, and loudness density normalization implemented according to a loudness model suggested by Moore.

Theoretical prescriptions are derived for three hypothetical hearing-aid candidates with known audiograms, representing mild to moderate relatively flat to sloping hearing losses.

For these persons we have calculated prescribed insertion gain responses for three listening environments: (1) A relatively easy listening situation with speech at 50 dB SPL and speech-shaped noise at 40 dB SPL, (2) speech at 65 dB SPL with noise at 60 dB SPL, and (3) both speech and noise at 75 dB SPL, representing a very difficult listening situation.

The prescriptions are documented in terms of insertion gain responses as well as
estimated loudness and Speech Intelligibility Index (SII).

**POSTER GROUP B**  
Wednesday afternoon-Friday noon

**PB1**  
Speech perception of reduced spectral resolution in noise background by normal hearing listeners  
Bom Jun Kwon and Christopher W. Turner  
University of Iowa

The purpose of this experiment was to investigate the effect of spectral resolution on speech perception by normal-hearing listeners in noise babble background. Speech stimuli were processed through several frequency bands, within which fine spectral information was removed, while the temporal envelope of waveform in each band was unchanged. Subjects identified consonants in /aC/a recorded by 4 speakers for speech and 12-talker-babble processed with 1, 2, 4, and 8 bands, in addition to the unprocessed speech and babble, for a range of S/N ratios. For high S/N ratios our results agree with earlier results, in that performance improved with increases in the number of bands, and higher transmission of information was observed for voicing and manner than place of articulation. However, as S/N decreased, the advantage of better spectral resolution was reduced. This suggests that for speech in a competing babble background, increased spectral resolution not only increases the intelligibility of speech, but at the same time increases that of background babble, so that babble could be a more effective distracter in such cases. Generally speaking, although 4 band-processed speech may provide sufficient spectral information in high S/N ratio cases, detailed feature analysis showed that the performance of identification varies along features of individual consonants, and speaker of recorded stimuli. For example, /k/ vs /t/ are confusable in 4-band condition, /f/ vs /θ/ in 8-band condition, and so on. Besides, interestingly, some speaker’s stimuli are more susceptible (or, more resistant) to loss the place information than others through the same processing. In conclusion, the benefit provided by fine spectral resolution depends upon the task, S/N ratio, and the stimuli.

**PB2**  
Real time digital hearing aid based on low power general purpose digital signal processing chip  
Neeraj Magotra  
University of New Mexico  
Pedro Gelabert, and Wai Lee  
Texas Instruments

This paper describes the development of a digital binaural hearing aid. The device is based on a general purpose digital signal processing (DSP) chip. Currently the chip used in the prototype model is Texas Instruments TMS320C3X chip. This DSP chip is a floating point chip and allows us to validate signal processing strategies for the hearing impaired without worrying about quantization effects.

The device uses a sampling rate of 32 KHz for each channel (left and right ears respectively) giving an effective bandwidth of 16 KHz. It permits us to implement the following digital signal processing algorithms: spectral shaping, adaptive (single channel) noise reduction, multichannel compression and binaural time-delay. The therapist essentially can choose from a combination of these algorithms, if necessary, to provide the most effective improvement in the subjects hearing capability. We have currently developed our own custom hardware and software to implement these algorithms on a
PC workstation (used by the therapist to program the device for each individual subject) as well as on a wearable prototype device.

Obviously, in order to prove the viability of this approach, we need to port the algorithms to smaller, low power DSP chips. Such chips are being developed primarily for use in the telecommunications market. We propose to port this system to Texas Instruments TMS320C54X family of fixed point DSP chips. This will necessitate performing a quantization analysis of the algorithms but this process will be worth the price since this chip is a powerful low power DSP chip. We will present two case studies on ultra-low power programmable DSPs. The first case study evaluated the TMS 320C549 to voltage levels as low as 1.05 V operating at 10 MIPS. This same chip can operate at 100 MIPS at 2.5 V (I/O voltage of 3.3 V). The second case study, used a modified process to obtain 0.6 V operation with 10 MIPS and 1.0 V operation with 63 MIPS.

These chips could feasibly provide the capability of designing an effective, fully programmable (in the algorithmic as well as parametric sense) digital hearing aid that is also aesthetically pleasing.

**PB3**

The behaviour of non-linear (WDRC) hearing instruments under realistic simulated listening conditions

Peter Nordqvist and Arne Leijon
Kungl Tekniska Hogskolan, Sweden

It is well known that the conventionally measured “static” AGC characteristics can not adequately describe the effective behavior of non-linear hearing instruments in everyday listening situations [e.g. Elberling, 1996 #4962, Verschuure, 1996 #4860]. Conventionally measured attack and release times are not sufficient to describe the temporal characteristics of non-traditional digital AGC algorithms.

This work attempts to illustrate some important practical consequences of the characteristics of non-linear (WDRC) hearing instruments in common conversational listening situations. Corresponding input and output signal are recorded simultaneously, using test signals consisting of two alternating voice sources from different distances, (1) in quiet, (2) in cocktail-party babble, (3) outdoors in fluctuating traffic noise together with brief instances of high-level transient sounds. The effective insertion gain frequency response is displayed for each of the two voice sources in each of the simulated listening situations. The effective compression is also illustrated, with regard to three different time scales of input-signal variation: (1) the “syllabic” compression within each of the two voice sources, (2) the gain adaptation between two alternating voice sources, and (3) the slow adaptation to changing overall acoustic conditions.

These non-linear effects are exemplified using a few commercially available hearing instruments, including a K-Amp instrument, a two-channel instrument with fast-acting compression, and a digital (Widex Senso) instrument.

**PB4**

Hearing aid fitting based on the nonlinearity of the individual ear

Henrik Olsen, Ann-Cathrine Lindblad, and Åke Olofsson
Karolinska institutet, Sweden

A sensorineural hearing loss may be characterised by a certain decrease of compressive nonlinearity (loss of outer hair cells) and a certain increase of attenuation (loss of inner hair cells). The purpose of
this paper is to report the initial results from a study conducted primarily to identify the amount of residual nonlinearity and on this basis try to suggest the best prescriptive strategies. For subjects with some degree of residual nonlinearity the aim is to place the level of speech optimally for preservation of the temporal modulation characteristics of speech. The speech levels of various frequencies are chosen according to this. A slow acting compressor (AVC) with a high compression ratio is used to keep these levels presumably optimal. For the subjects with a more or less total loss of compressive nonlinearity the aim is to set up individual models, which is used to fit the parameters of a fast acting compressor. This will be done in order to compensate the loss of compression in the impaired ear.

The first part of the project, which will be reported here, is a clinical study conducted in order to try to identify an optimal frequency-gain response on the subjects with residual compressive nonlinearities. By means of the PMTF method, Psychoacoustical Modulation Transfer Function, it is possible to show the level dependent ability of the ear to maintain the intensity variations of the speech signal (Lindblad et al. 1993). This will also reflect the active nonlinearity of the inner ear. 13 subjects with sloping symmetrical high frequency sensorineural hearing loss participated. All subjects had at least one year of experience with their own aid and they were all fitted monaurally. The prescription based on PMTF was described as sensation levels in three octave bands and was called PMTF-AVC. As a reference we chose a frequency-gain response according to NAL-R, NGAM (Nal frequency-gain response at moderate levels). Both prescriptions were implemented in the Oticon JUMP-1 platform configured as a slow acting compressor with a compression ratio of 6.

The study was designed as a prospective, single-blind, crossover, controlled study. There were two different trial periods: a 2x4 weeks crossover period with either PMTF-AVC or NGAM and a comparison period with both settings. The results reported now only cover the first period of trial. The results of speech reception in noise did not reveal any significant differences between the two prescriptive settings at the speech levels 60 and 75 dB SPL despite of the fact that there was a 2-3 dB more high frequency emphasis on the PMTF-AVC. Sound quality ratings showed a significant interaction between prescription setting and quality dimension. PMTF-AVC was rated significantly higher for clarity than NGAM. For the softness rating the NGAM was rated higher. The subjective evaluations did not show any difference between the two prescriptive settings. The attempt to show difference in speech reception between the two settings failed. This could be interpreted as if a smaller deviation in frequency response is not crucial for speech reception at least not for patients with residual compressive nonlinearities. Not surprisingly, the results indicate a trade-off between clarity and softness.

Controlled methods for assessing speech intelligibility in noise with directional hearing aid microphones
Larry Revit, Robert Schulein, and Mead C. Killion
Etymotic Research

Aside from improving the audibility of sounds through adaptive and selective amplification, the only effective current means of improving speech intelligibility in noise for hearing-aid wearers may be the use of directional microphones. However, the objective assessment of speech intelligibility in noise with directional hearing aid microphones presents a special set of experimental control problems, if results can adequately generalize to the real world. Our purpose in this endeavor is to create audio recordings used to compare intelligibility test results using directional and omni-directional in-the-ear microphones in the same adverse listening environments. The special problems relate to the calibration of the recorded test materials with respect to controlling the free-field signal-to-noise ratios that existed in the test environments.

We have developed two methods that address the above purpose. Both methods involve the recording of "real-world" diverse listening environments, along with intelligibility tests, using directional and omni-directional hearing-aid microphones simultaneously. We will describe both methods and present audio recordings demonstrating the results.

The use of sinusoidal modeling to preserve spectral contrast in speech
Janet Rutledge, University of Maryland
Juan-Carlos Tejero-Calado, University of Malaga, Spain
Peggy Nelson, University of Baltimore

Hard-of-hearing listeners have difficulty identifying spectral peaks in both noise and speech (Nelson & Revoile, 1998), particularly in high-frequency regions. This deficiency relates to their difficulty using important format information to identify speech. Sinusoidal modeling was applied to CID sentence materials to preserve spectral peaks for improved word identification. The sinusoidal model identified five primary spectral peaks in each 7.5-ms frame. The stimuli were then divided into five frequency bands, each containing one of the spectral peaks. The compression ratio for each frequency band was calculated from the dynamic ranges of the normal and hard-of-hearing (HoH) listeners at the frequency of the spectral peak in that band. The amount of gain for the HoH listeners was then determined for each band by multiplying the peak sensation level for normal listeners, and multiplying that by the compression ratio for the HoH listeners. Smoothing was applied between spectral bands to minimize distortion. The resulting processed speech had excellent sound quality and enhanced high-frequency spectral peak contrast. Processed and unprocessed sentences were presented to four listeners with mild (PTA = 40 dB HL) and four with moderate hearing loss (PTA = 80 dB HL). Differences between listener groups in amount of benefit from the processing will be presented.

Tolerable hearing-aid delays as a function of simulated hearing loss
Michael A. Stone and Brian C.J. Moore
University of Cambridge, England

When people who wear a hearing aid speak, there are three paths by which they hear their own voice; (1) through the aid and leakage around the ear-mould; (2) via the solid structures of their head; (3) through the air to the hearing aid microphone, and then
through the aid circuitry. These paths involve different time delays, which can give rise to disturbing echoes and interference effects. This situation is similar to that which arises when one hears echoes, such as in bathrooms, or other rooms with hard surfaces. Some signal processing algorithms proposed for use in hearing aids involve significant time delays, and these could lead to disturbing effects when the aid user talks.

With increasing hearing loss, the loudness of sound heard via paths (1) and (2) decreases and the aid-user relies more on aid-generated sound to monitor voice production. One might then expect the disturbance produced by the delay to be less perceptible. However, due to the increasing reliance on lip-reading cues with increasing hearing loss, longer delays are liable to disrupt audio-visual speech integration. We assessed the subjective annoyance as a function of the amount of delay, using a simulation of threshold elevation and loudness recruitment (Moore and Glasberg, 1993) to simulate four different hearing losses, ranging from mild to moderately severe. Each of two male talkers read a passage. One ear of the talker was fitted with a closed earmold, and simultaneous recordings were made at ear-level (where a hearing-aid microphone would be) and in the ear canal. The ear-level signal was amplified using four-channel fill dynamic range compression; compression ratios were selected by an algorithm based on the absolute thresholds assumed in the simulations of hearing loss. The insertion gains for 65 dB SPL speech were prescribed by the ‘Cambridge’ formula (Moore and Glasberg, 1998). The resultant output was then mixed with the in-ear signal with one of five values of delay (6, 13, 20, 30 and 40 ms) and the combined signal was processed using the four simulations of hearing loss.

20 normally hearing subjects were asked to give subjective ratings of the disturbance of the echo for each delay and each simulated hearing loss (presented in a counterbalanced order). The ratings generally increased on tonically with increasing delay. Average results show that delays are rated as ‘disturbing’ between 20 and 30 ms for mild-to-moderate losses, except for the moderately severe loss, where this rating was not quite achieved at 40 ms. For moderate losses, a speaker with low \( f_0 \) (70 – 140 Hz) was less disturbing than a speaker with a medium \( f_0 \), (100 – 180 Hz). This effect reversed for the mildest loss, for low values of delay.


**PA8**

**Two-channel adaptive noise cancellation for behind-the-ear hearing aids**

**Jeff Vanden Berghe and Jan Wouters**

Lab. Exp. ORL, Belgium

A number of techniques based on single and multi-microphone systems have already been applied to suppress unwanted background noise. Single microphone techniques generally perform poorly when the frequency spectra of the desired and the interfering sounds are similar, and when the spectrum of the interfering sound varies rapidly. By using more than 1 microphone, sounds can be sampled spatially and the direction of arrival can be used for discriminating desired from undesired...
signals. In this way it is possible to suppress stationary and non-stationary noise sources independently of their spectra.

We will report on the performance of a two-stage adaptive filtering strategy for noise suppression. It is implemented for a normal sized BTE hearing aid equipped with 2 identical front facing directional microphones, mounted in an endfire configuration. The intermicrophone distance is about 3 cm. The strategy was evaluated both off-line using physical criteria (improvement in signal-to-noise ratio) and tested perceptually using a real-time implementation with both normal hearing and moderately impaired listeners. A significant improvement, on average more than 5 dB, of speech reception threshold in background noise was obtained.

PB9
Two-channel compressed speech in noise: Effects of release time and compression ratio
Dianne van Tasell, and Chiquita Ewert, Starkey Laboratories, Inc.
Rachael Frush, University of Minnesota
William Woods

Static and dynamic parameters of compression can interact in complex ways to affect the sound quality of speech signals, especially in background noise. The complexity of these interactions presumably increases with number of compression channels; consequently, there currently is a lack of general agreement on how to select parameters of multichannel compression to optimize speech quality in noise.

In the work reported here, normal-hearing subjects provided rated preferences for pairs of two-channel compression-processed speech-in-noise stimuli at several different speech-to-noise (S/N) ratios. In each of the two processing channels, compression release time could take one of three values (30, 200, or 500 ms) and compression ratio could take one of three values (1, 2, or 10). Other compression parameters were held constant: attack time = 5 ms; compression threshold = 45 dB SPL; crossover frequency = 1500 Hz. Subjects provided preference ratings for all possible pairs (N=1176) of all possible compression processing combinations. Preference rating matrices were converted to similarity matrices and subjected to multi-dimensional scaling analyses. Results from a subset of stimuli showed a marked effect of signal-to-noise ratio: perceptual distances between stimuli from even the most disparate compression conditions decreased significantly with decreasing S/N ratio (that is, perceived quality of all compression-processed stimuli was similar at low S/N ratios, regardless of compression parameters used). Clustering of processed stimuli in perceptual space will be described, as well as implications of the data for selection of parameters for multichannel compression processing.

PB10
Preference judgments by normal-hearing and hearing-impaired listeners for hearing aid transduced speech, music, and everyday sounds
Nick J. Versfeld
University Hospital VU, The Netherlands

A listening experiment was conducted to assess the effect of the source signal on hearing-aid preference, and to investigate which type of signal is most suited to differentiate among hearing aids. Various sounds (male and female speech, speech in noise, different music fragments, and everyday sounds) were routed through twelve different hearing aids (mounted on a dummy head), and subsequently recorded. Hearing aid selection and adjustment was based on a typical high-frequency sensorineural hearing loss. The recordings
were presented to 24 normal-hearing and 24 hearing-impaired listeners in a pairwise-comparison format. The subjects' task was to indicate the hearing aid they would prefer assuming they had to wear it on a regular basis.

Between-subject differences were evident, so the data were analysed utilizing multidimensional scaling (MDS) techniques. Results show that, for both groups, music enables best differentiation between hearing aids. Clean speech also results in large differences for the hearing-impaired listeners, but not for the normally hearing. MDS shows that the principal factor is similar for all source signals and is associated with the low-frequency slope. Dependent on subject and signal, however, low-frequency emphasis is either preferred or disliked. The second factor often can be related to the high-frequency slope for normal-hearing listeners, and the frequency response near 2 kHz for hearing-impaired listeners. [Work supported by Philips Hearing Instruments BV, Eindhoven, The Netherlands.]

**PB11**
**Evident that mismatch negativity for speech sounds requires**
E. William Yund, Michael D. Szymanski, and David L. Woods
Veterans' Administration Medical Center

The mismatch negativity (MMN) component of the auditory event-related potential (ERP) has shown promise as an objective measure of the preattentive auditory information. Such an objective measure would be very useful in situations where more subjective measures might depend on the previous experience of the listener. Cochlear-implant or other hearing-aid patients, for example, may require considerable training or experience with a signal-processing algorithm before they are able to use the auditory information provided by the algorithm to perform discriminations or identify speech sounds. If the MMN were able to measure this potentially useful information, then it could be used to determine which algorithms transmitted critical information even before the patients had learned how to use it. Similarly, the MMN could guide the focus of training to those components of the information that were available to the individual patient. If the MMN measures attentive auditory processing, however, then it might have no special value over standard behavioral discrimination methods.

With the intention of developing the MMN as a tool to study hearing-aid signal-processing algorithms, we recorded ERPs to vowel-consonant-vowel (VCV) nonsense syllables randomly presented to the right and left ears in a demanding selective attention task. Normal-hearing subjects attended to one ear and detected intensity or phoneme changes in the consonant portion of the VCVs in that ear. In this paradigm, we measured the MMN for both attended and unattended deviant stimuli. The MMN for unattended deviants was about one half the amplitude of that for attended deviant stimuli. Furthermore, detailed analyses of the MMN for unattended stimuli suggested that it may also have been facilitated by attention because it was maximal for unattended target-type deviants and for unattended stimuli preceded by multiple attended stimuli. The unattended nontarget-type deviants that should have generated the maximum "preattentive" MMN, those preceded by multiple unattended stimuli, generated no MMN at all. These results indicate that the MMN requires attentive processing and suggests that the MMN will not be an appropriate tool for measuring...
preattentive auditory information in subjects with or without hearing impairment.
### Conference Attendees

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<td>Patrick Zurek</td>
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Names marked with an asterisk are recipients of student scholarships to attend the conference.
NOTES

1. All conference fees and other payments must be settled before the close of the conference.

2. Due to the altitude of Lake Arrowhead area, the conference center has no radio or television reception. Therefore, all conference sessions are held in the morning and evenings with free time for recreation, sports, sight seeing, etc., designated between the hours of 12noon and 5pm.

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