Issues in Advanced Hearing Aid Research

May 28 - June 1, 1990

UCLA Lake Arrowhead Conference Center
Lake Arrowhead, California

Conference Chairman
Sigfrid Soli, Ph.D.

Co-Chairman
Harry Levitt, Ph.D.

Sponsored by: House Ear Institute, Los Angeles, California
**DAILY CONFERENCE SCHEDULE**

**Monday, May 28**

3:30 p.m.  Check-in and conference registration
5:00 p.m.  Social
6:30 p.m.  Dinner
7:45 p.m.  Introductory Comments
8:00 p.m.  Evening session
10:00 p.m. Social

**Tuesday, Wednesday, Thursday, May 29, 30, 31**

8:00 a.m.  Breakfast
9:00 a.m.  Morning session
12:00 p.m. Lunch
1:00 p.m.  Free Time

8:00 a.m.  Breakfast  5:30 p.m.  Social
9:00 a.m.  Morning session  6:30 p.m.  Dinner
12:00 p.m. Lunch  7:45 p.m.  Evening session
1:00 p.m.  Free Time  10:00 p.m. Social

**Friday, June 1**

8:00 a.m.  Breakfast
9:00 a.m.  Morning session
12:00 p.m. Lunch
1:00 p.m.  Check-out; Buses and cars depart for airports

We extend special thanks to Pauline Davies, Publisher of Hearing Instruments, for help with the mailings for the conference.

** Invited Participants, please send records of travel expenses to Sig Soli:**

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I. Nature of Hearing Impairment
Psychophysics
Monday, Evening

Neal Viemeister, Discussant
Department of Psychology
University of Minnesota

7:45 p.m.

FREQUENCY RESOLUTION AND TEMPORAL RESOLUTION IN HEARING-IMPAIRED LISTENERS

David A. Nelson
Department of Otolaryngology
University of Minnesota

Results of recent experiments on frequency resolution and temporal resolution in hearing-impaired listeners will be presented. *Psychophysical tuning curves* (PTCs) obtained with simultaneous masking and non-simultaneous masking indicate that tuning in normal-hearing listeners at high levels is much broader than previously believed. Appropriate comparisons of high-level PTCs indicate that impaired ears do demonstrate abnormal frequency resolution, but not in terms of abnormal upward spread of masking. Impaired ears with abnormal tuning show abnormal downward spread of masking on the steep high-frequency sides of tuning curves and on the steep low-frequency sides of masking patterns. *Temporal masking curves* from normal and impaired ears indicate that time constants for recovery from short-term adaptation (forward masking) are only slightly longer in impaired ears, when comparisons are made at equal amounts of forward masking. However, because of the dependence of forward masking on the effectiveness of the masker, the rate of recovery from a high-level masker is much more gradual in the impaired ear, just as it is for a less effective masker in the normal ear. The interdependence of frequency and temporal resolution is seen in *masking patterns for fluctuating-envelope maskers*. With amplitude-modulated maskers, normal-hearing listeners demonstrate dramatically better temporal resolution when the test signal is higher in frequency than the masker. Cochlear hearing loss precludes the detection of cues available in periods of low masker energy, when the cues are at a frequency much above the masker frequency. These limitations imposed on auditory analysis by cochlear hearing loss have implications for amplified speech perception by hearing-impaired listeners.
825 p.m.

REPRESENTATION OF COMPLEX SOUNDS IN THE IMPAIRED AUDITORY SYSTEM

Marjorie Leek
Army Audiology and Speech Center
Walter Reed Army Medical Center

Decreased hearing sensitivity, reduced frequency selectivity, and altered temporal analysis all contribute to the difficulties in speech perception experienced by hearing-impaired individuals. Altered processing introduced at the level of the cochlea results in abnormal neural representations of incoming signals to higher centers in the auditory system. The research to be discussed in this presentation attempts to describe this abnormal "internal" representation for selected complex sounds that highlight particular aspects of cochlear processing. Listener responses on psychoacoustic tests of discrimination and identification of these stimuli are interpreted in the light of known deficits associated with hearing impairment to estimate the output of a damaged cochlea. Results from several studies of complex sound perception by hearing-impaired subjects will be discussed in terms of the internal representations generated by a model of cochlear function that simulates these processing deficits.

905 p.m.

COMODULATION MASKING RELEASE IN COCHLEAR-IMPAIRED LISTENERS

Joseph W. Hall
Division of Otology
University of North Carolina

In the phenomenon of comodulation masking release (CMR), the detection of a signal masked by a narrow band of noise centered on the signal is made more detectable by the presence of additional noise bands having the same fluctuation pattern as the on-signal noise band. This result suggests that the auditory system is able to extract signals from noise, based upon across-channel comparisons of temporal envelope information. Work will be described which attempts to relate this phenomenon to the difficulty which cochlear-impaired listeners have in processing signals in noise backgrounds. One of our interests at this time is to relate the magnitude of CMR in cochlear-impaired listeners to degree of sensitivity loss, frequency resolution, and temporal resolution. A second interest is to explore the ability of cochlear-impaired listeners to process a desired signal in the presence of several apparent competing sources, using a CMR paradigm. A third interest is to determine the possible relevance of CMR for speech signals, particularly with regard to cochlear-impaired listeners. Data (some preliminary) will be reported on the above three areas.
Models and Physiology
Tuesday, Morning

Jont Allen, Discussant
AT&T Bell Labs

9:00 a.m.

PSYCHOPHYSICAL MODELLING OF SPEECH INTELLIGIBILITY FOR THE HEARING IMPAIRED

Reinier Plomp
Department of Otolaryngology
Free University, Amsterdam, The Netherlands

The paper will review the main results of the development of a model in which the effect of hearing impairment on the speech-reception threshold is represented by two parameters: attenuation and threshold elevation in terms of signal-to-noise ratio, with three more parameters for the effect of the hearing aid. Further data will include the gain of binaural hearing and the case of having a competing voice, or the listener's own voice, rather than steady-state noise as the interfering sound. Together, these data give a useful picture of the difficulties of many hearing-impaired subjects in understanding speech in everyday conditions and can be used as guidelines in developing hearing aids.

9:40 a.m. - Break

9:50 a.m.

RESPONSES OF AUDITORY-NERVE FIBERS IN NORMAL AND IMPAIRED COCHLEAS

C. Daniel Geisler
Department of Neurophysiology and Department of Electrical and Computer Engineering
University of Wisconsin-Madison

The responses of normal auditory-nerve (AN) fibers to many kinds of complex acoustic stimuli, including speech, have been studied by a number of investigators. These studies will be briefly summarized, including the differences that have been observed between the responses of the different classes of AN fibers. Responses of fibers in impaired ears have been studied much less intensely and relatively little is known about their responses, especially to complex sounds. What is known will be reviewed and some new material presented. Supplementary to the sparse amount of actual data which exist will be models of AN fibers. These models will be used to give some further indications as to the expected nature of the responses of these fibers in impaired ears when stimulated by sounds such as speech in noisy backgrounds.
MODELLING THE USE OF PITCH-DIFFERENCES TO SEPARATE COMPETING SPEECH SOUNDS

Quentin Summerfield
MRC Institute of Hearing Research
University Park, Nottingham, England

Several groups in the UK and USA are exploring the extent to which the "autocorrelogram" of a sound is a useful representation for accounting for the ability of listeners to perceive concurrent pitches and to use them to segregate competing speech sources. Autocorrelograms can be computed by (i) filtering the stimulus with a bank of bandpass filters that simulate aspects of peripheral auditory frequency analysis, (ii) allowing the output of each filter to drive a model of the hair-cell synapse, such as the one described by Meddis (JASA, 1986,79,702-711), to simulate mechanical to neural transduction, (iii) computing the short-term autocorrelation function (ACF) of the resulting time-varying probability of neural discharge in each channel, and (iv) plotting the ACFs as a waterfall display. The "architecture" of the autocorrelogram can indicate the pitches that are present in the stimulus and the frequency regions over which they are defined, allowing competing voiced speech sources to be segregated. The possibility that an analogous representation is computed during auditory analysis will be assessed by considering recent psychoacoustical data on the perception of concurrent pitches, and physiological and psychoacoustical data on the neural representation and perceptual identification of concurrent pairs of synthetic vowels that differ in fundamental frequency. A video will be shown of time-varying autocorrelograms synchronized to their driving acoustical stimuli.

Special Populations
Tuesday, Evening

Arlene Carney, Discussant
Boys Town National Institute

7:45 p.m.

AMPLIFICATION SCHEMES FOR THE PROFOUNDLY HEARING IMPAIRED

Arthur Boothroyd
Graduate School and University Center
City University of New York

The profoundly deaf are characterized by: a) high threshold (in excess of 90 dBHL); b) restricted amplitude range (typically less than 30 dB); c) poor frequency and time resolution (leading to poor speech discrimination and extreme susceptibility to the interfering effects of noise); and d) sometimes, a restricted frequency range. Currently available solutions to the problems of providing effective amplification include: a) feedback reduction (to permit very high gains); b) automatic gain control (to minimize long-term variations of average speech intensity); c) high-frequency emphasis (to reduce the effective short-term dynamic range of intensity in the speech signal and to increase
overall signal-to-noise ratio; d) remote, close-talking microphones (to reduce gain requirements and enhance signal-to-noise ratio); and e) fast-release amplitude compression (to reduce the short-term dynamic intensity range of the speech signal). Future possibilities for improved performance include: a) digital feedback control; b) acoustic cue enhancement; c) digital noise suppression; and d) acoustic recoding. Important points to remember include: a) although maximal use of limited auditory capacity is a reasonable goal, it is not reasonable to expect acoustic modifications to compensate fully for a seriously compromised sensorineural system; b) superficially obvious solutions such as amplitude compression and frequency-shifting may turn out to be counterproductive in this population; and c) digital signal processing techniques have yet to prove their worth.

8:25 p.m.

INTEGRATING AUDITORY AND VISUAL CUES IN THE IDENTIFICATION OF SPEECH SEGMENTS

Louis D. Braida
Research Laboratory of Electronics
Massachusetts Institute of Technology

Auditory presentation of low-frequency speech components, and tactile or electrocochlear stimulation derived from speech, can improve the speechreading of speech segments. A method is described for predicting both the level of aided performance and the resulting error pattern from the confusion matrix for each modality separately. The method is based on a characterization of confusion matrices in terms of a multi-dimensional Thurstonian decision model that allows performance to be described in terms of sensitivity and bias. In the multimodal case, the decision space is assumed to be the product space of the decision spaces corresponding to the stimulation modes. In certain cases, multimodal sensitivity is roughly equal to the vector sum of the sensitivities for each stimulation mode, indicating that cues are integrated with little interference.

9:05 p.m.

SPECIAL NEEDS FOR THE GERIATRIC POPULATION

Brad Stach
Baylor University

Although many elderly patients with hearing impairment benefit substantially from conventional hearing aid amplification, only a small minority actually seeks hearing aid use. Unfortunately, of those who do, many do not benefit as much as might be expected. One of the major factors that adversely affects hearing aid benefit is the complex nature of presbyacusis and its effect on speech understanding. Other factors include the sound quality of hearing aids, physical limitations of the elderly population, and psychosocial influences. Recent data suggest that speech understanding deficits, which are specifically-auditory in nature and resemble central auditory processing disorder, are prevalent in the elderly population. These deficits have been related to increased hearing handicap, decreased hearing aid benefit, and a relative advantage in binaural listening. Alternative amplification strategies that enhance the S/N can be used very effectively to circumvent many of the problems associated with conventional hearing aid use by the elderly.
II. Design of Devices to Remediate Impairment

Amplification and Compression

Wednesday, Morning

Jerry Studebaker, Discussant
Memphis State University

9:00 a.m.

FREQUENCY-GAIN SHAPING FOR HEARING AIDS

Donald Dirks
School of Medicine, Division of Head and Neck Surgery
University of California, Los Angeles

The purpose of this experimental program is to determine critical variables which affect the choice of frequency-gain characteristics preferred by individuals with hearing impairment, and to investigate the potential use of adaptive test strategies for use within the hearing aid selection process. Variations of the simplex adaptive test strategy, incorporating paired-comparison judgements, were utilized in these initial experiments. A computer-based system was used to control the stimulus and the measurement procedures. Frequency shaping was accomplished digitally via a filter bank of five filters which could be use in a variety of combinations to provide different sets of frequency responses. The composite filter bands provided a frequency range extending from 200 Hz to 5800 Hz and were designed for a 13.79 kHz sampling rate. The realized bandwidth, however, was limited in its acoustic output to approximately 4500 Hz.

In the initial study, comparisons were made between the preferred frequency-gain characteristics chosen with a two-band system (formed by combining three filters for the lower band, 200 to 1200 Hz, and two filters for the upper band, 1200 Hz to 4500 Hz) and those chosen with a three-band system (a low frequency band, 200-600 Hz; a mid-frequency band, 600-1800 Hz; and a high-frequency band, 1800 Hz-4500 Hz). The computer system was used to control two and three dimensional simplex test strategies as well as to collect and tabulate the subject responses. Recorded connected discourse was used as the test stimulus.

For both the two (5x5 matrix) and three (5x5x5 matrix) dimensional strategies, the center cell of the matrix (initial frequency-gain setting) was programmed to provide the insertion gain predicted by the revised National Acoustic Lab (NAL) prescription scheme. The remaining cells in the matrix varied systematically: the characteristics in the center cell over a 24 dB range in plus or minus 6 dB steps. A probe microphone system, utilizing the acoustic method of Chan and Geisler (J.A.S.A. 1990) was used to localize the probe within 5-6 mm of the eardrum so that the actual insertion gain could be measured on each subject prior to initiating the experimental adaptive procedure. In most instances, the probe measured gain varied slightly from the target gain (as measured in a coupler) especially in the high frequencies. In order to achieve the NAL insertion gain, it was often necessary to modify the acoustic coupling by use of dampers to smooth or reduce the mid-frequencies and/or various horn options for enhancement of the high frequencies.
The results from ten hearing-impaired subjects indicated first, that both the two and three dimensional simplex procedures provided reliable test-retest preference judgements; second, similar preference judgments were obtained from both strategies, although several subjects with steeply-sloping high-frequency hearing loss consistently preferred more gain in the mid-frequency region, selectively available only in the three dimensional procedure; third, the averaged preferred frequency-gain characteristics of the group were similar to the NAL predicted insertion gain; and fourth, results of the probe measurements indicated occasional major deviations in the higher frequencies between the target insertion gain response (in the center cell) and the probe-measured acoustic response at the eardrum.

9:40 a.m.

NEW METHODS OF AMPLITUDE COMPRESSION

Harry Levitt
Graduate School and University Center
City University of New York

Digital signal processing techniques open up new possibilities for amplitude compression in hearing aids. One such possibility is a generalized form of compression in which the frequency-gain characteristic is determined by both the level and shape of the speech spectrum. Similar techniques operating on the speech-plus-noise spectrum can also be used for noise reduction. Computationally concise techniques for estimating spectrum shape in real-time have been developed. Despite promising predictions from a theoretical perspective, experimental evaluations of this generalized approach have shown small improvements over conventional methods of compression. A very useful feature of the generalized approach is the flexibility afforded to the experimenter in selecting and adjusting compression variables. As such, these techniques offer the means for investigating models of how amplitude compressed signals are processed by hearing-impaired persons. Improved models are clearly needed in order to maximize the improvements to be obtained from amplitude compression in the future generation of hearing aids.

10:20 a.m. - Break

10:30 a.m.

OPTIMIZATION AND EVALUATION OF TWO-BAND COMPRESSION SYSTEMS

Brian C.J. Moore
Department of Experimental Psychology, University of Cambridge, England
Department of Psychology, University of California, Berkeley

This talk will start by considering some of the rationales which have been given for using multi-band compression in hearing aids. These include: restoring the loudness relations among the components of complex sounds to what they would be for a normal ear; reducing the effects of narrowband interfering sounds; and ensuring that all of the important acoustic elements in speech are audible and comfortably loud. While all of these may have some value, I will argue that the third one is of primary importance. Our
philosophy has been to try to achieve that third goal while distorting the speech signal as little as possible in the spectral and temporal domain.

Results will be described using a prototype laboratory system which includes two different types of automatic gain control (AGC). The first is a slow-acting AGC operating on the whole signal, which compensates for variations in the overall level of speech from one situation to another. It includes a means for rapidly reducing the gain for brief intense transients, without affecting the long-term gain. This system is followed by a two-channel system with the option of having fast-acting compression in each band. Experiments will be described that determine optimum values for the time constants of the AGC in the different stages.

Preliminary results will also be presented of an evaluation of a two-channel aid manufactured by Resound Corporation. Unlike our laboratory prototype, which uses compression limiting, the Resound aid uses full dynamic range compression, with programmable compression ratios and gains in each band. Results will be presented of measures of speech intelligibility in quiet and in noise, under three listening conditions: unaided; with the Resound aid programmed as a linear aid; and with the Resound aid programmed as a two-channel compression aid.

11:10 a.m.

MULTIBAND COMPRESSION IN HEARING AIDS

Edgar Villing
Foundation for Hearing Aid Research

Coker called speech an error-resistant code because it has redundant cues that provide a reserve against the loss of cues masked by noise and other acoustic interference. This reserve makes it possible for normal listeners to understand speech in noisy environments. Evidence from experiments with processed speech suggests that the special vulnerability of the hearing impaired to interference is not an independent feature of the impairment but derives from a loss of these redundant cues. Multichannel compression in hearing aids is designed to restore lost speech cues, including redundant cues, to the impaired listener's perception. Compression also restores the audibility of some of the interference that has been suppressed by recruitment—interference that would remain below the listeners threshold with linear amplification—but multichannel compression has produced a substantial net improvement of speech intelligibility in the presence of interference.
Some performance problems of existing hearing aids include degradation of 
speech clarity and poor sound quality in high levels of environmental noise. High levels 
of background noise can saturate hearing aids, creating harmonic and intermodulation 
distortion so as to form additional masking components. This problem is caused in part 
by inadequate "headroom" resulting from low battery voltage and current requirements 
and small hearing aid receivers which saturate at low sound pressure levels. Another 
major problem is acoustic feedback instability which causes poor transient response and 
oscillation producing an annoying squealing sound.

Small hearing aid package sizes prevent use of conventional electronic 
technology to solve these problems. The hearing aid market in the United States is 
currently dominated by concern for cosmetics: in-the-ear and in-the-canal hearing aids 
represent almost 80% of sales. Some of the techniques that are packageable in these small 
hearing aids employed recently to alleviate the noise masking and saturation problems 
are: directional microphones, single-channel AGC, fixed and adaptive high pass filtering 
and multi-channel AGC circuits. The problem with these approaches to date are that 
most don't reduce background noise or keep hearing aids from saturating well enough.

Partially digital hearing aids have recently emerged in the form of analog 
amplification and filtering under digital control, in some cases with programmable 
digital memory. While some of these programmable devices are little more than 
"electronic screwdrivers" replacing mechanical trimmers, others offer significantly new 
fitting methodologies. Although there has been quite a bit of work (and publicity) on 
digital hearing aids formulated in the laboratory, up to now, there has only been one 
wearable hearing aid marketed that is "all digital".

Future technology will include higher order analog filtering, more multichannel 
analog circuits for better control of frequency response shaping, dynamic range and 
feedback reduction; custom digital signal processing computer chips to produce cosmetic 
"all-digital" hearing aids, 2-or more-piece hearing aids with wireless transmission in 
between, adaptive beamforming multi-channel arrays, and deliberate distortion of some 
components of speech to accentuate weak cues.
8:25 p.m.

IN SEARCH OF THE HIGH FIDELITY HEARING AID

Mead Killion
Etymotic Research

For some time, the writer and colleagues have been pursuing a practical hearing aid design that would be perceived as high fidelity by those with mid-to-moderate hearing loss. Some of the problems appear solved. Available hearing aid transducers and "acoustic plumbing" techniques permit 40-Hz to 16-kHz reproduction in hearing aids with fidelity comparable to that of professional studio monitors (Killion, 1978). A class D output amplifier design for Knowles Electronics, incorporated in their EP-series receivers, has largely circumvented the tradeoff between low battery drain and good "headroom" at high frequencies: Low battery drain and low high-frequency intermodulation distortion can now be obtained simultaneously without adding to the size of the hearing aid. Most recently, a new integrated-circuit "K-AMP" hearing-aid amplifier chip with level-dependent high-frequency emphasis shows promise of nearly "transparent" operation for the wearer. The addition of adaptive compression circuitry under license from Telex has helped circumvent the dilemma that no single time constant appears adequate for all situations. With adaptive compression the effective recovery time for short transients can be nearly 100 times faster than the recovery time for long-duration stimuli. Remaining problems will be discussed.

9:05 p.m.

VLSI DESIGNS FOR EAR LEVEL DIGITAL SIGNAL PROCESSING

Robert Morley
Department of Electrical Engineering and Computer Science
Washington University

Hearing aid features, requested by audiologists on the basis of past experiences in clinical practice and research, are in excess of those provided by any aid to date. Digital signal processing (DSP) techniques may be incorporated to bridge this gap. However, the power consumption of general purpose digital signal processors exceeds the power budget of a conventional hearing aid by several orders of magnitude.

A two-chip, custom VLSI design is proposed wherein one chip is responsible for data acquisition and reconstruction while a second chip is dedicated to the DSP circuitry. A low power, custom digital signal processor, capable of performing over 4 million multiply accumulate operations per second is presented in this paper. Power consumption is minimized while maintaining a wide dynamic range through the use of logarithmic arithmetic.
Noise Reduction and Signal Enhancement

Thursday, Morning

Sigfrid Soli, Discussant
House Ear Institute

9:00 a.m.

DIGITAL SPEECH ENHANCEMENT IN THE CONTEXT OF A HUMAN AUDITORY MODEL

Douglas Chabries
Department of Electrical Engineering
Brigham Young University

This paper presents the application of digital signal processing to the enhancement of speech signals corrupted by noise. Two problems are presented: first, adaptive digital filters are applied to the problem of speech corrupted by noise when a sample of the interference is available; and second, the removal of noise from speech is shown for this same problem when no sample of the interference may be obtained. In the two channel case, gains of up to 45 dB in signal-to-noise ratio are demonstrated with a concurrent increase of up to 45% in speech intelligibility as measured with CID word lists. The application of speech enhancement for single channel applications is demonstrated for several well known algorithms. The performance of these algorithms degrades intelligibility when the noise is removed; however, a digital model of the processing of the human auditory system used in conjunction with these same well known speech enhancement algorithms provides increased intelligibility as measured with CID word lists. Details of the digital model of the auditory system and its application to speech enhancement are presented. Several applications of the hearing model for improving speech quality and enhancing intelligibility will be presented.

9:40 a.m.

FEEDBACK CONTROL IN HEARING AIDS

James Kates
Graduate School and University Center
City University of New York

One of the factors that limits hearing-aid performance is feedback. This presentation discusses the problem of feedback in hearing aids, illustrated with examples based on a computer simulation of hearing-aid behavior. A new adaptive procedure for controlling feedback, based on estimating the feedback signal and subtracting it from the microphone input signal, is then described. In this feedback-cancellation system, the characteristics of the feedback path are estimated whenever changes are detected in the feedback behavior, and a set of filter coefficients is adjusted to provide an equivalent electrical path having the opposite polarity. The hearing aid is then returned to normal operation with the feedback-cancellation filter as part of the system. Simulation results indicate that more than 10 dB of cancellation can be obtained with this approach.

10:20 a.m. – Break
CONSTRANDED ADAPTIVE SPEECH ENHANCEMENT

Kevin M. Buckley
Department of Electrical Engineering
University of Minnesota

The discussion addresses speech enhancement, the process of removing undesired signal components from a desired speech signal, as applied to multi-microphone digital hearing aid signals to improve intelligibility and quality.

We present results from an ongoing investigation into a constrained optimization approach to speech enhancement for hearing aids. Optimization provides the mechanism for reducing the effect of undesired signals on the hearing aid output. We are currently primarily considering power minimization as an optimization cost function, since simple adaptive processors follow. However, the discussion will begin with a more general formulation which encompasses other costs which more directly relate to hearing aid objectives. Constraints on the processor’s spatial and temporal response provide control over the distortion of the desired speech signal. Linear equality constraints on the processor’s filtering coefficients are most commonly considered. Several nonlinear and inequality constraint approaches have recently been proposed for speech applications which are motivated to provide flexible control of the speech distortion/noise-reduction tradeoff. These approaches, based on power minimization, are actually very similar in formulation.

In this discussion the issues of relative performance and efficient adaptive implementation of these approaches are addressed. We include consideration of a multi-microphone speech enhancement processing approach that we have recently proposed which is based on: adaptive power minimization; quadratic inequality constraints which allow controlled desired-speech distortion; a linear equality constraint which fixes the quiescent response to be that of an optimum data independent beamformer; and a computationally simple implementational structure.

EXPERIMENTS IN SPEECH ENHANCEMENT WITH THE HEARING IMPAIRED AND DEAF

Sally Revoile
Center for Auditory and Speech Sciences
Gallaudet University

Acoustic-cue enhancement to facilitate consonant perception has been studied for more than 125 listeners categorized as moderately, severely, or profoundly hearing impaired. Cue enhancements have included amplification, duration modifications, or spectral alterations of segments associated with consonants in spoken speech. The consonants were located intervocally in syllables extracted from connected speech, or in word-initial or final position in citation-style stimuli. Given enhancements of particular consonants were tested with different sets of listeners. Especially for listeners with severe or greater impairment, improved perception from some cue enhancements has been seen for performance indices representing consonant voicing or manner
III. Clinical Research Issues

Fitting and Validation

Thursday, Evening

Harry Levitt, Discussant
Graduate School and University Center
City University of New York

7:45 p.m.

ISSUES IN BINAURAL AMPLIFICATION

Janet Koehnke
Department of Communicative Disorders
Boston University

Despite the large research effort comparing the benefits of monaural and binaural amplification, a method for determining the optimal hearing aid configuration for individual listeners has not yet emerged. Data obtained from a survey study of binaural performance in hearing-impaired listeners has revealed a number of results pertinent to this issue. Specifically, we have found that (1) binaural performance in hearing-impaired listeners is usually (but not always) poorer than normal, (2) binaural performance cannot be predicted simply from the audiogram, (3) even in regions where audiometric thresholds are normal, binaural performance is not necessarily normal, and (4) compensating for interaural threshold asymmetries does not necessarily improve binaural performance. In view of these results and other experimental data obtained from hearing-impaired listeners as well as theoretical evidence suggesting that speech intelligibility may be predictable from measures of binaural detection, we have developed a project design to compare various amplification configurations on tests of binaural detection, speech intelligibility, and sound localization. Results on the detection tests show that some individuals obtain fairly large MLDs (15 dB) with binaural amplification, while others have small or no MLDs regardless of amplification configuration. Contralateral interference measures indicate that some subjects are able to listen with their more-favorably placed ear (nearer the signal) even in the presence of high-level noise, but others have more difficulty detecting signals with noise present in the contralateral ear. These results indicate that binaural detection data are likely to provide information that may be helpful in selecting the most appropriate amplification configuration.
OBJECTIVE VERSUS SUBJECTIVE EVALUATION OF SPEECH INTELLIGIBILITY AND QUALITY

Chaslav V. Pavlovic
Speech and Hearing Center
University of Iowa

The number of possible combinations of sets of parameter values in programmable auditory prostheses (hearing aids, vibrotactile aids, or cochlear implants) is virtually infinite. With some reasonable reductions in the number of discrete states that each parameter can take, this amount could be substantially reduced. Still, the number of alternatives is easily in the hundreds of thousands.

With this many alternatives the question arises as to which psychophysical procedures should be used to evaluate or predict users' satisfaction with various processing schemes. In this talk the author's current or recent research results will be presented that address various validity, reliability, sensitivity, and efficiency issues of different "objective" and "subjective" evaluation procedures. In addition, an attempt will be made to clarify the relationship among various speech quality attributes (satisfaction, acceptability, intelligibility, pleasantness, clarity, noisiness, etc.). Ways to evaluate these attributes will be discussed also.

SPECIAL CONSIDERATIONS IN PEDIATRIC POPULATIONS

Patricia G. Stelmachowicz
Boys Town National Institute

The task of fitting hearing aids to young children is often complicated by the limited availability of audiological data, reduced test reliability, and the inability to utilize subjective measures in the fitting process. Newly-developed technologies that have expanded fitting options for adults may not necessarily be appropriate for the pediatric population. At the present time, the audibility of speech and the aided dynamic range remain the most critical issues when fitting young children with amplification. The purpose of this paper is to report a series of studies related to these issues and describe a technique that can be used to quantify the aided dynamic range on an individual basis.
ASSESSING HEARING AID BENEFIT: CONCEPTUAL AND METHODOLOGICAL ISSUES

Marilyn Demorest
Department of Psychology
University of Maryland, Baltimore County

Advances in hearing aid technology have progressed more rapidly than our ability to evaluate their benefits. Hearing aid performance can be evaluated in at least three ways: (1) by evaluating the hearing aid itself; (2) by evaluating perception and communication performance of individuals wearing the hearing aid; and (3) by obtaining information from hearing aid users about the performance of their hearing aid in everyday life. The first approach emphasizes the signal processing aspects of hearing aid fitting and evaluation. It can be used to determine whether a particular hearing aid functions according to design, whether it conforms to specifications, whether real-ear measures conform to a particular prescription, and so on. The second approach is behavioral and typically employs clinical tests of speech perception. Such test have the advantage of directly sampling the perceptual domain that hearing aids are designed to enhance, yet their limitations have led to widespread concern about their reliability and the generalizability of test scores to performance in everyday life. In principal, however, there is no reason why more valid clinical tests cannot be developed. Test materials can include sentences or connected discourse, listening conditions can include noise and reverberation, and the speech signal can be presented in both audio and audiovisual modes. It is likely, however, that the more valid the test, the more it will reflect factors that are not strictly auditory. Because of the difficulties of evaluating hearing aid performance in the clinic or laboratory, the third approach, self-report, has been widely used to obtain information about hearing aid performance in daily life. The self-report method has well-known limitations, but well-constructed instruments provide information that cannot feasibly be obtained in any other way. Because each of the three approaches provides a different, yet conceptually valid, perspective on hearing aid benefit, a comprehensive evaluation should include all three.

At a conceptual level, it is important to distinguish between hearing aid benefit and other similar concepts such as hearing aid use, satisfaction, and success. Benefit and use are constructs that can be assessed either behaviorally or via self-report, and they can be measured either directly or indirectly. Hearing aid satisfaction is an inherently subjective evaluation made by the client and it is likely to be influenced by psychosocial variables that are not under the control of the hearing healthcare professional. Interpretation of the existing literature on hearing aid benefit is made difficult by recent failure to carefully distinguish among these concepts. It is also important to distinguish between evaluation of population or group trends and assessment of individuals. The former requires reasonably large representative samples of individuals from the target population, the latter requires an adequate number of observations (in the form of test or questionnaire items) on a given individual.
THE RELATIONSHIP OF SPEECH GAIN TO INSERTION GAIN OF HEARING AIDS

Harvey Dillon
National Acoustic Laboratories
New South Wales, Australia

Hearing aid fittings can be evaluated in a number of ways. These include using informal or formal questionnaires, paired comparisons of alternative fittings, measurements of insertion gain and comparison to target values calculated by a prescription method, and measurement of speech perception ability with and without the hearing aid in noise or in quiet. This paper quantifies the relationship between two of these: insertion gain and speech gain in the absence of intentionally added masking noise. Speech gain is defined here as the sideways shift in the speech identification performance-intensity (PI) function that results when the aid is worn. That is, the amount relative to unaided listening by which the signal level can be decreased when the aid is worn, without affecting performance on a speech identification test. The amount of speech gain depends on the type of speech material used, and also depends on the presentation level chosen for the unaided testing. The presentation will show how the articulation index (AI) model can be used to predict the aided PI function from the unaided PI function, and hence to predict the speech gain. In addition to the insertion gain, other factors which determine the speech gain are internal aid noise, the subject’s unaided free field thresholds, and the spectral shape of the speech signal used. All of these factors can be readily incorporated in the AI method of predicting the speech gain of hearing aids.

10:20 a.m. - Break

10:30 a.m.

EVALUATION OF HEARING AID PERFORMANCE IN THE LABORATORY AND IN DAILY LIFE

Robyn M. Cox
Department of Audiology and Speech Pathology
Memphis State University

This presentation will describe recent research in measurement and prediction of hearing aid performance and benefit. This research is supported by the Veterans Administration and the overall goal of the program is to develop validated methods for prediction of hearing aid benefit. Research questions to date have centered on the amount of benefit delivered by linear, monaural, non-directional, earlevel hearing aids. We will discuss the relationship between objective and subjective measures of benefit; the effects of three typical daily listening environments on benefit; the effects of frequency response and visual cues on benefit; and the use of response time to speech stimuli as a measure of benefit. Two questionnaires have been developed and evaluated for measurement of benefit in daily life. Preliminary data exploring the use of a new “loudness bother” scale for SSPL90 prescription will be presented.
EVALUATION OF ADAPTIVE FREQUENCY SHAPING IN THE CLINIC AND LABORATORY

Diane Van Tasell
Department of Communication Disorders
University of Minnesota

Most wearable "noise-reduction" hearing aids operate by reducing low-frequency gain in the presence of noise; we refer to these devices as "adaptive frequency response" (AFR) hearing aids. AFR hearing aids seldom enhance recognition of broadband speech in broadband noise. They can, however, improve recognition of high-frequency speech information in low-frequency narrowband noise. Aided speech recognition threshold (SRT) for a set of high-frequency loaded monosyllabic words was measured in low-frequency (600-800 Hz) narrowband noise with subjects wearing a master hearing aid operating in either AFR or non-AFR mode. In noise levels high enough to achieve the AFR function, SRTs were lower for the AFR than for the non-AFR condition. We sought to predict the extent of the SRT improvement by comparing aided masking patterns for the narrowband noise obtained with the hearing aid in the AFR versus the non-AFR mode. In the AFR mode masked thresholds were lower at frequencies above the masker than they were in the non-AFR mode. In the non-AFR mode, however, the master hearing aid apparently was being driven into saturation by the 70-dB noise; mid-frequency distortion products were observed in the noise spectrum measured in the ear canal with the hearing aid in place. It is possible that the improved speech intelligibility and sound quality sometimes reported for AFR hearing aids result from the reduction of distortion rather than from improvement in S/N ratio. Masking data obtained under laboratory conditions of simulated AFR will also be reported, as well as distortion measurements on commercially-available hearing aids.
CONTRIBUTED POSTER ABSTRACTS

SINGLE PHOTON EMISSION COMPUTED TOMOGRAPHY (SPECT): A TECHNIQUE TO MEASURE AUDITORY-RELATED CHANGES IN REGIONAL CEREBRAL BLOOD FLOW (rCBF)

Chie H. Craig, Ronald S. Tikofsky, Robert S. Hellman, James A. Bashford, Raymond G. Hoffman
University of Wisconsin

Newly developed functional brain imaging techniques are potentially effective tools for the noninvasive study of the brain's response to cognitive stimulation. This poster introduces an original application of Single Photon Emission Computed Tomography (SPECT) developed at The University of Wisconsin-Milwaukee and The Medical College of Wisconsin. This SPECT application is part of an on-going collaborative research program directed toward a better understanding of: 1) basic human cerebral function and central auditory processing activities and, 2) the feasibility of employing quantitative SPECT rCBF measures to study the effects of different types of auditory stimulation. Specific aspects of the SPECT methodology and statistical analyses are described. Preliminary studies of SPECT rCBF measures using normal hearing listeners and a speech understanding task based on contrasting levels of spoken word predictability are presented. Future research implications are discussed.

CAN METHODS FROM RESEARCH IN SEMI-AUTOMATIC LABEL ALIGNMENT OF LARGE SPEECH DATABASES BE APPLIED IN CODING ACOUSTIC-PHONETIC INFORMATION FOR LIP-READING?

Paul Dalsgaard
University of Aalborg

One approach to perform semi-automatic labelling of large corpora of speech materials is to apply a self-organizing network as a preprocessor for an alignment process, which is then carried out by Viterbi decoding. The technique uses acoustic-phonetic features in its processing, and it is now interesting to discuss whether part of the acoustic-phonetic features derived with this technique can also be applied in assisting profoundly deaf people in lip-reading.

To be used with semi-automatic labelling, the neural network is trained in a self-organizing stimulation and calibration process. Here input data are taken from a manually labelled reference speech database consisting of four passages of natural speech, each passage being of approximately two minutes duration. After training each neuron in the neural network is further associated with a vector of numbers specifying the probabilities that specific acoustic-phonetic features are present in the speech analysis frame.

The goal of involving a neural network in the alignment process, is to use this in the process of transforming speech analysis cepstral coefficients into a continuous valued acoustic-phonetic feature vector valid for each speech frame.
In the task of semi-automatic labelling, the goal is to identify a phonetic feature vector covering an entire speech segment. This is performed on the basis of frame-based acoustic-phonetic feature vectors identified by the neural network being subjected to a Viterbi search. The search is constrained by an independent string of phonemes e.g. given in broad auditory notation - given with the speech signal.

Results from applying the semi-automatic label alignment technique to unknown speech material will be presented. Histograms are given demonstrating the ability of the alignment technique to correctly position the segment boundaries. The accuracy of the segment boundaries are given for pair of phoneme classes, where the manually positioned segment boundaries from the test database is used for referencing.

It is hoped that during the poster presentation the technique presented here can be used as a basis for discussing the potential use of the acoustic-phonetic features for coding information, which are necessary for lip-reading. Also, it might be relevant to discuss different ways to present this information optimally to users.

**PAIRED-COMPARISON JUDGMENTS FOR HEARING AID SELECTION IN CHILDREN**

Laurie Eisenberg  Harry Levitt  
House Ear Institute  City University of New York

The paired-comparison technique uses the listener's judgments of preference to select a hearing aid. Hearing aid conditions are paired systematically and evaluated until one condition emerges as the preferred choice. The paired-comparison technique has been shown to be a reliable and efficient method for selecting hearing aids for adults. This study investigates the feasibility of using a paired-comparison technique with children. Two experiments determine the age at which normal-hearing and hearing-impaired children can make paired-comparison judgments of auditory clarity. A third experiment investigates the effectiveness of the paired-comparison technique to select appropriate hearing aids for hearing-impaired children. Preliminary results will be presented.

**DO "ADAPTIVE FREQUENCY RESPONSE" (AFR) HEARING AIDS REDUCE UPWARD SPREAD OF MASKING?**

David A. Fabry  
Walter Reed Army Medical Center

Speech recognition scores in noise are improved for some subjects wearing hearing aids that reduce low-frequency noise with an adjustable high-pass filter circuit. Fabry and Van Tasell (1989) argued that because speech-to-noise ratios are not altered by AFR hearing aids, improvements in speech recognition may be related to a reduction in upward spread of masking experienced by the listener (D. Fabry & D. Van Tasell, JASA, 85, S25). To evaluate this hypothesis, pure-tone masking patterns for a low-frequency band-pass noise were measured in normal and hearing-impaired subjects. The noise skirt of the masker was very steep, with attenuation above the 1000 Hz cutoff greater than 100 dB per octave. Masking patterns for the same noise were also obtained in the presence of
a high-pass filter that stimulated the effects of an AFR hearing aid. Difference in the
masking patterns can be taken as a measure of upward spread of masking. Subjects with
high frequency hearing loss tended to demonstrate greater amounts of upward spread of
masking than did normal-hearing listeners. Further, monosyllabic speech recognition in
noise testing indicated improvements in performance of the hearing-impaired subjects
related to the decrease of upward spread of masking in the high-pass filtering conditions.

RESIDUAL FREQUENCY SELECTIVITY IN THE PROFOUNDLY HEARING-IMPAIRED
LISTENER

Andrew Faulkner, Stuart Rosen, and Brian C.J. Moore
Department of Phonetics and Linguistics
University College London

The extent to which auditory frequency analysis is retained in profoundly
hearing-impaired listeners has major implications for hearing aid design. We have
measured simplified psychoacoustic tuning curves in 9 such listeners, using sinusoidal
probes at 125 and 250 Hz, and 80-Hz wide narrow-band noise maskers. Two listeners
showed PTCs whose shapes were independent of probe frequency and parallel to their
absolute thresholds, indicating the complete absence of frequency selectivity. Seven
listeners showed evidence of frequency selectivity at either 125 Hz or 250 Hz. Estimated 3
dB bandwidths were two to three times larger than those typically found in normal
listeners. Notched-noise masking results at 250 Hz from the least hearing-impaired
listener gave an estimated 3 dB bandwidth in reasonable agreement with that from the
same listener's PTC data. Speech-related spectral discrimination performance from one
listener indicated that residual frequency selectivity could be of substantial importance
in the development of new acoustic hearing aids based upon speech pattern elements.

SPEECH PATTERN HEARING AIDS FOR THE PROFOUNDLY HEARING-IMPAIRED
LISTENER: SPEECH PERCEPTION AND AUDITORY ABILITIES

Andrew Faulkner, Virginia Ball, Adrian J. Fourcin, Brian C.J. Moore,
Stuart Rosen, and John R. Walliker
Department of Phonetics and Linguistics
University College London

Auditory speech pattern aids were compared to amplification using thirteen
profoundly deafened adults, for whom auditory area, frequency discrimination, gap
detection, and frequency selectivity were also assessed. The SiVo aid (Rosen et al., J.
Rehab. Res. Devt. 24, 239-260, 1987), which presents a simple voice fundamental frequency
pattern, provided three subjects with significantly more voicing information than speech
presented through an amplifying aid with an extended low-frequency response. The
reception of voicing was similar for both aids in eight subjects, while two subjects could
use neither aid. Three subjects extracted more manner information from amplified
speech than from the SiVo aid. Intonation reception was never worse and often better
with the SiVo aid, particularly when frequency discrimination was poor. The two
subjects who benefited most from the SiVo aid showed no evidence of frequency
selectivity; limited dynamic range at 500 Hz and above also favoured the SiVo aid. Five
subjects were tested using a compound speech pattern aid, which conveyed information about the amplitude envelope of the speech, and signalled the presence of frication with a burst of low-frequency noise. Three subjects received more information from this aid than from amplified speech, and all five received more from the compound pattern aid than from the fundamental frequency pattern. Compound speech pattern aids of this type may prove to be more effective for the profoundly hearing impaired than either conventional amplifying aids or cochlear implants.

MULTICHANNEL SIGNAL AMPLIFICATION CONTROLLED BY THE SPECTRO-TEMPORAL PROPERTIES OF NOISE

Joost M. Festen, Janette N. van Dijkhuizen, and Reiner Plomp
Free University, Amsterdam

Limited dynamic range is a major problem in sensorineural hearing impairment. With variations in environmental noise, a talker automatically adapts his/her speech level so that in general both levels covary. To compensate for these temporally and spectrally varying levels of speech and noise, an impaired ear needs frequency dependent adaptive amplification. It is shown that according to Articulation Theory the improvement of intelligibility by selective attenuation of noisy low-frequency bands is maximally 75 dB. A four-channel signal processing scheme is proposed that estimates the noise level per frequency band and adapts the gain. The speech-reception thresholds in noise for hearing-impaired listeners obtained with such a four-channel adaptive gain control are up to 4 dB better than the threshold obtained with wideband gain adjustment.

THE EFFECT OF TACTILE AIDS ON COMMUNICATION SKILLS OF CHILDREN WITH DUAL SENSORY DEFICITS

Barbara Franklin
San Francisco State University

This paper will present the results of a 3-year study (86-89) to investigate the effect of a 2-channel vibrotactile (Tactaid II) and a 16-channel electrotactile aid (Tacticon) on communication skills of deaf-blind children. The Tactaid vibrators are worn on the wrist or chest and the Tacticon is worn as a belt of electrical stimulators on the abdomen. A total of 3 communicative behaviors were selected for each child, resulting in 3 separate sub-studies per child over the first 2 years of the project. A single-subject alternating treatment design was used and the data were analyzed using the Combined Wilcoxon test. In the first sub-study, subjects displayed significantly more of the desired behaviors with the Tactaid than the Tacticon, and in the replication-across-change agents, study, both devices produced significantly more of the desired behaviors than no device (.05 level). This research was supported by the Office of Special Education and Rehabilitative Services (OSERS Grant No. GOO8630416) and is presently being validated with infants and preschoolers with dual sensory impairments using only the 2-channel Tactaid II+ vibrotactile device (89-92, OSERS Grant No. H086G90010).
PROCESSOR-CONTROLLED, ADAPTIVE, EAR RESPONSIVE HYBRID HEARING AID

Samuel Gilman and J. Phil Mobley  
House Ear Institute

A digitally-controlled, multi-channel analog hearing aid with each channel forming a complete servo loop so as to maintain the desired spectrum and level at the eardrum is described. Average band spectrum levels at the eardrum are compared to stored reference levels and each channel gain is independently controlled to provide the desired sound spectrum established by the audiologist. Adaptive signal-to-noise enhancement will be evaluated.

DESIGN AND EVALUATION OF FIR FILTERS FOR DIGITAL HEARING AIDS WITH ARBITRARY AMPLITUDE AND PHASE RESPONSE

Lars B. Nielsen, Ed Wu, Michael W. Hoffman, Kevin Buckley, and Sigfrid D. Soli  
House Ear Institute  
†University of Minnesota

An automated design algorithm has been developed for design of arbitrary amplitude and phase response FIR filters for hearing aids. This method uses a weighted least squares frequency sampling approximation with these features: (a) frequency samples specifying amplitude and phase can be arbitrary and closely spaced in critical regions, (b) each frequency sample can be weighted and (c) interpolation can be done between samples using units appropriate to hearing. A method for objective evaluation that provides a perceptually-based measure of the closeness of fit has also been developed. Filter responses and their objective evaluations are presented for a number of typical hearing losses that include compensation for the transducer's amplitude and phase response.

NEW ELECTROACOUSTIC TESTS FOR HEARING AIDS HAVING NON-LINEAR SIGNAL PROCESSING

Lawrence Revit  
Frye Electronics, Inc.

Traditional electroacoustic tests fall short of describing the performance of hearing aids having non-linear signal processing. Several new tests, now available in clinical hearing-aid test equipment, more completely describe the performance of such hearing aids. These new tests include:

- Families of gain-and-output-versus-frequency response curves using wideband, "speech-like" test signals;
Input/output curves, input/gain curves, and attack/release-time
tests, using variable test-signal frequencies, wideband, speech-like
signals, and stimuli of varying duration;
Frequency response curves using a pulsed, wideband signal
(simulating speech) in the presence of a continuous bias tone
(simulating noise).

This poster will describe examples of these tests and their applications to
evaluating "multi-band-compression," "ASP," "adaptive compression," and normal "AGC"
circuits. Other proposed tests, now available only in the laboratory, will be discussed, as
well.

EVALUATION OF A PROCEDURE FOR MEASURING LOUDNESS DISCOMFORT IN
CHILDREN

Richard C. Seewald, Debra L.C. Zelisko, Jean-Pierre Gagne
University of Western Ontario

The output limiting characteristics of a hearing aid may be the most important
factor related to a successful hearing aid fitting (Hawkins, 1984). Loudness Discomfort
Levels (LDLs), are often used to set output limiting levels because threshold levels can
not be used to accurately predict loudness discomfort (Hawkins, et al. 1987). The
objective of this study was to evaluate a procedure for determining LDLs in school-age
children. The procedure outlined by Hawkins et al. (1987) for adults, was adapted for
application with children. Nine severe-to-profoundly hearing-impaired children between
the ages of 12-19 years served as subjects. LDL was defined as the average level on the last
5/8 trials where the subjects indicated the sound was uncomfortably loud, when pulsed
pure tones were presented at the audiometric frequencies 250-2000 Hz. Test stimuli were
delivered to the subject's test ear via a high-output button-type receiver attached to a
custom earmold. A 14 mm vent was drilled in each custom earmold to insert a probe-tube.
The ear canal SPLs associated with the LDSs were measured directly, eliminating several
possible sources of measurement error. The within and across session intra-subject
variability associated with these measures, along with several additional findings will be
presented.

FITTING THE CID DIGITAL HEARING AID: ACCURACY AND SUBJECTIVE
FREQUENCY/GAIN PREFERENCES

Marilyn French-St. George and Hope Metzger
Central Institute for the Deaf

A four channel digital hearing aid was fitted to 12 hearing-impaired participants
according to the NAL prescription (MSE of fit less than 3 dB). Six alternative
frequency/gain functions were calculated to modify the NAL response below 3 K Hz.
Using an elimination tournament procedure, subjects selected frequency/gain contours
that maximized clarity. Results suggest that NAL under-estimates the low frequency
gain required at low signal inputs (55 dBA), while somewhat over-estimating the gain
required at high signal input levels (75 dBA). Furthermore, preliminary data suggest that
naive users are less tolerant of background noise and prefer relatively less low frequency amplification in noise compared to quiet at all signal levels. The implication of these data for future hearing aid developments will be discussed.

ADAPTATION OF THE TELEPHONE SIGNAL FOR THE HEARING IMPAIRED

Mark Terry
University of Colorado

The telephone acts to reduce the bandwidth of the speech signal. Ordinarily this results in minimal loss in intelligibility. However, for the hearing impaired telephone communication can be problematic, even for those fitted with hearing aids. This study investigates the effectiveness of tailoring the telephone signal for the hearing impairment.

Listening tests using a model of sensory-neural hearing loss indicated that frequency dependent amplification and compression would be effective. Initial experimental studies indicate that adaptively modifying the telephone speech signal can increase intelligibility for hearing impaired users.

AGING AND SPEECH PERCEPTION: AUDITIVE AND COGNITIVE FACTORS

John C.G.M. van Rooij
Free University, Amsterdam

The aim of the research reported was to determine the relative contribution of auditive and cognitive factors to speech perception in elderly listeners. A test battery was constructed comprising auditive, cognitive, and speech-perception tests. The battery was administered to 72 elderly listeners. Multivariate analyses showed that the deterioration of speech perception in the elderly consisted of two statistically independent components: (1) a large auditive component mainly representing the progressive high-frequency hearing loss, and (2) a smaller cognitive component mainly representing a general performance decrement, indicated by a general slowing of performance and a reduced memory capacity.

In a second study, intended to replicate these results on a group of elderly that are less likely to participate in laboratory research, no significant cognitive effect was found. This discrepancy may have been due to differences in cognitive ability between the groups and/or differences between the lists of sentences that were used to measure speech-reception threshold. These possibilities were investigated by two additional experiments. In the first experiment, the validity of a procedure to quantify the linguistic entropy, i.e. information content, of speech stimuli was tested. In the second experiment, this procedure was used in order to resolve the discrepancy between the two earlier studies.
EFFECT OF PROCESSING PARAMETERS ON THE PERFORMANCE OF MULTICHANNEL COMPRESSION HEARING AID

E. William Yund and Krista M. Buckles
Veterans Administration Medical Center, Martinez

Previous research (E.W. Yund, H.J. Simon and R. Efron, J. Rehab. Res. Dev. 24, 161-180, 1987) has demonstrated the effectiveness of an 8-channel compressing hearing aid for individuals with sensorineural hearing loss in the task of speech discrimination in a background of speech-band noise. The goal of the present research is to improve the performance of such a multichannel compression system by determining how its performance varies with changes in its parameters. Data will be presented from studies of several parameters: (1) the number of channels, 4- to 16-channel systems were tested; (2) the distribution of channels along the frequency spectrum, systems with concentration of narrow bands in different frequency regions were compared; (3) the method of compression, three methods were examined – that of our aid, the "standard" method and an "instantaneous" method; (4) the time window of our method of compression, 5 msec to 200 msec were used. Collateral studies concerning the effect of our multichannel compression system on the speech recognition ability of a normal-hearing subject and comparisons of multichannel compression with flat and frequency-equalized linear amplification will be included. [Work supported by VA Rehabilitation Research and Development Service].
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<td>Quentin Summerfield</td>
<td>MRC Institute of Hearing Research</td>
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