Issues in Advanced Hearing Aid Research

Lake Arrowhead Conference Center
May 30 - June 3, 1994

Chair: Dianne Van Tasell
Co-Chairs: Louis Braida and Stuart Gatehouse
Organizational Co-Chair: Sigfrid Soli

Sponsors

House Ear Institute  Acoustical Society of America  American Academy of Audiology  Hearing Industry Association
# DAILY CONFERENCE SCHEDULES

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**NOTE:** Invited participants are requested to send original records of travel expenses BEFORE JUNE 15 to: Susann Böehlke, House Ear Institute, 2100 W Third St Los Angeles CA 90057-1902 Voice: 213/353-7047 Fax: 213/413-0950 Email: susann@hei.org
CONFERENCE PROGRAM

ISSUES IN ADVANCED HEARING AID RESEARCH

May 30- June 03, 1994

Diane Van Tasell, Chair
Louis Braida and Stuart Gatehouse, Co-Chairs
Sigfrid D Soli, Organizational Co-Chair

I. Compression Nonlinearities and Nonlinear Hearing Aids

Monday, Evening

Jont Allen, Moderator
AT&T Bell Laboratories

7:50 PM

Dynamic Range Compression in Cochlear Mechanics

Stephen T. Neely
Boys Town National Institute

The dynamic range of sensory hair cells within the cochlea is no more than 60 dB, from their thermal noise floor to saturation of their mechano-electric transducer. The dynamic range of normal hearing is closer to 100 dB, from just audible to unbearably loud sounds. The cochlea allows our sense of hearing to have this expanded dynamic range because it is able to accomplish dynamic range compression prior to mechano-electric transduction by the inner hair cells. Dynamic range compression is the primary function of outer hair cells, which have the unique ability to lengthen and contract in response to internal voltage changes. The non linearity of loudness summation, two-tone suppression, and distortion product otoacoustic emissions are all well known side-effects of this dynamic range compression. Loss or impairment of normal outer hair cell function causes the loss of these nonlinear features of cochlear mechanics. Loudness recruitment in ears with hearing loss should be interpreted as a loss of dynamic
range compression within the cochlea. The functional implementation of dynamic range compression can be linked to saturation of mechano-electric transduction in outer hair cells, if one assumes that outer hair cells are amplifying cochlear vibrations. A better understanding of how the cochlea implements dynamic range compression may lead to ideas for improving compression in hearing aids.

8:30 PM

Dynamic Range Compression Using a Loudness Model

Birger Kollmeier, Volker Hohmann
Universität Oldenburg, Germany

Patients with a sensorineural hearing loss suffer from a distorted mapping between the level of acoustical signals and the perceived loudness (i.e., the "recruitment" phenomenon). To compensate for this effect, a binaural multi-band-dynamic-compression algorithm was developed which incorporates a loudness model to control the inter-band interaction. This loudness estimation model is based on the "Würzburger Hörfeldskalierung", i.e., categorical loudness judgments on a scale ranging from "inaudible" and "very soft" to the values "very loud" and "too loud". It was validated with psychoacoustic tests in normal and impaired listeners. The resulting algorithm was implemented in real-time on a multi-signal-processor digital master hearing aid and was tested with impaired listeners employing speech intelligibility and transmission quality tests. The preliminary results in quiet appear promising with respect to restoring the loudness impression while maintaining a high transmission quality. In addition, the maximum obtainable speech intelligibility score is improved indicating that the algorithm compensates for the "discrimination loss" observable in some sensorineurally impaired listeners.
Recent Results in Compression Amplification

Harry Levitt, Arlene Neuman and Elizabeth Kennedy
City University of New York

There are several forms of compression amplification, each of which has both advantages and shortcomings. The benefits of appropriate compression amplification are multi-dimensional. Among the variables to be considered are comfort, sound quality and other perceptual attributes in addition to intelligibility. Experiments on the evaluation of several of these attributes in single-channel compression systems will be reported.

A promising method of signal processing which can be viewed as a phonetically based form of amplitude compression is that of automatic adjustment of the consonant-vowel ratio. The increase in consonant-vowel ratio needed to maximize consonant recognition as a function of phonetic environment and audiogram shape has been investigated. The results of these experiments will be summarized and their implications for the design of the compression hearing aids will be discussed.

II. Long-term Effects of Auditory Deprivation and Amplification

Tuesday, Morning

Denis Byrne, Moderator
National Acoustics Laboratories

9:00 AM

Acclimatisation to Amplified Speech - Experiments to Investigate the Basis of Effect

Stuart Gatehouse and Ken Robinson
Experiments have shown that a form of perceptual learning can occur for hearing aid users, whereby speech identification through an unfamiliar frequency-gain characteristic improves over a time-course of some two to three months. Though the generality of this effect remains open to question and is the subject of further study, the experiments reported here aim to investigate the hypothesis that this "acclimatization" is due to a frequency-specific rescaling of loudness coding according to the distribution of intensities experienced by the auditory system.

The hypothesis suggests that measures of intensity discrimination and/or loudness growth should change with the time in ears that have been provided with amplification, relative to those ears that remain unaided. In subjects with bilateral symmetric sensorineural hearing impairment who are long-term users of a single hearing aid, there is a systematic effect of sound pressure level intensity discrimination ability, such that at frequencies where the hearing aid is providing material gain, the intensity discrimination abilities of a normally aided ear are superior to those of the normally unaided ear, while at lower presentation levels the converse is true. This finding does not hold at a frequency where the hearing aid does not deliver material gain, and the pattern of psychoacoustic results by ear-of-aiding follows closely the pattern previously obtained for speech identification scores.

Further experiments on subjects fitted with a hearing aid for the first time replicate the retrospective findings, indicating that intensity discrimination abilities improve systematically at high presentation levels and frequencies where the hearing aid delivers gain in the normally aided ear, but not in the normally unaided ear. The results suggest that there is a potentially tractable auditory basis to the changes in speech identification scores which can be investigated in terms of recognized psychoacoustical phenomena and suggest to us that a comprehensive remapping of the intensity resolution could occur changing patterns of auditory exposure. The determinants, time course, limits and functional value of this re-allocation are matters of substantial theoretical and practical interest, and form the basis of future studies.
Late Onset Auditory Deprivation

Shlomo Silman
Brooklyn College, CUNY

Recently, numerous studies in several countries reported that speech perception of the unaided ear of monaurally-aided subjects decreased over time while the speech perception of the aided ear of monaurally-aided and binaurally-aided subjects stayed unchanged or improved. This phenomenon has variously been termed deprivation (Silman et al., 1984; Hurley, 1990, 1993; Poole and Jerger, 1994), central suppression (Hood, 1990), acclimatization (Gatehouse, 1989), inactivity (Dieroff, 1989 and 1993), and ear dominance (Hattori, 1993).

The purpose of this presentation is to present up-to-date research in this area, both perspective and retrospective, from several laboratories.

10:20 AM -- BREAK

10:30 AM

Michael Merzenich

11:10 AM

Maturation Of Hearing Aid Benefit In Elderly First-Time Users.

Robyn M. Cox.
Memphis Speech & Hearing Center

Previous research determined that the benefit provided by a hearing aid shortly after fitting generally underestimated the benefit seen after 10 weeks of hearing aid use (Cox and Alexander, 1992). The effect was called benefit maturation and it was observed in both objective and subjective types of benefit data. Furthermore, the amount of
benefit maturation was quite variable across individual new hearing aid users. In objective tests, benefit maturation was observed quite prominently in a testing condition in which subjects listened to speech in noise without access to visual cues, whereas the effect was not seen under the same auditory conditions when visual cues were available. This led to a hypothesis that maturation effects were due to optimization of the use of newly-audible high-frequency speech cues.

These results have implications for both clinical and research endeavors because, in both realms, amplification systems are often evaluated or compared using benefit measured shortly after a hearing aid fitting and the results are assumed to reflect the performance that would be obtained over the long term. These procedures should be viewed with caution. Before we can predict the benefit that would be obtained from a particular hearing aid after a period of adjustment to amplification, a clearer understanding is needed of the variables influencing benefit maturation.

This paper reports a follow-up study of benefit maturation which attempted to replicate the effect in a larger group of elderly novice hearing aid wearers and addressed the following additional issues:

1. Can we determine in advance which individuals will experience a sizable maturation effect?

2. Is maturation observable in some speech features and not others, consistent with improved utilization of high-frequency cues?

3. Could the effect be explained by learning for the test situation rather than by actual performance improvement?

4. Is the presence or absence of benefit maturation related to long-term success in hearing aid use?

5. Do maturation effects continue beyond three months of regular hearing aid use?
III. Speech Perception I: Hearing Aid Processed Speech

*Tuesday, Evening*

Harry Levitt, Moderator
CUNY Graduate Center

7:50 PM

The Perception of Amplified Speech by Hearing Impaired Listeners: Acoustic Correlates

Patricia Stelmachowicz
Boys Town National Research Hospital

In general, audibility-based approaches to hearing aid selection have focused on the audibility of the long-term average speech spectrum (LTASS). Recent evidence, from a number of studies with adults, suggests that changes in spectral shaping do not alter performance as long as the majority of the LTASS is audible. For young hearing impaired children in the process of learning speech and language, however, the audibility of specific speech sounds may be critical since they will not have the same linguistic competence as adults. In addition, ongoing advances in amplification technology (e.g., multiband signal processing, level-dependent frequency shaping, full dynamic range compression, adaptive compression) make it difficult to predict the audibility of short-term components of speech from the amplified LTASS. This study was designed to investigate how the audibility of specific phonemes, as processed by two different hearing aid circuits (linear and full dynamic range compression), is related to performance on a nonsense syllable recognition task. Data were obtained from 3 subjects with moderate sensorineural hearing loss. Nine unvoiced consonants were presented in two vowel contexts (/i/ and /a/) in both the pre- and post-vocalic position at three intensities. Estimates of audibility were based upon each subject's thresholds and an acoustic analysis of the amplified signal that varied across phonemes and consonant position. While the performance on selected conditions appeared to vary by hearing aid type, only one subject showed a statistically significant difference between the two
hearing aid systems. Acoustic analysis revealed that the consonant-to-vowel ratio varied across consonant, hearing aid type, input intensity, and consonant position. A number of strategies were derived in an attempt to predict performance from a simple measure of audibility and the relative merits of these measures will be discussed.

8:30 PM

The Effects of Single-Band Compression on Temporal Speech Information

Dianne J. Van Tasell and Timothy D. Trine
University of Minnesota

Compression with time constants shorter than 100 ms (syllabic compression) is being used in many recently-developed personal hearing aids. One outcome of syllabic compression is that the broadband envelope of speech is smoothed relative to uncompressed speech; it is not known how this alteration in the waveform envelope affects its information content. Two assumptions formed the basis for this work: 1) Single-band compression should have a negligible effect on intelligibility for normal-hearing listeners because any temporal information that might be removed by compression will be redundant with spectral speech information; and 2) The effects of compression will therefore be observable only when redundant spectral information is removed or minimized. Spectral information was removed from speech testing materials by multiplying the rectified speech waveforms by a constant-amplitude noise; the resulting signal is speech-correlated noise (SCN). Normally-hearing subjects listened to SCN stimuli derived from a set of 19 /aCa/ nonsense disyllables and from a set of limited-context sentences, processed with a variety of single-band amplitude compression parameters. Only the most severe syllabic compression (compression ratio = 8; time constant = 50 ms) affected subjects' abilities to use speech temporal information, and even then the effects were small. Comparison of results across processing conditions suggests that temporal phonemic information can be conveyed either by the relatively slow syllabic envelope of speech or by the complex temporal pattern related to voiced and
unvoiced speech segments. When either form of information is removed, normal-hearing subjects can rely on the other. When both are altered, the temporal information content of speech is reduced.

9:10 PM

Reexamining the Benefits of Adaptive Frequency Response for Noise Reduction Hearing Aids

David Fabry
Mayo Clinic

The primary complaint of many hearing aid users remains performance in background noise. Single microphone hearing aids that use an adaptive high pass filter to alter frequency response in noisy listening environments were initially heralded as a solution to the problem, but results have failed to support claims of improved speech recognition in noise under most test conditions. Multiple-microphone digital array processors offer promising results, but wearable ear-level units remain a few years away from reality. This talk will focus on the results of two recent experiments that use prototypes of commercially available hearing aids that use: 1) a directional microphone and adaptive frequency response algorithm, 2) a two-microphone FM system that uses adaptive frequency response on the environmental (ear-level) microphone only. Results of these two studies suggest that the improved performance in noise from these wearable devices is on par with many digital algorithms under development for noise reduction.
IV. Loudness/Dynamic Range Considerations in Hearing and Hearing Aid Fitting

Wednesday, Morning

Jerry Studebaker, Moderator
Memphis Speech and Hearing Center

9:00 AM

Discrimination of Changes in Intensity of Gated Sinusoids Based on Auditory Nerve Spike Counts

Evan M. Relkin and John R. Doucet
Syracuse University

A simple working hypothesis is that the intensity of an acoustic stimulus is coded by the number of action potentials (hereafter referred to as the spike count) produced in one or more auditory-nerve neurons in response to the stimulus. By applying the principles of signal detection theory, it is possible to determine the just noticeable difference (JND) in intensity for an 'ideal observer' who uses spike count as the decision variable. This JND places a limit on the best performance (i.e. minimum JND) of a listener whose 'central processor' uses the same code. Minimum JND's for single neurons can approach those seen psychophysically but only for a severely restricted range of intensities relative to the dynamic range of hearing. Data will be presented from ours and other laboratories that show how the dynamic range of single neurons can be shifted by the actions of masking stimuli and efferent activity. However, these effects are not nearly sufficient to solve the dynamic range problem. Therefore intensity discrimination must be determined by the responses of many, if not all, neurons. Modeling done by others shows that the optimal combination (in a strict mathematical sense) of the outputs of the neurons innervating even a single hair cell results in computed JND's that exceed psychophysical performance by as much as an order of magnitude over much of the dynamic range hearing. While these models establish that there is sufficient information in optimally combined spike counts to account for intensity discrimination, they do
not tell us how the neural outputs are actually combined in the central auditory nervous system.

One sub-optimal code that has often been suggested is simply the sum of the action potentials fired by all auditory neurons. This hypothesis has been tested theoretically by many, but in each case, their models incorporated assumptions and/or simplifications now known to be false. Rather than repeat history, we decided to test the 'net spike count' hypothesis empirically. To do so we have developed a new technique for recording a compound potential that is proportional to the net spike count. We call this potential the Peristimulus Compound Action Potential (PCAP) because it can be recorded throughout the duration of a stimulus and not just at stimulus onset (and other transients) as is the case for the traditional Compound Action Potential (CAP). Our results show that for gated, high frequency sinusoids, with and without the presence of a simultaneously gated notch-noise, JND's based on the net spike count agree well with comparable psychophysical data. (The experiments pertaining to the PCAP are the basis of the doctoral dissertation of JRD.)

9:40 AM

Intensity Discrimination and Loudness

Neal F. Viemeister
University of Minnesota

For auditory communication it is clear that intensity or amplitude changes in a signal are of fundamental importance. This paper will review the basic psychophysical data on sensitivity to intensity changes, both "static" (intensity discrimination) and dynamic (envelope detection), in normal and impaired hearing. The current theoretical understanding of the processing of such information will be briefly reviewed. Loudness phenomena, particularly as they relate to intensity discrimination and processing will be discussed, as will recent attempts to relate intensity discrimination and loudness. Special emphasis will be devoted to the phenomenon of loudness
recruitment and its implications for intensity discrimination and the processing of supra-thresholds intensity changes.

10:20 -- BREAK

10:30 AM

Loudness Scaling Versus Comfort and Discomfort Levels in Normal and Impaired Hearing

Claus Elberling
OTICON Research Unit, Eriksholm, Denmark

The relationship between the 'classical' measures of loudness growth: most comfortable level (MCL) and uncomfortable level (UCL) and the 'fashionable' measures of loudness growth: loudness scaling either as category rating (CAR) or [restricted] magnitude estimation (RME) has been evaluated in a series of experiments. Pure tones at 500 and 2000 Hz were presented over insert earphones to 10 subjects with normal hearing and 29 subjects with mild to moderate sensorineural hearing impairment. Prior to each test the subjects were given written instructions.

Since MCL is a poorly defined quantity we use most comfortable range (MCR) with a lower and a higher limit, MCLlow and MCLhigh. They are defined as the 50%-points on the ascending and descending psychometric function respectively as measured by the method of limits. The UCL is defined as the 50%-point on the ascending psychometric function also as measured by the method of limits.

Loudness scaling is performed by a procedure where the stimulus levels are generated automatically and presented in random order. The listeners respond on a touch pad equipped with either a scale divided into seven categories (CAR) or a scale without categories (RME). The range of levels is adapted to the individual's dynamic range whereas the number of levels is kept constant.
The mean values of MCR and UCL as a function of hearing loss and the test-retest variances are in accord with results from comparable experiments reported in the literature. When referenced to dB HL, there is no significant difference between the data obtained at the two frequencies.

For both subject groups, CAR and RME loudness functions can be adequately fitted by an exponential function. For the normal hearing subjects, the fitting constants comply with those from similar studies reported in the literature and when referenced to dB HL, there is no significant difference between the data obtained at the two frequencies. In a logarithmic presentation the exponentially fitted loudness functions become linear with the slope being the most important parameter. For this quantity the test-retest variance is low.

The individual values of MCLlow, MCLhigh and UCL are projected on to the scaling axis through the corresponding loudness function and here expressed in 'scaling' units. In this way comparison can be made between the two groups of subjects. Across frequency and type of loudness scaling it appears that the two groups display the same relationship between the MCR and UCL data and the loudness functions.

The slope of the individual loudness function can be predicted from other data. Threshold (HTL) alone is the most important single parameter and explains about 60% of the variance. However, by incorporating the MCR and UCL-data about 80% of the variance can be explained ($r = 0.89$). Most interestingly, a combination of HTL, MCLlow and MCLhigh alone can explain about 77% of the variance ($r = 0.88$).

The results seem to warrant the following conclusions:

* MCR and UCL can be measured with high reproducibility.
* Loudness scaling can be carried out with relatively simple equipment and the reproducibility is high.
* CAR and RME are equally well suited.
* There exists a functional relationship between MCR and UCL data and the loudness scaling and both measures seem to be related to the perception of loudness.
* The loudness scaling functions can be predicted accurately from HTL, MCLlow and MCLhigh in most cases.

11:10 AM

Evaluating the Potential of Signal Processing Schemes Using Simulations of Reduced Frequency Selectivity and Loudness Recruitment

Thomas Baer
University of Cambridge, England

This talk will be concerned with work progress in Brian Moore's group at the University of Cambridge. Signal processing has been developed to stimulate reduced frequency selectivity (Baer & Moore, 1993, 1994) and loudness recruitment/threshold elevation (Moore & Glasberg, 1993). The purpose of these simulations is to tease out, using normally hearing subjects, the separate effect on speech intelligibility of these two components of hearing impairment. Results, partly reported at the last meeting, suggest that both components contribute to reduced intelligibility of speech in noise, and loudness/recruitment threshold elevation also affects intelligibility of speech in quiet. The focus of the present talk will be on the role of the simulations in assessing the need to compensate for reduced frequency selectivity and loudness recruitment and in developing and evaluation signal processing schemes to do so.

The principles underlying the simulation of reduced frequency selectivity were used to devise a spectral enhancement scheme. The simulation uses spectral smearing to produce excitation patterns associated with broadened auditory filters in a normal auditory system. The enhancement scheme uses a spectral sharpening algorithm that explicitly attempts to produce near normal excitation patterns in an auditory system with broadened filters. The simulation of reduced frequency selectivity was used to evaluate the enhancement scheme with normally hearing subjects. Results suggest that the spectral
enhancement can partially compensate for the deleterious effects of reduced frequency selectivity on speech intelligibility in noise. Experiments with actual hearing impaired subjects, and use of the recruitment simulation to demonstrate and evaluate different hearing aid processing schemes will also be discussed.

V. Developments in Signal Processing

James M. Kates, Moderator
CUNY Graduate Center

7:50 PM

Performance of Objective Measure System for New Compensation Techniques

Janet C. Rutledge and Apichat Tunthangthum
Northwestern University

With the explosion of DSP microprocessor technology, digital hearing aids with complex compensation processing techniques are becoming a reality. Evaluation of these new techniques requires extensive subject-based tests in order to truly conclude anything about their effectiveness. However, during the development stage, it isn't practical to do this each time a parameter is changed. Therefore an objective measures system based on neural networks has been developed to predict the results of a subject-based test. This objective measures system was developed using the subject-based test results of ten hearing impaired listeners. Parameters related to the loudness level of the compensated speech signal are extracted from its frequency spectrum. These parameters are then used to train a neural network based phoneme classifier. It gives reasonable prediction results when training and testing are performed on different subsets of the group having similar audiograms with both the same and different compensation techniques. These results will be compared with the performance of alternative implementations of the objective measure system.
Directional Hearing Aid Based on Multi-Microphone Array Technology

Wim Soede, Witteveen+Bos Consultants
F.A. Bilsen and A.J. Berkhout, Delft University of Technology
The Netherlands

The hearing impaired often have great difficulty to understand speech in surroundings with background noise or reverberation. A directional hearing aid might be beneficial in reducing background noise in relation to the desired speech signal. To this end microphone systems were developed with strongly directional characteristics, using array techniques. Considerable attention was paid to optimization and stability.

Free-field simulations of several robust models show that a Directivity Index of 9 dB can be obtained. Simulations were verified with a laboratory model. The results of the measurements show a good agreement with the simulations.

Based on simulations and measurements, two portable models were developed and tested with a KEMAR-mannequin. The KEMAR measurements show that the two models give an improvement of the signal-to-noise ratio of 7 dB in a fully diffused soundfield.

The benefit of these microphone arrays for the hearing impaired was tested in a sound insulated room. One loudspeaker was placed in front of the listener simulating the partner in a discussion, and a diffused background noise was produced by eight loudspeakers placed on the corners of a cube. The hearing impaired subject is placed in the center of the cube. The speech-reception threshold in noise for simple Dutch sentences was determined with a normal single omni-directional microphone and with one of the models.

The results of the listening tests with 44 hearing-impaired subjects will be presented showing an average improvement of the S/N-ration of 7.0 dB for monaural fitting.
Automatic Speech Recognition to Aid the Hearing Impaired

Louis D. Braid
Massachusetts Institute of Technology

Although many hearing impaired individuals make use of speechreading, the ability to communicate through speechreading alone is severely constrained because many acoustic distinctions important to communication are not manifest visually. In the Manual Cued Speech system these ambiguities are resolved by the speaker through the display of handshapes at various locations near the face. Our studies of the reception of manual cued speech by highly trained receivers indicate that high levels of accuracy can be achieved at near normal speaking rates. Previous attempts to develop an automatic cueing system that does not require the cooperation of the speaker were based on automatic speech recognition techniques that were incapable of achieving the performance required. Our measurements of several existing recognition systems suggests that recognition technology may now be adequate for the generation of cues that aid speech reading. We discuss our approach to the design of an automatic cueing system and relate it to other applications of automatic speech recognition to the alleviation of the effects of hearing impairment.
VI. Industry Short Reports / Manufacturers' Forum

Thursday, Morning

David A. Fabry, Moderator
Mayo Clinic

9:00 AM

Jeremy Agnew
Starkey Laboratories

9:30 AM

Nikolai Bisgaard
GN Danavox as, Denmark

The successful fitting of tomorrow's hearing aids based on sophisticated signal processing systems will require transmission of substantial amounts of data. The results of a multitude of diagnostic investigations, residual hearing capacity evaluations and acoustic measures need to be transferred to a fitting program that can derive the appropriate coding of the signal processing system in the hearing aid. This code set then needs to be transferred to the hearing aid and results of evaluation procedures need to be recorded in the client record and applied for refinement of the fitting procedures.

The PC is the natural platform to build this system around, connecting audiometric equipment, acoustic test systems and a hearing aid programmer. The NOAH software system and the HI-PRO interface is a framework for seamless connectivity between measurement equipment, fitting programs and hearing aids from a large group of manufacturers.

As a result of a unique industry co-operation NOAH defines a set of protocols for exchange of data between the system elements. All data relevant to the client and the fitting professional is kept in an open database structure, whereas manufacturer specific information is
stored in private data units only accessible through the manufacturers own programs.

10:00 AM

Tsuyoshi Mekata
Matsushita Electric Industrial Co., Ltd., Japan

We have developed a portable multi-function digital hearing aid. The size of the hearing aid is 59 x 63 x 26 mm and the weight is 98 grams. We mounted bare DSP (Digital signal processor) and amplifier chips on the circuit board directly to realize this small size. The processing speed of the DSP chip is 6.6 MIPS. It works 30 hours continuously on one rechargeable Lithium battery. The DSP used in it has an A/D converter and two D/A converters built in and works with 3V, 30mA power supply. The output signal of one D/A converter controls preamplifier gain and can be used for digitally controlled AGC. The sampling frequency is 10.7kHz and frequency band width for speech processing is 4.5 kHz.

We introduced four kinds of basic software which can be combined: improved temporal enhancement for consonants, impulsive sound suppression, 3 channel compression and enhancement of spectral contrast (quasi formant enhancement). By combination of software, 14 kinds of programs can be installed in DSP on-chip ROM. A master hearing aid based on a personal computer measures the residual hearing ability of the hearing impaired and sends parameters for the programs to an Electric Erasable Program ROM. One of the parameters used to select a program for a specified user. Each program has four kinds of function. The user can select a function using push switch depending on the environment. We plan to evaluate and improve the hearing aid. Products are expected to come on the market in 1995.
Research needs for hearing aid manufacturers include:

1) More complete models of the auditory system that stimulate auditory system functions and pathologies that may effect the benefit provided by hearing aids.

2) Determining how various types of hearing aid signal processing algorithms interact with different auditory system pathologies.

3) The appropriate psychoacoustic test data about the auditory system from which candidacy can be established a priority about which type of hearing aid signal processing algorithm would provide the most benefit.

4) Testing materials for quantifying speech recognition in noise to quickly and reliably distinguish between different types of hearing aid signal processing in "real world" listening situations.

Despite recent suggestions to the contrary, all hearing aids are not basically the same and all fitting methods are not equally good. Recent progress in 2nd-order directional microphones and fitting target software will be discussed, to indicate that many of the traditional problems with hearing aids may be close to reasonable solutions.
VII. Speech Perception II: Speech Perception Capacity

Thursday, Evening

Larry E. Humes, Moderator
Indiana University

7:50 PM

Evaluating the Speech Recognition Performance of Hearing Impaired Listeners

G. A. Studebaker and R. L. Sherbecoe
Memphis Speech and Hearing Center

We previously described an articulation index based procedure that predicts average speech recognition scores for particular test materials. That procedure has now been extended to include suprathreshold effects and the effects of age. The data forming the basis for these extensions will be described. If a suitable metric for the comparison could be found, the procedure could be used to compare the performance of individual subjects with the hypothetical average performance of subjects with the same characteristics of hearing loss and age. The statistical characteristics of three methods of comparing obtained and predicted results will be described based on theory and on data from normal hearing subjects. Preliminary data from a group of hearing impaired subjects using two of these methods also will be presented.

8:30 PM


Arthur Boothroyd
City University of New York

Intersubject and intergroup comparison of hearing aids and other sensory aids calls for tests that are: a) maximally sensitive to sensory
information, b) minimally sensitive to linguistic and cognitive status, but c) predictive of sentence-level speech perception performance. This need is acute for children with severe and profound hearing losses. THRIFT uses an oddity task in which the S must identify the odd-man-out in three utterances. The goal is to measure the probability of detection of phonologically significant contrasts in an unpredictable phonetic context without confounding by speech skills, reading ability or phonological knowledge. In a group of severely and profoundly deaf children aged 6.5 years and above, 50% of the variance in THRIFT scores was explained by a combination of hearing loss and sensation level, 7% by age, none by communication mode (oral vs TC) and 9% by error. This leaves 34% attributable to individual differences other than age and hearing loss. These findings suggest that THRIFT goes a long way towards meeting the goals of maximal sensitivity to sensory factors and minimal sensitivity to language and cognition. They also highlight the key role of sensation level (and, therefore, of Articulation Index) as a key determinant of performance in the severely and profoundly deaf. Unfortunately, the extent to which the unexplained variance in these data might reflect differences of phonological knowledge rather than of sensory capacity is not clear. To explore this issue further, the acoustical properties of the THRIFT recordings have been examined in detail and the results used to predict the performance of an ideal observer approaching the task acoustically, rather than phonetically. The relationship between the human and the ideal observer have been further explored in a psychoacoustic analog of THRIFT. The results suggest that the prephonological child is likely to be at a considerable disadvantage when taking THRIFT, especially on those contrast such as consonant manner and place for which the acoustic correlates are complex, multidimensional, and context-dependent.

9:10 PM

Perception of Temporal Speech Information by Hearing Impaired Listeners.

Christopher Turner, Pamela Souza, and Lauren Forget
Syracuse University
The temporal acuity of listeners with sensorineural hearing loss is currently a matter of some controversy. In this study, speech signals were digitally processed to remove the original spectral information, resulting in the time-varying speech envelope amplitude modulating a noise carrier. In addition to unprocessed speech stimuli, several processed-speech conditions were employed; the envelope of a broadband speech signal modulating a broadband noise, a lowpass speech signal modulating a lowpass noise, a highpass speech signal modulating a highpass noise, and a 2-channel signal comprised of the low- and high-pass signals combined. Recognition of envelope stimuli in modulated noise backgrounds was also tested. The hearing-impaired listeners performed more poorly on a recognition task than the normal-hearing listeners for unprocessed speech signals. However, for listeners with hearing losses of either flat or sloping configuration, there was essentially no deficit in their ability to use the temporal cues of speech, even in frequency regions of hearing loss up to 70 dB. These results imply that moderate-to-severe sensorineural hearing loss does not impair the temporal (nonspectral) acuity of listeners in terms of speech recognition. (Supported by NIDCD Grant R01 DC00377)

VIII. Binaural Hearing and Hearing Aids

Friday, Morning

9:00 AM

Binaural Directional Hearing with Hearing Aids

Sigfrid D. Soli
House Ear Institute

Binaural directional hearing can improve speech reception thresholds in noise by 6 dB or more for normally-hearing listeners. This and other related findings indicate that binaural hearing plays an important role in our ability to communicate in noisy environments. The common complaints of hearing aid users about difficulty hearing in noise suggest that several issues related to binaural directional hearing and hearing aids need to be addressed. First, the binaural capacity of the
hearing impaired individual should be assessed as part of the audiological evaluation of their impairment. In response to this issue, a clinical procedure for characterizing binaural capacity has been developed. Results of its use with hearing impaired individuals will be described.

Second, hearing aid selection, fitting, and evaluation should also include an assessment of aided binaural directional hearing to determine whether the hearing aid(s) enable the hearing impaired individual to utilize their binaural capacity. A clinical procedure for assessing aided directional hearing has also been developed to address this issue. We will report results of its use to assess directional hearing of hearing impaired subjects fitted with conventional hearing aids.

Third, hearing aid design and fitting goals should be expanded to include signal processing which maintains binaural directional ability, while compensating for loss of sensitivity. In response to this issue, we have developed a prototype binaural hearing aid and associated fitting system which has been evaluated in preliminary tests with normally hearing and hearing impaired subjects. The fitting procedure and the results of these preliminary tests will also be described.

9:40 AM

The Importance of Onsets in Lateralization

Ervin R. Hafer

The binaural "precedence" effect describes the fact that secondary sources or echoes have little effect on the perceived location of a source. Its existence has been used to argue that information beyond the initial stimulus transient is somehow lost or inhibited. This paper will describe a series of experiments whose goal has been to define the binaural enhancement of transient information over that in the ongoing signal, the functional place of this process for spatial hearing and the rules that govern what is a transient and what is not. In addition I hope to suggest some possible roles for that portion of the post-onset binaural signal that is processed and to discuss how the
primacy of transients for localization might enhance the usefulness of binaural processing in hearing aids.

10:20 BREAK

10:30 AM

Significance of Auditory Localization for Hearing Aid Fitting

Denis Byrne
National Acoustic Laboratories

William Noble
University of New England

This paper discusses the results of several studies of auditory localization in terms of their significance for hearing aid fitting. One study shows that hearing-impaired listeners report significant auditory localization difficulties and that this is correlated with reported speech recognition difficulties, after controlling for hearing level. Experiments show that hearing impaired listeners almost invariably have impaired localization, especially in the vertical plane. People with conductive/mixed (CM) hearing losses have greater difficulty with horizontal plane localization than do people with sensorineural (SN) hearing losses. For sounds at a comfortable listening level the wearer of hearing aids tends to make the localization worse for SN listeners but sometimes improves it for CM listeners. For people with moderate or severe hearing losses, bilateral fitting of hearing aids gives substantially better localization than monaural fitting. For people with mild hearing losses, experienced wearers of unilateral fittings localize as well, on average, as wearers of bilateral fittings. For sounds of unpredictable level, ITE hearing aids appear to offer no advantage, in localization, over BTE hearing aids. The paper concludes by discussing some possibilities for improving aided localization by optimal choice earmould type.
Binaural Hearing in Children Having a History of OME

Joseph W. Hall
University of North Carolina

Children with a prolonged history of otitis media with effusion were tested on the masking-level difference (MLD) both before and after insertion of PE tubes. The results of this binaural test indicated that MLDs were reduced not only before the child received PE tubes, but also reduced for a relatively long period of time after receiving PE tubes, even though hearing thresholds in quiet were normal. Such children also typically had abnormal auditory brainstem evoked responses, as evidenced by an abnormally long wave I-III interval. Furthermore, interaural asymmetries in the I-III interval of the ABR were correlated significantly with reduced MLDs. These results will be discussed in terms of early sound experience, and will be compared to results that have been obtained on adults with acquired conductive hearing.
CONTRIBUTED POSTER ABSTRACTS


Harvey B. Abrams, Theresa Hnath-Chisolm, Katrina Farrow, and Nancy Diss
University of South Florida

In this pilot study, a self-report questionnaire was used to assess the benefits of adaptive processing circuitry among 35 experienced and inexperienced hearing aid users. To determine benefit, the responses of adaptive processing circuitry users were compared to the responses of linear circuitry users. The results indicated that adaptive processing was beneficial, but only in terms of perceived quality of speech in noise among the experienced hearing aid users. In addition, both experienced and inexperienced hearing aid users reported satisfaction with their new hearing aids regardless of circuitry, suggesting that a careful clinical approach to fitting the instruments may be even more critical to patient satisfaction that the type of circuit selected.

Restoration of Loudness and Intelligibility Based on Critical Band Width, Speech Input Level and Dynamic Range of Hearing

S. Basseas, Ph.D., M. Meskan, M.A.
Beltone Electronics Corporation

The Zwicker Loudness Summation model has been modified to incorporate the widening of the critical bands as a function of both the speech input level and the dynamic range of hearing of an individual. A similar model is used for the estimation of the Intelligibility Index for varying input levels. The goal of fitting a hearing aid with a capability to adapt its frequency-gain response to different input levels is to restore loudness to normal while minimizing the error in loudness distribution among the critical bands. At the same time the range of the Intelligibility Index values should have resemblance to the curve of a normal hearing subject. This curve changes in a non-monotonic parabolic-like fashion as a function of input level to the cochlea. This suggests that maximizing the Intelligibility Index for speech at an average level of vocal effort is proper for both fixed response and an
adaptive hearing aid, but maximizing the Intelligibility Index for all input levels is not an appropriate goal for an adaptive aid. The critical-band based models require only the audiologic data of the air / bone conduction thresholds and LDLs at standard test frequencies using pulsed narrow band noise. Examples will be given demonstrating the importance of LDL in loudness matching. Depending on the LDLs of a hearing impaired individual, a prescriptive formula such as NAL-R provided too much or too little loudness even at the normal input level.

On the Use of Parametric Transformation of Speech Signals in Hearing Aid Research

Flemming K. Fink
Aalborg University, Denmark

Standard hearing aids are based on time domain manipulation of (speech) signals, although most experience in hearing impairment are related to frequency domain such as 1) frequency dependent hearing loss, 2) frequency-selectivity and 3) frequency dependent recruitment. Profound deafness is an example of hearing impairment that cannot be compensated for using time domain signal processing.

Research in digital signal processing and progress in development of implementation technology gives the opportunity to enter new directions in hearing aid research. This poster presents the PARTRAN concept giving new possibilities in hearing loss and hearing aid research based on manipulating the input signal in the frequency domain.

The PARTRAN concept is based on a parametric modeling of speech signals, transformation of the parametric model, followed by resynthesis of the transformed speech. The parametric modeling is standard AR-modeling. The resulting feature vector from this modeling is decomposed using a stable, numerically robust pseudo-decomposition technique generating second order sections each representing a resonant frequency (format). These characteristic elements in the frequency domain are each described by three parameters: resonant frequency, bandwidth and energy. This representation of speech frame is in the sense of hearing abilities very useful. Besides the efficient way to describe the signal this parametric description provides the possibility of independent manipulation of different characteristics of the speech signal:

Manipulating each energy value independently as a function of frequency, which is one method to compensate for frequency dependent hearing loss concerning threshold elevation.
Manipulating the bandwidths of each resonant area gives the opportunity to compensate for reduced frequency selectivity.

Recruitment or reduced dynamic range is very hard to handle in time domain. Recruitment can be frequency dependent and normally related to the amount of threshold elevation.

Profoundly deaf only have a narrow frequency range available. By transforming the most important areas in the frequency domain into this range, the hearing disabled can be provided some new information.

For research a hearing loss simulator will be of great benefit. As well as compensating different hearing losses, this concept can be used to simulate the hearing disabilities. Such a simulator provides the researcher with the possibility to evaluate a prototype hearing aid, in order to carry out a preliminary assessment of new prototype hearing aid.

[1] U Hartmann, K Hermansen, F K Fink: "Feature Extraction for Profoundly Deaf People", EUROSPSCECH'93, September 1993, Berlin, Germany

Method For Fitting Binaural Hearing Aids

Shawn Gao, Jean Sullivan, Sriram Jayaraman, and Sigfrid D. Soli
House Ear Institute

For a hearing aid wearer to perform binaural sound localization and to utilize directional hearing in noisy environments, it may be important to maintain at audible levels the binaural cues (i.e., interaural time and level differences) present without the hearing aid(s) in place. We have developed a method of achieving this hearing aid fitting goal for use with a prototype digital signal processing hearing aid. The method includes two major steps: Hearing Aid Equalization (HAE) and Hearing Loss Compensation (HLC). HAE is achieved with an FIR filter, which equalizes the amplitude and phase insertion effects of the hearing aids and maintains the binaural cues with the hearing aid(s) in place. The HAE filter coefficients are obtained from in situ probe tube measures of aided and unaided test signals using optimal filter calculations. HLC is also achieved with an FIR filter and associated gain. The HLC filter coefficients are obtained by using a weighted least-squares filter design technique. The target response for the HLC filter is determined from measures of electrical signals levels in the hearing aid circuit during threshold tests and during reference signal presentations in the soundfield. The HAE and HLC filters are convolved to produce a single filter. A description of the filter procedure will be provided, as well as examples of its use with hearing impaired individuals.
Development Of The Hearing In Noise Test for Children (HINT-C)

Donna Gelnert, Ann Sumida, Michael Nilsson, and Sigfrid D. Soli
House Ear Institute

The HINT-C, a version of the Hearing In Noise Test (HINT) for use with children as young as 6 years of age, is under development. The HINT-C is comprised of 13 phonemically balanced lists of 10 sentences and is designed to measure sentence speech intelligibility in conditions where speech communication commonly occurs, and where hearing impaired listeners demonstrate difficulty. This paper describes the procedure used to select the sentences and establish testing procedures for the HINT-C. Preliminary reliability data shows test retest reliability which compares quite favorably with comparable measures for adults. Preliminary normative data for 36 children will also be reported.

A Wearable Digital Hearing Aid

Stig Arlinger, Johan Hellgren, Thomas Lunner
University of Linköping, Sweden

A portable device for evaluation of signal processing schemes in hearing aids has been designed. The system consists of a single processing unit and two modified hearing aids. This makes it possible to obtain a binaural fitting with only one unit. The signal processing unit is based on a fixed point Digital Signal Processor from Texas Instruments, TMS320C50 (40MHz), which performs 20 million multiplications per second. The audio codec used has two AD converters and two DA converters. All converters are of Delta Sigma type. The program and the parameters are stored in a RAM with battery back up. This configuration makes it possible to store data about how the subjects have used the device, e.g. accumulative time of use and time each volume control setting or alternative programs has been used. Communication with the unit is made via the serial port of a PC. The processing unit runs on four penlight batteries (R6), which last for about 8 hours (alkaline batteries).

More Measures of Hearing Aid Benefit Over Time

Amy R. Horwitz and Christopher W. Turner
Syracuse University

This research further examines how the benefit derived from a hearing aid may change with time after the initial hearing aid fitting. Our study attempts to clarify
unresolved issues by providing controls for variables that have potentially confounded results to date. Specifically examined were the effects of test learning and familiarization and the effects of changes in preferred volume control settings. The time course of hearing aid benefit is measured in both new and experienced hearing aid users. Results of performance-based as well as self-assessed benefit measures are obtained and can be compared (using scores from both the NST and the PHAB).

Results to date (approximately 10 weeks) show little or no group mean changes in hearing aid benefit over time. However, results are variable across individual subjects. [Work supported by NIDCD Grant Number DC 0037]

Nonlinear Distortion on Speech Sounds
Influence on Speech Recognition for Hearing Impaired Listeners

Ann-Cathrine Lindblad
Karolinska Institutet, Sweden

Nonlinear distortion can influence both speech recognition and perceived sound quality negatively. It appears in amplifiers with peak clipping as well as in more advanced limitation systems like automatic gain control systems. The question is: How much nonlinear distortion can be tolerated in a hearing aid? And also—are the effects of distortion products more disastrous for certain speech sounds or certain frequency ranges than for others?

Nonsense rhyme words were used to test Swedish medical vowels or initial, voiceless consonants with either quadratic or cubic distortion, separately. The quadratic distortion was 25 or 50% of the original sound, the cubic distortion 10 or 20%. These distortion levels were chosen to be about equally detectable for the two kinds of distortion. In the consonant test there was distortion only to one consonant. The test was performed at the individual’s most comfortable level, with a high signal-to-noise ratio, and with TDH-39 headphones- not with hearing aids- to avoid introducing uncontrollable parameters.

The sixteen test subjects had sensorineural hearing losses of various severity and slope, mean HL (0.5 - 4kH) 63 dB, range 49 - 95 dB. Their age ranged from 25 to 68, median 40.

Results:
- The lower distortion levels did not give any significant changes in total consonant or vowel recognition compared to the undistorted condition.
- The higher distortion levels at both quadratic and cubic distortion caused significant decreases in vowel recognition for the whole group, and
significant decreases also in consonant recognition for subjects with flat audiograms, but not for subjects with sloping audiograms.

About half of the subjects with sloping audiograms could use the distortion products at low frequencies to increase their scores for consonants with main energy at high frequencies. The better the frequency selectivity was at low frequencies compared to the selectivity at high frequencies the larger was the positive effect. For a couple of subjects something similar also happened to vowels.

As a whole, vowels suffered more than consonants, and short vowels more than long vowels. Especially vowels with formants at low frequencies seemed vulnerable because of fusion between original formant and extra maxima introduced by distortion products below approximately 1 kHz. This was shown by combining confusion matrices with formant and distortion patterns for the vowels.

A Comparison of Dynamic Expansion and Additive Noise Simulations of Sensorineural Hearing Loss

Paul Duchnowski, David S. Lum, Patrick M. Zurek, and Louis D. Braida
Massachusetts Institute of Technology

Both dynamic level expansion (e.g., Villchur, J. Acoust. Soc. Am. 62, 665-674) and additive masking noise (e.g., Zurek and Delhorne, J. Acoust. Soc. Am. 82, 1548-1559) are capable of simulating the effects of sensorineural hearing impairment for listeners with normal hearing by raising detection thresholds and inducing loudness recruitment for tones. These simulation differ both with respect to phenomenological realism and in the relation of the acoustic properties of processed stimuli to the normal listener's hearing. We present the results of recent studies that compare the ability of the two simulation approaches to account for the impaired perception of speech and non-speech sounds. For moderate hearing impairments associated with sensitivity losses that can be simulated using both techniques, speech reception scores similar to those achieved by impaired listeners are achieved by listeners with normal hearing using both techniques, although there is a tendency for the simulations to lead to higher scores when high-frequency emphasis is applied. Measurements of simultaneous narrowband masking patterns and forward-masked psychophysical tuning curves indicate that both simulation techniques produce reductions in frequency selectivity that are similar to those observed in listeners with sensorineural hearing impairments (e.g., Dubno and Schaefer, J. Exp Psych. 43A, 543-564). By contrast, the expansion simulation improves intensity discrimination for tones whose loudness grows more rapidly than normal, while the masking approach does not. Although the masking simulation
of intensity discrimination is more consistent with findings for listeners with sensorineural hearing impairments (e.g., Florentine et al., J. Acoust. Soc. Am. 94, 2575-2586) marking can only be used to simulate losses of roughly 70 dB because of the intense noise levels required. The expansion approach can be applied to the simulation of these and more severe impairments.

A Guided Selection Method of Fitting Hearing Aids

Gil Magilen
Hearing Centers' Network

The Guided Selection Method (Audicibel 40(1):16-20, 1991) is a procedure in which the dispenser serves as a knowledgeable, responsible guide, who assists the adult client in selecting the optimal amplification characteristics for the client's needs. The goal of the fitting strategy is to obtain a final fitting in which the client can conclude that sounds are "natural" and understanding is "optimal". The conflict between the concepts of natural and optimal create the workspace within which a dynamic interactive fitting process can occur between the dispenser and the client.

Natural is defined as "nothing artificial added". A hearing aid sounds "natural" when it accurately reflects the external environment, without distortion, in the user's judgement. Often, the client initially does not know how a proper hearing aid fitting should sound. The client is instructed that "natural = good" (as opposed to "amplified = good"). At times, if what appears unnatural sounding to the client may provide better hearing in the estimation of the dispenser, then the client should be assessed for willingness to participate in an A/B test or acclimatization experiment. Optimal means the hearing aid provides the sound required for best understanding in all of the client's environments.

The method consists of stages of (1) client assessment, (2) client training (to become a trained listener for fitting purposes), (3) interactive fitting, and a (4) field evaluation period. A client assessment determines the full spectrum of the client's problems relating to hearing, and determines potential expectations. The client is then trained to become a communicative listener. All hearing aids are selected and adjusted to maximize the audibility of speech sounds. Natural is determined with an "object/context" paradigm. Common complex auditory objects (CCAOs), external and self-generated, are introduced in a variety of environmental contexts, and a "natural/unnatural" determination is requested from the
client. Electroacoustic adjustments and earmold modifications are made to eliminate unnatural sounds from the CCAOs while maintaining a maximal AI. A speech discrimination battery is used to probe the effects of the modifications. The resultant settings are assessed for AI, and the client is sent into the field (his environments) with instructions to evaluate all CCAOs and environments as natural/unnatural and to report back. Follow up visits are used to further enhance the natural/optimal balance.

This method of hearing aid fitting selects for hearing instruments that maximize the predetermined goals. Results of this method are reflected in the types of hearing aids currently dispensed by the author's facility. The author has personally fit nearly 1000 ReSound hearing instruments over the past three years (65% of total aids dispensed), fits primarily BTE hearing aids, and has a 96% repeat user and customer referral business. The method instills client confidence and satisfaction with both the quality of hearing improvement and the dispenser's efforts.

Demonstration of a Portable Multi-function Digital Hearing Aid

Tsuyoshi Mekata
Matsushita Electric Industrial Co., Ltd., Japan

At this session we will perform a demonstration of a portable multi-function digital hearing aid. Specification and functions of the hearing aid will be reported at the oral session on Thursday, June 3, in the morning. We will show four kinds of basic programs: temporal enhancement for consonants, impulsive sound suppression, 3 channel compression and enhancement of spectral contrast (quasi formant enhancement).

Norms for a Headphone Simulation of the
Hearing In Noise Test: Comparison of Physical
and Simulated Spatial Separation of Sound Sources.

Michael J. Nilsson and Sigfrid D. Soli
House Ear Institute

We have developed norms for a headphone-based procedure for administering the Hearing In Noise Test [Nilsson et al., J. Acoust. Soc. Am., (in press)]. Sentence speech reception thresholds (SSRTs) were measured adaptively in the presence of spectrally matched noise for 23
young, normal hearing male and female adults using two presentation systems: a soundfield presentation and a headphone simulation of the soundfield. Speech was presented at 0° azimuth in all conditions, and noise (when present) was presented at either 0°, 90°, or 270° azimuth at 65 dB(A). The headphone simulation used head-related transfer functions (HRTFs) measured off a KEMAR mannequin to introduce the appropriate interaural cues. Correction factors between the current headphone and soundfield measurements were calculated to correct previous soundfield norms. SSRTs were based upon 3 repeated measurements in each condition, and differences between the soundfield and headphone data varied between approximately .5 dB and 2.5 dB, depending upon the condition. SSRTs in quiet are elevated in the headphone system, most likely because of a higher noise floor caused by the instrumentation. SSRTs in noise are all lower in the headphone system, attributable to the elimination of room and speaker effects. Improvements in SSRTs with spatial separation of the signal and masker were 6.38 dB in the soundfield and 6.65 dB under headphones. [Work supported by Hoover Foundation.]

Improved Estimates of Aided Thresholds Using Probe-Tube Measurements

Lawrence J. (Larry) Revit
Etymonic Design Incorporated, Canada

Knowing (or estimating) aided sound-field hearing thresholds can be helpful in hearing aid fitting, such as in estimating aided sensation level or aided articulation index. To estimate aided sound-field thresholds using real-ear probe-tube microphone measurements, the functional equivalence of insertion gain and functional gain, for linear hearing aids, gives:

\[ \text{AHTLSF} = \text{UHTLSF} - \text{REIG}, \]

where \( \text{AHTLSF} \) is the estimated aided sound-field threshold (in dB HL), \( \text{UHTLSF} \) is the unaided sound-field threshold (in dB HL), and \( \text{REIG} \) is the real-ear insertion gain, which is the real-ear unaided response (REAR) minus the real-ear unaided response (REUR). Some researchers (e.g., Mueller and Killion, 1992) have suggested that the above relation can be applied to predicting aided sound-field thresholds from earphone thresholds, by:

\[ \text{AHTLSF} = \text{HTLEp} - \text{REIG}, \]
where HTL<sub>Ep</sub> is the hearing threshold (in dB HL) measured with earphones. However, when the REUR and/or the middle-ear impedance are different from the young-adult average, the assumed equivalence of earphone thresholds and sound-field thresholds can break down, causing the above relation to contain potentially considerable error. A mathematical proof is offered, showing that improved estimates of aided sound-field thresholds can be made from earphone thresholds by:

\[ AHTL_{SF} = HTL_{Ep} + D - (REAR - UR), \]

where HTL<sub>Ep</sub> is the hearing threshold using insert phones, D is the difference between the individual's real-ear-to-coupler difference (RECD) and the average RECD, and UR is the average REUR. The implications are 1] the use of individual REUR measurements is contra-indicated in estimating aided sound-field thresholds from insert-phone thresholds; and 2] when the middle-ear impedance is different from the normal adult average, individual measurement of the RECD can further improve estimates of aided sound-field thresholds. Applicability of the above relation to audiology using supra-aural earphones and to KEMAR insertion-gain measurements will also be discussed.

The Role of Voicing, Fundamental Frequency, Amplitude Envelope and Voiceless Information in the Identification of Intervocalic Consonants

Stuart Rosen,
Northwestern University
Andrew Faulkner and Kirsti Reeve,
University College London, United Kingdom

Two experiments were performed in order to determine the relative importance of various kinds of acoustic information in the identification of the 24 English consonants presented intervocically, both with and without lipreading. Video tape recordings of a single female speaker were used. There were 5 different lists, each of which consisted of 2 tokens of each of the 24 consonants in a random order, making for 48 trials. Each subject typically ran 10 lists in each condition, but only the last 6 lists were analyzed further. A short training session preceded testing the first time each condition was tested.

Experiment 1 consisted of 9 conditions, in which 4 different sound signals were used:
V - A fixed-frequency, fixed-amplitude signal indicating vocal fold vibration.

V(A) - as for V but with added amplitude envelope, derived from the original speech.

Fx - A fixed-amplitude signal whose periodicity followed the speaker's fundamental.

Fx(A) - as for Fx, but with an amplitude envelope added.

Each of these conditions was run with (L+) and without lip-reading, with a lip-reading alone condition (LA) making a total of 9. The following table presents the mean performance obtained, averaged over 6 sessions and 5 subjects.

<table>
<thead>
<tr>
<th></th>
<th>correct</th>
<th>manner</th>
<th>place</th>
<th>envelope</th>
<th>voicing</th>
</tr>
</thead>
<tbody>
<tr>
<td>L+Fx(A)</td>
<td>84</td>
<td>78</td>
<td>94</td>
<td>71</td>
<td>84</td>
</tr>
<tr>
<td>L+Fx</td>
<td>82</td>
<td>75</td>
<td>93</td>
<td>70</td>
<td>84</td>
</tr>
<tr>
<td>L+V(A)</td>
<td>82</td>
<td>74</td>
<td>95</td>
<td>68</td>
<td>84</td>
</tr>
<tr>
<td>L+V</td>
<td>77</td>
<td>68</td>
<td>93</td>
<td>61</td>
<td>83</td>
</tr>
<tr>
<td>LA</td>
<td>56</td>
<td>61</td>
<td>91</td>
<td>29</td>
<td>17</td>
</tr>
<tr>
<td>Fx(A)</td>
<td>16</td>
<td>33</td>
<td>25</td>
<td>43</td>
<td>62</td>
</tr>
<tr>
<td>Fx</td>
<td>16</td>
<td>33</td>
<td>25</td>
<td>44</td>
<td>66</td>
</tr>
<tr>
<td>V(A)</td>
<td>13</td>
<td>30</td>
<td>23</td>
<td>39</td>
<td>59</td>
</tr>
<tr>
<td>V</td>
<td>12</td>
<td>27</td>
<td>22</td>
<td>40</td>
<td>60</td>
</tr>
</tbody>
</table>

"Correct" refers to the overall percentage of consonants correctly identified by the subjects. The other four columns contain the information transfer percentages for various abstract features. Values with a common symbol in the same column (*, #, @) are indistinguishable statistically. Note that more information tends to lead to better performance. However, neither fundamental frequency variations, nor envelope, influence performance very much, although the two in combination can lead to significant improvements, at least when lip-reading, and for some features.
Experiment 2 was primarily concerned with the role of frication information and envelope. There were 3 different sound signals, presented with and without lip-reading, making for a total of 6 conditions. Apart from Fx(A) used above, the other two sound signals were:

Fx(A)+Nz - as for Fx(A) above, with a band of fixed-level noise present during periods of voiceless excitation.

Fx(A)+Nz(A) - as above, but with an amplitude envelope on the noise, as well.

With procedures as described for Experiment 1, the following table presents the mean performance obtained, averaged over 6 sessions and 5 subjects.

<table>
<thead>
<tr>
<th></th>
<th>correct</th>
<th>manner</th>
<th>place</th>
<th>envelope</th>
<th>voicing</th>
</tr>
</thead>
<tbody>
<tr>
<td>[L+Fx(A)+Nz(A)]</td>
<td>75*</td>
<td>72*</td>
<td>82*</td>
<td>77*</td>
<td>88*</td>
</tr>
<tr>
<td>[L+Fx(A)+Nz]</td>
<td>76*</td>
<td>73*</td>
<td>80*</td>
<td>82*</td>
<td>92*</td>
</tr>
<tr>
<td>[L+Fx(A)]</td>
<td>66*</td>
<td>60*</td>
<td>78*</td>
<td>63*</td>
<td>81*</td>
</tr>
<tr>
<td>[Fx(A)+Nz(A)]</td>
<td>26@</td>
<td>45@</td>
<td>27@</td>
<td>63@</td>
<td>77@</td>
</tr>
<tr>
<td>[Fx(A)+Nz]</td>
<td>24@</td>
<td>45@</td>
<td>24@</td>
<td>61@</td>
<td>72@</td>
</tr>
<tr>
<td>[Fx(A)]</td>
<td>19@</td>
<td>32@</td>
<td>24@</td>
<td>44@</td>
<td>61@</td>
</tr>
</tbody>
</table>

Again, although information about voiceless excitation can improve performance significantly, never does the addition of envelope on its own lead to statistically significant differences.

In short, the primary cues to consonant identification in these stimuli (which lack most of the spectral structure of real speech) have to do with the temporal on-off patterning of voicing and voicelessness. Neither amplitude envelope, nor fundamental frequency variations, affect performance a great deal. Both these features, however, are certain to play a much larger role in connected discourse, where prosodic information can play an important role.

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Experiments to Evaluate Beam Forming Digital Signal Processor Algorithms in Simulated and Real Time Wearable Applications

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AudioLogic, Inc., and University of Colorado


We will report a number of early clinical evaluation studies on hearing impaired subjects over the past 18 months.

Measures included: 1) use of an automated scoring version of the California Consonant Test developed by Terry et al (1992) which includes Reaction Time analyses; 2) a method of adjustment (m.o.a.) procedure using the Speech Intelligibility Rating materials developed by Cox and McDaniel (1984, 1989) requiring subjects to adjust the 'Off Beam' babble noise to a achieve a selected intelligibility rating (Schweitzer & Terry, 1994); 3) sentence intelligibility measures on Cochlear Implant subjects; 4) 'off line' ratings of noise interference as a 'value added' feature for existing hearing aids, and 5) Field Trial questionnaires.

The results are all support the continued development of the digital hearing aid and expanded evaluations.

Auditory Distance Measures and Hearing Aid Tests

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Institute for Telecommunication Sciences

Over the past few years, the Institute for Telecommunication Sciences has developed and tested objective measures of speech quality for
telecommunications devices. Current research is being directed toward perception based measures of the distance between two auditory stimuli. This auditory distance measure may be useful in the assessment of hearing aids. Several tests of hearing aids are proposed. These tests would allow evaluations of a hearing aid's ability to separate signals from noise in a variety of noise environments.

Spectral Enhancement Using a Two-Filter Model of Impaired Frequency Selectivity.

Michael A. Stone and Brian C.J. Moore
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Neural tuning curves can be described as comprising two shapes: a narrow 'tip' filter with high sensitivity, and a wider 'tail' filter with lower sensitivity. When cochlear function is impaired, the sensitivity of the tip filter decreases. The tail filter characteristics are difficult to measure in normal-hearing subjects using psychophysical techniques such as the notched-noise method. However, with mild to severe cochlear hearing loss, the tail filter has a greater influence on the results, and becomes easier to measure. Auditory filter shapes were estimated using the notched-noise method over a wide range of center frequencies for subjects with differing degrees of hearing loss. These data were used to construct a model that predicts typical filter shapes as a function of frequency and hearing loss. The model is being used to improve the accuracy of a spectral contrast enhancement method intended to compensate for the effects of reduced frequency selectivity. The method is similar to one also being developed in this laboratory [Baer T., oral presentation this meeting], but uses a filter bank rather than overlap-add FFTs. Results of intelligibility tests using speech in speech-shaped noise, enhanced by this method and tested on hearing-impaired subjects, will be described. (Work supported by the MRC (UK), travel grant from Royal Society (UK)).

An Adaptive, Single Filter Algorithm for Mapping Loudness Contours in Digital Hearing Aids

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Typical hearing aid strategies involve amplification, some filtering, and dynamic range compression (Dirks 1992). Amplification raises the signal above the increased hearing threshold; filtering with either a filter bank or a fixed single filter attempts to match the frequency-dependent hearing loss, or to emphasize the parts of the speech signal with more relevant information, or both; and wide-band or multi-band dynamic range compression reduces the output so that high intensity inputs are not amplified above the maximum comfort level.

We have developed a configurable software simulation environment for comparing a matrix of the possible architecture described above. That is, we simulate uniform, linear single-filtered, and multi-band amplification, each followed by either wide or multi-band compression (six total algorithms). This environment also provides graphical time, frequency, and energy data for all signals; generates shaped noise to emulate a given hearing loss; and allows for quick A/B play-back comparison.

Hearing aids that use multi-band compression typically involve amplifying and compressing each band separately before mixing the sub-bands (Gilman 1986). Sub-band separation can lead to unnecessary inter-band discontinuity artifacts. In this study, signal shaping and compression algorithm choices are independent. As mentioned above, shaping can be achieved by multi-band, single filter, or uniform amplification, and compression can be either multi-band and wide-band. Initial experimentation suggested uniform amplification followed by multi-band compression as the most promising approach. This algorithm, however, does not directly account for frequency-dependent losses.

Therefore, in an attempt to map impaired constant loudness contours from normal ones, we have developed an algorithm that involves mixing a phase-aligned direct (unity gain) version of the input with a single-filter, linearly amplified version. The mixture is adaptive based on the intensity of the input signal. Lower intensity inputs cause the mixture to favor the linearly-filter version, while higher intensity inputs pass largely unaffected. Our implementation has a hardware complexity cost of a single 16-64 tap FIR filter, an 8-32 state delay pipe, two output multipliers, an adder, and a total energy estimator.

Compared to compression-based approaches, this algorithm noticeably reduces undesirable background noise, and possibly acoustic feedback and amplifier clipping. Preliminary evaluation suggests that continuous mapping of loudness contours without inter-band discontinuities improves intelligibility and naturalness, especially in naturally noisy and dynamic environments. Using shaped noise to emulate a moderate, gradual slope hearing loss, and samples of the HINT sentences (Nilsson, Soli, Sullivan 1994) recorded in naturally noisy environments, our first subject correctly identified 32%, 63% and 85% of the words with the uniform amplified/wide-band compressed, uniform amplified/multi-
band compressed, and our single filter mixture algorithms, respectively. The "naturally noisy" recordings were made in a car on a freeway, and at the beach. Recordings and processing are stereo.

We will also present results comparing these different schemes in a reverberant cocktail party environment. Both hearing impaired and masked normal hearing subjects will participate.

A Framework for Assessing Speech Reception with Binaural Hearing Aids

Jean Sullivan, Donna Gelnott, and Sigfrid D. Soli
House Ear Institute

This poster will describe a framework for assessing speech reception within which the researcher or clinician can judge the merits of binaural hearing aids. The framework includes a speech reception measure, a set of signal presentation conditions, and a set of listening conditions. We use the Hearing In Noise Test (Nilsson, Soli and Sullivan 1994) to measure sentence, speech-reception thresholds (SSRT) in quiet and in noise. Thresholds in noise are obtained with and without spatial separation of the speech and noise signals. The speech signal originates from a 0° azimuth source and the noise from one of three sources on the horizontal plane: 0°, 90°, or 270°. Listeners are tested unaided, with the test binaural hearing aids, and with filtered speech presented to headphones. The headphone filters are designed to compensate for a listener's loss of sensitivity and to simulate soundfield listening with and without spatial separation of speech and noise signals. We refer to this condition as the binaural "capacity" condition.

We will describe how we create a performance profile for a listener by comparing thresholds for the test binaural hearing aids to thresholds in each of the test conditions. Differences between aided and binaural "capacity" thresholds indicate if the listener has reached or exceeded his speech reception potential. Differences between aided and unaided thresholds indicate whether the listener benefits from the hearing aids. Finally, differences between and aided normal-hearing thresholds indicate whether the listener will experience a loss of speech reception while wearing the hearing aids. We also examine threshold differences resulting from changes in the signal presentation condition. Differences between thresholds with and without spatial separation of the speech and noise sources indicate the degree to which the listener is able to take advantage.
of acoustic cues for binaural, directional hearing. This poster will display performance profiles and discuss our resulting conclusions for subjects tested in our laboratory.

Hearing Aid Using Digitally Processed Microphone Array

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In recent years, much work has been performed using fixed and adaptive acoustic beam formation array techniques to achieve directional gain in some preferred spatial direction and attenuation in some other directions. While the adaptive approaches have yielded impressive results in a free space condition, they do not necessarily perform well in reverberation and other realistic environments. For an array with a fixed number of sensors and taps, we would like to have the following desirable characteristics: a user steerable main-beam spatial "look-direction", a user selectable spatial attenuation band, a mainlobe energy concentration region given the imposed constraints, a user imposed frequency domain characteristic, and with these desirable properties valid over a large frequency band. Now, we consider an optimum equally spaced linear array that possesses these properties. The array has R sensors with L taps per sensor and it maximizes energy concentration over some desired spatial "look" region and frequency band subject to user imposed spatial and frequency attenuation constraints. This problem is first expressed as a constrained maximization of the form \( w^*Aw/w^*Bw \), where A and B are specified in terms of parameters specified in the spatial and frequency domains, w is the array weight vector, and the constraining subspace is specified by the array response values, derivative values, and source spatial locations. Then upon two transformations, the optimum array weight vector is obtained as the solution of an unconstrained full-rank lower-dimensional generalized eigenvalue problem. Numerical examples for several cases that are practical for hearing aid are considered to illustrate the usefulness of this approach. Finally, simulation results based on free-space sound wave propagation confirm the results obtained from analysis.
Discrimination of Vowels and Stop Consonants After Multichannel Compression.

E. William Yund and Thomas R. Crain
Veterans Administration Medical Center

The effect of full-range multichannel compression (MCC) on vowel and stop consonants discrimination was studied. For normal hearing subjects MCC was fit to four hypothetical flat losses, with thresholds ranging from 60 to 90 dB SPL, and one hypothetical sloped loss, with thresholds normal at 500 Hz and 90 dB SPL at 4 kHz. Hearing impaired subjects were tested with flat MCC as well as MCC fit to the subject’s hearing loss. Compression ratios for the flat compression ranged from 1.75 to 7.00. Numbers of channels in both flat and fitted MCC ranged from 2 to 3. Robinson-Huntington compression [C.E. Robinson and D.A. Huntington, J. Acoust. Soc. Am. 54, 314 (1973)] with 10-msec symmetrical time windows was used throughout. Unprocessed stimuli and frequency-equalized linear amplification (NAL) were control conditions. There were no deleterious effects of MCC on discrimination of voice onset time (VOT). Normal hearing subjects had difficulty with vowel and stop consonant place discrimination for the most severe MCC conditions. Hearing impaired subjects showed more difficulty with vowel discrimination for flat MCC than did the normal hearing subjects, but no more difficulty for fitted MCC than for the control conditions. Hearing impaired subjects needed either fitted MCC or frequency-equalized linear amplification to perform consistent place discriminations on these synthetic vowel-consonant-vowel (VCV) syllables, and showed no differential effect for MCC versus linear or for the number of channels in the MCC. These results indicate that there is little reason to be concerned about the negative effects of full range MCC fit to the hearing impairment, as long as the number of channels does not exceed 31 (the maximum number tested in the experiments). [Supported by Department of Veterans Affairs.]