The International Hearing Aid Research Conference 2000

August 23 - 27, 2000

Granlibakken Conference Center
Lake Tahoe, California

The International Hearing Aid Research Conference 2000

Sponsored

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House Ear Institute

National Institute on Deafness and other Communication Disorders

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International Hearing Aid Research (IHEAR) Conference 2000

Planning Committee

Technical Chair

Dianne Van Tassel
Starkey Laboratories, Inc.
6600 Washington Avenue South
Eden Prairie, MN 55344

Technical Co-Chairs

Stuart Gatehouse*
MRC Institute of Hearing (Scottish Section)
Queen Elizabeth Building
Royal Infirmary
Glasgow, Scotland G31 2ER

Robyn Cox
University of Memphis
Speech and Hearing Center
807 Jefferson Avenue
Memphis, TN 38105

Organizational Co-Chairs

Sigfrid Soli
House Ear Institute
2100 West Third Street, 5th Floor
Los Angeles, CA 90057

Lynn Luethke
National Institute on Deafness
and other Communication Disorders
6120 Executive Blvd., MSC 7180
Bethesda, MD 20892-7180

**Stuart Gatehouse replaces Denis Byrne who passed away earlier this year.*

Scholarship Recipients

Name	Institution	Field of Study
Ben Hornsby	Vanderbilt University	Audiology
Michael Lockwood	University of Illinois	Electrical/Computer Engineering
Kumiko Boike	University of Washington	Speech/Hearing Science
Ayaskanta Rout	Purdue University	Audiology
Sarah Hargus Ferguson	Indiana University	Speech/Hearing Science
Judith Reese	University of South Florida	Audiology
Mareen Coughlin	Indiana University	Speech/Hearing Science
Rupa Balachandran	CUNY Graduate School	Speech/Hearing Science
Koushik Narayan	Sri Venkateshwara College	Engineering
Lisa Huettel	Duke University	Electrical/Computer Engineering
Peninah Rosengard	Harvard/MIT	Speech/Hearing Science
Virginia Kitch	University of Washington	Speech/Hearing Science
Rachel McArdle	University of South Florida	Audiology/Experimental Psych.
Regina Nuzzo	Stanford University	Biostatistics
Melissa Ruscetta	University of Pittsburgh	Audiology
Joong-Seok Moon	University of So. California	Electrical Engineering
Sharba Bandyopadhyay	Johns Hopkins University	Biomedical Engineering
Kevin Munro	University of Southampton	Hearing Science
Martina Huss	Cambridge University	Experimental Psychology
Bas Franck	Academic Medica	Experimental Audiology
Marion Schabert	University Erlangen-Nuremberg	Telecommunications

Daily Schedule

Wednesday, August 23

5:00 PM	Opening Social
6:30 PM	Dinner
7:30 PM	Opening remarks
7:45 PM	Evening session

Thursday, August 24

7:00AM	Breakfast
8:00 AM	Morning session
9:45 AM	Poster Session
12:20 PM	Lunch
5:00 PM	Social/Poster Session Continues
6:00 PM	Keynote Address
7:00 PM	Dinner
8:00 PM	Event in honor of Denis Byrne

Friday & Saturday, August 25-26

7:00AM	Breakfast
8:00 AM	Morning session
9:45 AM	Poster Session
12:20 PM	Lunch
5:00 PM	Evening session
7:00 PM	Dinner
8:20 PM	Social/Poster session continues

Sunday, August 27

8:00 AM	Breakfast and checkout
9:00 AM	Morning session
12:00 PM	Adjournment (busses leave for airport with box lunches for passengers)

Program Summary

Wednesday, August 23

Session One

7:45PM-10:05PM

Hearing Aids and Cochlear Physiology

Moderator: Jont Allen

Brian C. J. Moore	Functional consequences of hair cell damage
Charles M. Liberman	Masking and anti-masking in the auditory periphery: A neurophysiological perspective
Christopher Turner	Providing speech information to persons with sensorineural hearing loss
Graham Naylor	Combination tones in ears with precipitous high-frequency losses

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Thursday, August 24

Session Two

8:00AM-12:00NOON

Compression: Revisited

Moderator: Harvey Dillon

Inga Holube	Multi-channel dynamic compression: Concepts and results
Martin Hansen	Effect of multiband compression time constants on subjectively perceived sound quality and speech recognition
William Woods	Computational analysis of several measures of multichannel compression performance as a function of a number of independent compression channels
David D. Anderson	Time constants in multi-band compressive gain hearing aids
Benjamin Hornsby	The interactive efforts of multichannel compression, presentation level and signal-to-noise ratio on speech recognition in normal-hearing subjects.
Allen F. Ryan	**Keynote Address** Future directions in the prevention and treatment of hearing disorders.

Friday, August 25

SESSION THREE

8:00AM-11:00AM

Feedback

Moderator: Carl Ludvigsen

Johan Hellgren	Feedback cancellation in hearing aids with Filtered-X LMS and direct closed loop identification
James M. Kates	Reverberation effects in feedback cancellation
Brent Edwards	Clinical finding on the user of digital feedback suppression

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Friday, August 25

SESSION FOUR

11:00AM-12:00NOON

Fitting

Moderator: Arlene Newman

Teresa Ching	Strategies for optimizing hearing aid fitting of hearing-impaired children
Claus Elberling	Fitting hearing aids by listening preference

Friday, August 25

SESSION FIVE

5:00PM-800PM

Speech Processing in Hearing and in Hearing Aids

Moderator: Pamela Souza

Eric D. Young	Insights into hearing aid signal processing from the neural representation of speech
Ayaskanta Rout	Effects of spectral enhancement on the speech recognition performance of individuals with cochlear hearing loss
Patricia Stelmachowicz	The effect of stimulus bandwidth on fricative perception in normal-hearing and hearing impaired children and adults.

Saturday, August 26

SESSION SIX

8:00AM-11:00

Alternative Therapies for Sensorineural Hearing Loss

Moderator: Lynn Luethke

Robert Shannon	Holes in hearing: Implications for cochlear implants and hearing aids
Sigfrid Soli	Direct mechanical stimulation of the ossicles via middle ear implant devices
David Preves	Technical considerations for a disposable hearing aid

Saturday, August 26

SESSION SEVEN

11:00PM-12:00NOON

Binaural Hearing and Auditory Grouping

Moderator: Monica Hawley

Volker Hohmann	Binaural noise reduction and a localization model based on the statistics of binaural signal parameters
Chris Darwin	Auditory grouping and attention

Saturday, August 26

SESSION EIGHT

5:00PM-7:00PM

Outcome Measures

Moderator: Arthur Boothroyd

Robyn Cox	Measured Satisfaction: What does it tell us?
Ruth Bentler	Impact of digital labeling on outcome measures
Stuart Gatehouse	Aspects of auditory ecology and psychoacoustic function as determinants of benefits from and candidature for non-linear processing in hearing aids

Sunday, August 27

SESSION NINE

8:00AM-10:00

Hearing Aids: Issues and Innovations

Moderator: Michael Nilsson

Anne Strouse	Listeners who prefer monaural to binaural hearing aids
Wouter A. Dreschler	Active noise reduction in digital hearing aids
Peter Nordqvist	Automatic classification of different listening environments in a generalized adaptive haring aids.

Sunday, August 27

SESSION TEN

10:00AM-12:00NOON

Engineering Issues and Innovations

Moderator: Steven Thompson

R.N. Miles	A biologically inspired directional microphone concept for hearing aids
Michael E. Lockwood	Simulation and real-time implementation of a new optimum beamforming technique
Arne Leijon	Finite-state modeling for specification of non-linear hearing instruments

Detailed Program
Wednesday, August 23

SESSION ONE

Hearing Aids and Cochlear Physiology

Moderator: Jont Allen

7:30 PM Introductory comments

7:45 PM Functional consequences of hair cell damage

Brian C. J. Moore, Department of Experimental Psychology,
University of Cambridge, United Kingdom

This presentation will review the role of inner hair cells (IHCs) and outer hair cells (IHCs) in auditory perception and the effects of damage to those hair cells. The IHCs transduce mechanical vibrations in the cochlea into neural activity. The OHCs have several functions: they enhance the sensitivity to weak sounds by increasing the response on the basilar membrane (for frequencies close to the characteristic frequency, CF); they increase the sharpness of tuning on the basilar membrane; they produce nonlinear (compressive) input-output functions for frequencies close to CF; and they contribute to the production of combination tones.

Damage to the IHCs produces: less efficient transduction of mechanical vibration into neural activity; loss of sensitivity (threshold elevation); reduced information flow in the auditory nerve ("noisy transmission"); and, in extreme cases, no transduction of activity at some regions of the basilar membrane ("dead regions"). Damage to the OHCs produces: reduced sensitivity to weak sounds (threshold elevation); reduced sharpness of tuning on the basilar membrane, which results in loss of frequency selectivity as measured behaviourally; more linear input-output functions on the basilar membrane, which results in loudness recruitment; and reduction of combination tones. Note that the commonest clinical measure of hearing impairment, threshold elevation, can arise from either IHC or OHC damage, or a mixture of the two.

Recently, we have developed a test for detecting dead regions and defining their limits (Moore, B. C. J., Huss, M., Vickers, D. A., Glasberg, B. R., and Alcántara, J. I. (2000). "A test for the diagnosis of dead regions in the cochlea," Br. J. Audiol., in press). A dead region is defined in terms of the range of CFs that would normally be associated with the dead region. The test involves measurement of the threshold for detecting pure tones in a spectrally shaped broadband noise designed to produce equal masked thresholds at all frequencies. If the masked threshold is 10-dB or more higher than the mean value obtained for normally hearing subjects, and there is at least 10 dB of masking, this is taken to indicate a dead region. The results indicate that the presence or absence of dead regions cannot be reliably inferred from the audiogram, although if the hearing loss is greater than 80 dB at high frequencies this is nearly always associated with a dead region. Experiments using speech lowpass filtered with various cutoff frequencies (and amplified with a frequency-gain characteristic appropriate for the hearing loss) indicate that, for people with high-frequency dead regions, speech identification is usually better with lowpass filtered speech than with broadband speech. However, optimum performance is obtained when the filter cutoff frequency is slightly above the boundary of the dead region (see the poster presentation by Vickers et al.). This suggests that some useful information can be obtained from frequencies falling a little inside a dead region.

Providing speech information to persons with sensorineural hearing loss Christopher Turner, Bruce Gantz and Richard Hurtig, University of Iowa. One of the goals of our laboratory has been to characterize the effectiveness of amplification for patients with various degrees of sensorineural hearing loss, and also to investigate possible solutions for cases where amplification does not work. The ability of the impaired ear to transmit amplified speech information to the brain is dependent upon the frequency region and degree of the hearing loss, and the types of speech cues that reside in the various frequency regions of speech. In general, severe hearing loss compromises the ability of the ear to transmit the speech cues located in the higher frequency regions of the spectrum. For cases where amplification of speech is not effective, proportional frequency compression of speech has shown some benefits. More recently, we have been working with a "short-electrode" cochlear implant, which allows patients to use their residual low-frequency acoustic hearing and obtain high-frequency speech information by way of electrical stimulation.

8:20 PM MASKING AND ANTI-MASKING IN THE AUDITORY PERIPHERY: A NEUROPHYSIOLOGICAL PERSPECTIVE

Charles M. Liberman, Eaton-Peabody Laboratory,
Massachusetts Eye and Ear Infirmary, Boston

A great deal has been learned about the neurophysiological basis for noise masking of the response to tone signals in the auditory nerve, and about the possible roles of the middle-ear muscle and olivocochlear reflexes in the control of masking. The masking of auditory nerve responses by noise includes both suppressive and excitatory components. Suppressive masking arises by mechanisms similar to two-tone suppression and thus originates in non-linear mechanics within the cochlear partition. Suppressive masking can decrease response to a signal without evoking discharge, and is the primary component underlying the upward spread of masking. Excitatory masking has two aspects: the classic "line busy" effect and an additional effect due to adaptation. The latter component arises at the hair-cell synapse. Recent neurophysiological studies suggest that the middle-ear muscle reflex can greatly reduce the suppressive masking of high-frequency neurons by low-frequency noise, and that the olivocochlear reflex can reduce the excitatory masking of high-frequency neurons by high-frequency noise. The effects of hair cell damage on masking have not been well studied at the level of the auditory nerve. However, speculations will be offered based on normal data and known alterations in other basic response properties associated with damage to inner or outer hair cells.

8:55 PM PROVIDING SPEECH INFORMATION TO PERSONS WITH SENSORINEURAL HEARING LOSS

Christopher Turner, Bruce Gantz and Richard Hurtig
University of Iowa

One of the goals of our laboratory has been to characterize the effectiveness of amplification for patients with various degrees of sensorineural hearing loss, and also to investigate possible solutions for cases where amplification does not work. The ability of the impaired ear to transmit amplified speech information to the brain is dependent upon the frequency region and degree of the hearing loss, and the types of speech cues that reside in the various frequency regions of speech. In general, severe hearing loss compromises the ability of the ear to transmit the speech cues located in the higher frequency regions of the spectrum. For cases where amplification of speech is not effective, proportional frequency compression of speech has shown some benefits. More recently, we have been working with a "short-electrode" cochlear implant, which allows patients to use their residual low-frequency acoustic hearing and obtain high-frequency speech information by way of electrical stimulation.

9:30 PM Combination tones in ears with precipitous high-frequency losses

Graham Naylor
Oticon Research Centre, Denmark

This paper reports preliminary results of some psychoacoustical measurements carried out on ears having near-normal thresholds at low frequencies and steeply rising thresholds above some 'corner' frequency. A battery of psychoacoustical tests is being developed, with the aim of distinguishing sub-groups within the broad 'ski-slope' category of losses. Such sub-groups may well benefit from differing amplification approaches, and with luck their distinctive psychoacoustical characteristics may indicate appropriate differences in amplification.

The two measurements which are discussed here are forward-masked psychoacoustical tuning curves (PTCs) and simultaneously-masked masked thresholds. PTCs show clear occurrences of off-frequency listening and broadened filters at high frequencies, often somewhat broadened tuning at low frequencies too, and generally rather sharp tuning at the corner frequency. Masked thresholds for high-level maskers show considerable upward spread of masking (USOM) for maskers at low frequencies, but maskers at frequencies above the corner frequency often appear to produce little, no or even strongly 'negative' USOM. Evidence from supplementary experiments will be presented, supporting the contention that the explanation is to be found in aural combination tones.

The balance of opinion in the literature seems to be that combination tones are weakened or absent in the damaged cochlea, but the present data suggests that they can occur, as long as the combination frequency is within a region of near-normal threshold. Furthermore, both primary tones need not be above quiet threshold for the combination tone to be audible. The results will be discussed both with respect to theories of how combination tones are generated and for their potential implications in a hearing aid context.

Thursday, August 24

SESSION TWO

Compression: Revisited

Moderator: Harvey Dillon

8:00 AM Multi-channel dynamic compression: Concepts and results

Inga Holube, Matthias Wesselkamp and Volkmar Hamacher
Siemens Audiologische Technik GmbH, Erlangen, Germany

Most digital hearing instruments offer dynamic compression in several frequency channels (multi-channel dynamic compression). However, some of the implemented concepts differ substantially in the number of compression channels and the applied time constants. These parameters influence the spectral and temporal structure of processed speech signals. Therefore, this paper focuses the effects of different combinations of the number of compression channels and the time constants on features of speech and speech intelligibility. An evaluation is presented which investigates the effect of compression parameters based on speech intelligibility predictions as well as on measurements with hearing-impaired listeners. The results of the investigation and calculations are analyzed and discussed with findings from current research.

8:35 AM Effect of multiband compression time constants on subjectively perceived sound quality and speech recognition

Martin Hansen
Widex ApS, Audlab, Vaerloese, Denmark

A common observation with sensorineural hearing loss is the recruitment phenomenon, i.e., the occurrence of a steeper-than-normal loudness growth function, together with an elevated absolute threshold. The typical means of compensating for this recruitment in a hearing instrument is a non-linear compressor circuit. The general aim of the compressor is to provide higher gain for softer sounds than for louder sounds.

Two important parameters of the compressor are its attack and release times, which result from the time constants of the input level estimator. Most commercially available hearing instruments use relatively short time constants ("syllabic compression"). This is in line with physiological data on outer haircell function and basilar membrane motion (Ruggero & Rich, 1991), which indicate the existence of a very fast active compression in the healthy cochlear, as well as with loudness models and the relation of perceived loudness of short versus long sounds (Buus,1999). The shorter the time constants of the compressor are, the more closely will the actual sound reproduction follow the intended non-linear characteristics. On the other hand, compressors with fast time constants may have a negative influence on the sound quality of the system, e.g., due to an increased amplification of the general background noise.

This paper presents measurements on the influence of attack and release times of a multiband compression hearing instrument on the subjectively perceived sound quality and speech intelligibility for normal hearing and hearing impaired subjects. One aim was to find the optimal parameter adjustment empirically. To this end, a hearing instrument with a third-octave filterbank and independent compressors in each band was simulated. The compression ratios and compression

kneepoints were kept constant during this experiment, while the gain for a certain input level was adjusted individually according to the hearing loss, measured by in-situaudiometry, of each subject. Four different settings for the attack and release times were investigated, ranging from fast (1-10 ms) to slow (4 s) time constant settings. Binaural recordings of acoustic environments, exhibiting real-life signal-to-noise ratios, and several music signals were used as test stimuli. Subjective preference with respect to sound quality and speech intelligibility was assessed in a complete paired comparison paradigm.

The results show a clear preference for longer release times (4 s) compared with shorter release times (40 ms, 400 ms), both for normal hearing and hearing impaired subjects. The effect of the attack time (1 ms vs. 100 ms) is less pronounced when the release time is long. The preference for a certain time constant setting is more clearly marked when speech signals are assessed, compared to non-speech and music signals. These empirical results contradict several theoretical considerations which would suggest that a fast-acting non-linear multiband signal processing be optimal for restoring loudness perception in the healthy cochlea. A fast-acting recruitment compensation may, therefore, not be the most important goal in reaching an optimal ("most satisfactory") improvement of the hearing of sensorineurally hearing impaired persons.

9:10 AM Computational analysis of several measures of mutlichannel compression performance as a function of a number of independent compression channels

William Woods, Martin Rickert and Dianne Van Tassell
Starkey Laboratories, Eden Prairie, Minnesota

The number of independent channels of compression available in modern digital hearing aids currently varies from one to twenty. The number of channels that can be realized in a DSP aid is constrained primarily by integrated circuit size, power, and signal-processing complexity. Although it is intuitive that multiple channels of compression increase flexibility in fitting steep or irregular audiometric configurations, the question addressed in this computational analysis is: How many channels are actually required to achieve asymptotic levels on several objective performance measures? For a set of representative audiograms, the effect of the number of compression channels was estimated according to each of three quantitative criteria: 1) maximum achievable Speech Intelligibility Index (SII) [ANSI S3.5–1997, Methods for Calculation of the Speech Intelligibility Index]; 2) SII achieved by using the Cambridge non-linear loudness-based prescriptive fitting method [Moore et al. (1999) Use of a loudness model for hearing aid fitting. II: Hearing aids with multi-channel compression, British J. Audiol, in press]; and 3) closeness of achievable match to the Cambridge prescriptive fitting target. For all criteria, asymptotic predicted performance was achieved with fewer channels if the system had variable rather than fixed compression channel bandwidths. In some instances, systems with fixed, equal logarithmic spacing of channels actually showed worse predicted performance with increasing number of channels, due to interactions between channel bandwidths and specific audiometric configurations. The biggest gains in predicted performance were associated with increase from one channel to two (variable) channels. Asymptotic predicted performance was always achieved with five or fewer variable channels of compression, and usually with three or fewer. The small number of compression channels needed to optimize performance is likely related to the low dimensionality of the audiogram space, as revealed by principal component analysis of the representative audiograms.

11:10 AM Time constants in multi-band compressive gain hearing aids

David D. Anderson, Georgia Institute of Technology
Douglas M. Chabries & Richard W. Christiansen, Brigham Young University, Utah

Hearing aids that have a compressive or non-linear gain are often characterized using time constants. The attack and release time of the amplifier in a hearing aid give an indication of how rapidly the gain is adapted to changing input levels. If the gain is adapted too quickly, annoying artifacts and audible distortion corrupt the sound. If the gain is adapted too slowly, the sound levels presented to the listener may be painfully loud or too quiet. Multi-band hearing aids present many more options than single band aids and are difficult to characterize with the simple time-constant model. In particular, each band may have a different gain adaptation rate or method. This paper presents an approach to controlling the adaptive gain in each band of a multi-band hearing aid based on a homomorphic signal processing analysis of a perceptually based audio representation. The gain adaptation rate in each band is easily determined in a way that implicitly takes into account the critical bands of hearing. The resulting gain adaptation introduces little or no audible distortion yet it is very fast, delivering appropriate listening levels for stimuli which may vary dramatically in intensity. This new algorithm has been used in numerous tests and is similar to that used in the Natura(tm) hearing aid by Sonic Innovations, Inc. Finally, a framework for analyzing and describing hearing aids using the new gain adaption approach is presented.

11:45 AM The interactive effects of multi-channel compression, presentation level and signal-to-noise ratio on speech recognition in normal-hearing subjects

Benjamin Hornsby and Todd Ricketts
Vanderbilt University, Nashville

To improve speech understanding and combat the effects of reduced audibility and loudness recruitment, persons with hearing loss frequently listen to amplified speech, processed through hearing aids which incorporate fast-acting, multiple-channel, wide dynamic range compression (WDRC). Previous research has demonstrated, however, that speech recognition may worsen when the speech is presented at higher-than-normal levels. This is particularly true when the speech is presented in the presence of speech-shaped background noise (Studebaker et al., 1999). In addition, the use of fast-acting, multichannel compression, may reduce spectrotemporal cues important for speech recognition (Plomp, 1994).

What is not clear is how these factors (Signal-to-noise ratio (SNR), level and compression) may interact to further affect speech recognition. The primary objective of this study was to examine the interactive effects of SNR, speech presentation levels, and compression on consonant recognition in noise. To meet this objective we had subjects with normal hearing identify compressed and uncompressed CV and VC nonsense syllables in a background of speech-shaped noise (0 and + 6 dB SNR's) at three presentation levels (65, 80 and 95 dB SPL). A simulated 3- channel, fast-acting, wide dynamic range hearing aid, utilizing compression ratios of 1:1, 2:1, 4:1 and 8:1, was used to compress speech stimuli. Consonant recognition performance decreased as compression ratio increased and presentation level increased. Interaction effects were noted between SNR and compression ratio and presentation level and compression ratio. Performance decrements due to increases in compression ratio were larger at the better (+6 dB) SNR and at the lowest (65 dB SPL) presentation level. At higher levels (80-95 dB SPL), such as those experienced by persons with hearing loss, the negative impact of increasing compression ratio was minimal, particularly at a poorer SNR. Results will also be discussed in terms of feature transmission as a function of level and compression ratio.

6:00 PM Keynote address:

Future directions in the prevention and Treatment of Hearing Disorders

Allen F. Ryan
University of California, San Diego

Hearing aids are currently the treatment of choice for hearing disorders doe to loss of hair cells and primary auditory neurons, and are likely to remain so for the foreseeable future. For this reason, the improvement of existing hearing aid strategies and the development of new aids are important goals. However, recent discoveries in the cellular and molecular biology of the inner ear offer the promise of new therapeutic avenues. For example, survival programs that counteract the effects of damaging stimuli can be activated in hair cells and cochlear neurons by growth factors. Intracellular events such as G-protein and tyrosine kinase signaling, the action of free radicals, and glutamate toxicity have been shown to mediate aspects of hair cell and neuronal damage. The identification of these events has led to methods for the prevention of hearing loss in experimental models, and could result in treatments for patients in the relatively near future.

Another source of potential new treatments for hearing loss has emerged from molecular genetic studies. Genes that influence the development and function of the inner ear, and that when mutated produce inherited forms of hearing loss, are being identified at an accelerating rate. Understanding the molecular substrates of hearing may allow us tomanipulate genetically abnormal and damaged cells in innovative ways. Application of gene therapy to the inner ear can be used to correct genetic defects or to deliver protective substances to cochlear cells. Finally, molecular biological techniques have been developed with which to pursue new avenues of research. The identification of gene promoters specific to hair cells permit a variety of studies not possible in the past. For example, the identification of factors that control gene expression in hair cells may lead to techniques for inducing cells to adopt the hair cell phenotype. Hair cell-specific expression of in vivo markers also allows us study living hair cells during the process of cell damage and death. They also permit the transfer of cells between normal and damaged sensory epithelia, which may lead to methods for hair cell transplantation. (Supported by grant DC00139 from the NIH/NIDCD, the Research Service of the VA, and by the NOHR.)

Friday, August 25

session THREE

FEEDBACK

Moderator: Carl Ludvigsen

8:00 AM Feedback cancellation in hearing aids with filtered-X LMS and direct closed loop identification

Johan Hellgren
Technical Audiology Linköpings universitet, University Hospital
Linköping, Sweden

The negative effects of the acoustic feedback can be reduced using feedback cancellation. An estimate of the feedback path is then used together with the output signal to obtain an estimate of the input signal without feedback. The main problem with this kind of signal processing is to find a good estimate of the feedback path without modifying the output of the hearing aid such that the sound quality of the hearing aid is reduced. In this presentation feedback cancellation with Filtered-X LMS and direct closed loop identification is analyzed. Filtered-X LMS is a modification of the LMS algorithm, where known characteristics of the feedback path can be incorporated in the model by a fixed filter. This can be useful as some of the characteristics of the feedback path are relatively constant (e.g. resonances of the receiver). In direct closed loop identification the output and input signals of the hearing aid are used together with an open loop model to find an estimate of the system, even if the data is collected in closed loop. A benefit with this approach is that the feedback can be identified continuously without the additive noise on the output signal required in some other approaches. A drawback is that, depending on the characteristics of the input signal, a bias in the estimate may be introduced. An analytic expression of the estimate that recursive identification algorithms with a quadratic norm (e.g. Least Mean Squares, Filtered-X LMS, and Recursive Least Squares) will converge towards is presented. It is also shown that the bias can be avoided by prefiltering, if the spectrum of the input signal to the system (without feedback) is known. It is however not possible to both identify the feedback path and the spectrum of the input signal from the used signals, if the hearing aid is linear and no noise is added to the output. The tracking characteristics (speed of adaptation and variance of estimate) of Filtered-X LMS when used with direct closed loop identification have been analyzed. This approach has the advantage that the error at frequencies where the hearing aids is about to oscillate is emphasized, resulting in higher adaptation speed and reduced bias at these frequencies. Alternative choices of the fixed filter have been evaluated regarding the bias that can be expected when used in daily life. An individually adjusted fixed filter can be assumed to perform best.

8:35 AM Reverberation effects in feedback cancellation

James M. Kates
AudioLogic, Boulder, CO

Room reverberation has a strong effect on feedback cancellation in hearing aids. These effects were studied using a behind the ear (BTE) hearing aid mounted on a dummy head and coupled to the ear canal via an open fitting. The impulse response of the hearing aid feedback path was measured for eight closely-spaced locations in a typical office. The feedback cancellation in the hearing aid used a

set of filter coefficients that were initialized for one location within the room, and then allowed to adapt to the feedback path measured at the same or to a different location. The maximum stable gain for the hearing aid was then estimated without feedback cancellation, for the initial set of feedback cancellation filter coefficients prior to adaptation, and for the feedback cancellation filter after adaptation.

The impulse response measurements show that the feedback path impulse response can be divided into a short-time portion consisting of the ear-level factors of the microphone, receiver, ear canal, vent, and pinna resonances, and into a long-time portion consisting of the room reverberation. The short-time portion of the feedback path response can be accurately represented using a low-order ARMA model. Increasing the adaptive filter length had only a small benefit in improving the feedback cancellation performance due to the inability of the system to model the room reverberation. The mismatch between the modeled and actual feedback paths limits the headroom increase that can be achieved when using feedback cancellation, and varies with the location within the room.

9:10 AM Clinical findings on the use of digital feedback suppression

Brent Edwards and Laurel Olson
RxSound (formerly of GN ReSound)

Acoustic feedback imposes many difficulties on hearing aid wearers: (i) the amount of gain that can be provided by the hearing aid must be limited to prevent unstable oscillations or whistling, often limited to an amount below that which is prescribed for their hearing loss; (ii) certain actions by the wearer—such as bringing a phone to the ear, chewing or bending over—become a problem for the wearer because the actions change the characteristics of the acoustic feedback path and induce whistling. Solutions to these problems traditionally involve reducing the gain in the frequency regions of the feedback problems, typically resulting in reduced audibility of high-frequency cues. Adaptive feedback cancellation is one technique that attempts to solve these problems without limiting audibility by subtracting out the feedback signal from the microphone signal. This technique allows more gain to be provided in the hearing aid beyond the limit imposed by the feedback path, and its adaptive nature reduces the occurrence of whistling when the acoustics of the feedback path changes.

This talk will focus on clinical results obtained with GN ReSound's Digital Feedback Suppression (DFS) algorithm. This feedback cancellation technique estimates the transfer function of the feedback path when the hearing aid is first fit, and tracks changes to the acoustics of the feedback path during normal use of the hearing aid. Factors relating to the practical use of DFS will be discussed. These include the relationship between DFS, fitting algorithms and speech understanding. Issues relating to the interaction of DFS with other signal processing algorithms will be addressed, as will issues relating to verification and validation techniques of the adaptive and nonadaptive aspects of the algorithm.

Friday, August 25

SESSION FOUR

FITTING

Moderator: Arlene Neuman

11:10 AM Strategies for optimizing hearing aid fitting of hearing-impaired children

Teresa Ching and Mandy Hill, National Acoustic Laboratories, Australia
Colleen Psarros, Children's Cochlear Implant Centre, Australia

This paper 1) presents a procedure for evaluating hearing aid fittings by a systematic use of parents' and teachers' observation of a child's functional performance in everyday life; 2) examines the validity of this procedure by relating it to paired comparisons judgment of speech intelligibility; 3) evaluates whether young children have different amplification requirements from adults with similar hearing losses; and 4) evaluates whether children who wear cochlear implants with hearing aids have different requirements to those who use hearing aids in both ears. Hearing-impaired children depend on the auditory signal to develop speech and language. It is imperative that amplification be optimized to facilitate their development. Currently in Australia, children are fitted according to a NAL formula. This formula has been validated to prescribe frequency response and gain that agreed with the average preferences of hearing-impaired adults and children, and is therefore a useful starting estimate of the characteristics needed by any individual. What is best on average, however, may not always be best for the individual. To optimize hearing aid fitting for each child, evaluation or fine-tuning is necessary, and the need becomes greater as hearing loss becomes more severe. Evaluation procedures are available for adults, but few are available for children.

For children older than six years, a paired comparison procedure based on audio-visual presentation of speech can be used to identify the most effective option among a few alternatives. For younger children, a systematic use of parents' and teachers' observations of children's aural/oral performance in every-day life situations can give information about the effectiveness of a fitting. Both methods are applied to the evaluation of hearing aid fittings of children who use hearing aids only, and children who use hearing aids with cochlear implants. Results will be discussed with reference to the amplification needs of these two groups. The clinical applicability and relative usefulness of the procedures will also be discussed. Partly funded by the Cooperative Research Centre for Cochlear Implant and Hearing aid Innovation, Australia

11:45 AM Fitting hearing aids by listening preference

Claus Elberling and Thomas B. Sivertsen
Oticon Research Centre, Eriksholm, Denmark

The conventional way of fitting a hearing aid doesn't always arrive at a setting that satisfies the hearing aid user. This is partly because most fitting rationales are formulated from group-average data and partly because the fine-tuning process demands the user to describe to another person how the amplified sounds are perceived. One way to remove these shortcomings might be to involve the user more directly into the fitting process. To investigate this possibility we have performed a series of laboratory and field experiments. 30 hearing-impaired test subjects individualized the setting of a portable, experimental hearing aid by adjusting to their preferred setting when listening to a series of auditory scenes. Subsequently a preliminary field test was carried out, where 17 test subjects used

their individualized hearing aid in their daily environment for a week. The functionality of the individualized hearing aids was described by acoustical measurements using the auditory scene signals. From these measurements the insertion gain and compression ratio at 50 dB SPL (third octave) were obtained at three frequencies and analyzed as a function of the pure tone threshold. The test-retest variance was also analyzed using the data from 13 test subjects who went through the preference adjustment procedure twice with a two weeks interval. The preliminary field test was evaluated by a subjective comparison to the test subjects' own multi-channel compression hearing aids. This was done using three questionnaires, with both absolute and relative ratings of different dimensions of perception. The results demonstrate the following: (1) the preferred settings appear to be reasonable i.e. in accordance with established non-linear fitting formula. (2) the adjustment accuracy is surprisingly high i.e. with a test-retest variance of about 3.5 dB. (3) the individualized hearing aids were judged to be superior to own aids for a number of perceptive dimensions related to difficult-to-listen situations. (4) the test subjects liked to be personally involved and to perform the setting of the hearing aid themselves.

Friday, August 25

SESSION FIVE

Speech Processing in Hearing and in Hearing aids

Moderator: Pamela Souza

5:15 PM Insights into hearing aid signal processing from the neural representation of speech

Eric D. Young, Teng Ji, Ian C. Bruce, and Murray B. Sachs,
Center for Hearing Sciences and Department of Biomedical Engineering
Johns Hopkins University

Hearing aids have generally been designed and evaluated using human perceptual performance. Because the primary lesion in sensorineural hearing loss (SNHL) is in the cochlea, we are studying the neural representation of sounds in the auditory nerve as a more direct index of the pathologies and the effects of signal processing algorithms.

Previously, the representation of the vowel EH has been studied in the auditory nerve of normal cats and cats with SNHL due to high-level sound exposure (Miller et al., JASA 101:3602, 1997). Hearing thresholds in this preparation increase from about 10 dB at 1 kHz to about 60 dB at 2-4 kHz. Beyond the shift in sensitivity, the clearest deficits in the neural representation are the degraded cochlear filtering. Normally, cochlear frequency analysis produces a tonotopic representation in which components of the stimulus are separated by frequency, so that the activity of each nerve fiber is driven by stimulus components at frequencies near the fiber's best frequency. The responses are dominated by formant frequencies, so that the first two formants each capture the responses of fibers tuned to frequencies near the formant frequencies. Other spectral features of the vowel may also be represented, but do not capture of fibers' responses. In ears with SNHL, fiber tuning is broadened and responses to the first two formants are not confined tonotopically. Instead, responses spread widely, to essentially all fibers tuned to frequencies at and above the frequencies of the formants. This means that there are no fibers responding exclusively to the second or third formants. As a result, the representation of these stimulus features, as measured by their discriminability based on the neural activity, is reduced. The deficits in ears with SNHL can be partly alleviated by an algorithm that amplifies the second and third formants without amplifying the first formant or the energy between formants (Miller et al., JASA 106:2693, 1999).The results above were obtained with a steady-state vowel. It is necessary to repeat this work using realistic connected speech in order 1) to determine whether the deficits are the same or more complex with dynamic speech and 2) to evaluate whether the algorithm described above can be applied when the formants are changing with time. By tracking the dominant frequencies of the neural responses, we show that the deficits in the representation of connected speech are similar to those for steady vowels, although there is less capture of responses by the formants. Preliminary results with the amplification algorithm described above will be described. [Supported by NIDCD grant DC 00109]

5:50 PM Effects of spectral enhancement on the speech recognition performance of individuals with cochlear hearing loss

Ayaskanta Rout and Brandon Laflen, Purdue University
Todd Ricketts, Vanderbilt University

Previous research has demonstrated that as a result of cochlear hearing loss, auditory filters are broadened and the internal representation of speech spectrum is smoothed. This may result in poor speech recognition by hearing impaired listeners in the presence of background noise. At a theoretical level, it appears promising to preprocess speech and enhance the spectral contrasts (i.e., the peak to valley distance in the short term spectra of speech) to counteract the spectral smoothing that takes place in the damaged cochlea.

A novel approach was taken to design a spectral enhancement algorithm. A filtered second derivative based peak detection mechanism was designed to precisely detect peaks and valleys in the short term spectrum of speech, and then enhance the peaks while attenuating the valleys. Acoustic and perceptual evaluation of the spectrally enhanced speech was performed. Sentence materials at +5 dB signal to babble ratio were processed using the algorithm at three levels of enhancement (mild, moderate, and strong). Results obtained from eight listeners with sensorineural hearing impairment indicated that a mild amount of spectral enhancement improved speech recognition scores for half of the subjects. The subjects who benefited most from spectral enhancement had more severe hearing losses and poorer performance on the baseline condition (no enhancement) compared to those subjects who did not show any change in speech recognition scores with enhancement. A detailed description of the various steps of the spectral enhancement algorithm and its evaluation will be presented.

6:25 PM The effect of stimulus bandwidth on fricative perception in normal-hearing and hearing impaired children and adults.

Patricia Stelmachowicz, Andrea Pittman, Brenda Hoover and Dawna Lewis
Boys Town National Research Hospital

The perceptual strategies of young normal-hearing (NH) children and hearing-impaired (HI) adults have been found to differ from those of NH adults (Nittrouer & Crowther, 1998; Nittrouer, Crowther, & Miller, 1998; Doherty & Lutfi, 1996, 1999). These differences are likely due to a lack of experience with speech perception for the children and the presence of hearing loss for the adults. Hearing-impaired children present a unique problem in that hearing loss and immature perceptual skills may interact to produce perceptual weighting strategies that differ from all other groups.

This study examined the perceptual weighting strategies and performance-audibility functions of 11 moderate- to moderately-severe HI children, 11 age-matched NH children, 11 moderate to moderately-severe HI adults, and 11 NH adults. The purpose was to (a) determine the perceptual weighting strategies of HI children relative to the other groups, and (b) to determine the audibility required by each group to achieve a criterion level of performance. Stimuli were four nonsense syllables (/us/, /uS/, /uf/, and /uT/). The vowel, transition, and fricative segments of each nonsense syllable were identified along the temporal domain and each segment was randomly amplified within each syllable during presentation. Point-biserial correlation coefficients were calculated using the amplitude variation of each segment and the correct and incorrect responses for the corresponding syllable.

Results showed that for /us/ and /uS/, all four groups heavily weighted the fricative segments during perception whereas the vowel and transition segments received little or no weight. Performance-audibility functions were constructed for these segments and the mean audibility levels required to achieve > 70 % recognition were calculated for each group. Results showed that the HI children and adults required similarly low levels of audibility relative to the NH adults and children who required significantly higher levels. A decision theory approach, used to confirm the audibility criteria for each group and phoneme, yielded similar results. The present study revealed: (a) similar perceptual weighting functions for the HI children relative to the other groups; and (b) lower audibility requirements for the HI groups relative to the NH groups when perceiving the fricative segments of /us/ and /uS/.

Saturday, August 26

SESSION SIX

ALTERNATIVE THERAPIES FOR SENSORINEURAL HEARING LOSS

Moderator: Lynn Luethke

8:00 AM Holes in hearing: Implications for cochlear implants and hearing aids

Robert Shannon and John Galvin, House Ear Institute.
Deniz Baskent, University of Southern California, Dept. of Biomedical Engineering

Research on cochlear implants has revealed several aspects of signal processing for speech that also have implications for hearing aid processing. Auditory pathology can produce a localized loss of hair cells and neurons, which creates a "hole" in the tonotopic distribution of spectral information. In these patients the pattern of neural activity presented to the brain is distorted around the hole relative to a normal ear. In both cochlear implants and hearing aids the signal level from that spectral region is usually increased until the sound is audible. But this audibility may simply result from the excitation spreading to intact tonotopic regions on the edges of the hole, resulting in a local distortion in the tonotopic pattern of activity. Small tonotopic shifts in frequency information may be beneficial in terms of preserving the information that would otherwise be lost in the hole. However, larger distortions in the tonotopic pattern can be devastating for speech recognition. [Funded by NIDCD]

8:35 AM Direct mechanical stimulation of the ossicles via middle ear implant devices

Sigfrid Soli, Rachel Cruz, Paul Lewandowski,
House Ear Institute, Los Angeles, California

Middle ear implant (MEI) devices are implanted in the middle ear and stimulate the auditory system via direct mechanical coupling with the ossicles. The placement, alignment, and loading of the device on the ossicular chain affect the efficiency of energy transfer from the MEI through the ossicles to the cochlea. The presence of the MEI device can also affect the efficiency of conventional sound energy transfer from the tympanic membrane through the ossicles to the cochlea. The interactions of the MEI with the ossicular chain are of both scientific interest and practical significance. This presentation will describe methods and instrumentation for characterizing the efficiency of MEI and sound stimulation and the interactions between these two modes of stimulation when they occur simultaneously. The presentation will also address issues related to patient selection and outcome assessment for MEI devices that are related to this unique means of auditory stimulation.

9:10 AM Technical Considerations for a disposable hearing aid

David Preves, Songbird Medical Inc, New Jersey

A disposable hearing aid has been developed for mild-to-moderate hearing losses. Due to employing a one size shell, the need for ear impressions is eliminated. The methodology used in the design of this uni-ear, one-size, deep fitting instrument with a 3-D CAD program will be described. Because it is not subject to the rigors of long-term wear, the instrument can make use of materials not normally employed in hearing aids. For example, a very soft tip is designed to make a deep, yet comfortable fit and acoustic seal in the bony portion of the ear canal. The acoustic advantages of providing a deeply fitting earmold have been well documented, particularly for CIC hearing aid fittings. Degree of fitting success data will be presented showing the percentage of male and female ears on which the device can be fit to ITC or CIC depth without acoustic feedback problems.

The electroacoustic performance of the instrument is not varied by the hearing professional with trimmers or programmable parameters, but, rather by selection from a group of nine fixed prescription formats. A rationale for selection of one of the nine prescription formats for a patient will be presented. The circuitry has WDRC with TILL compression and there is no volume control. In quiet listening environments, persons with mild hearing loss wearing hearing aids often complain about hearing excessive circuit noise. One of the primary causes is amplified microphone circuit noise. A new low noise microphone in which the entire hearing aid circuit is packaged will be described. Examples of 1/3-octave noise measures compared to audiograms predict whether the hearing aid noise will be audible to hearing aid wearers.

APHAB and HINT results from field trials of the disposable instrument will be reported to document day-to-day benefit and improvement of speech recognition in noise. Aided loudness growth data will be presented that illustrates how well low, moderate and high intensity sounds were mapped into the residual dynamic range of patients.

Saturday, August 26

SESSION SEVEN

BINAURAL HEARING AND AUDITORY GROUPING

Moderator: Monica Hawley

11:10 AM Binaural noise reduction and a localization model based on the statistics of binaural signal parameters

Volker Hohmann, Johannes Nix, Thomas Wittkop, and Birger Kollmeier
AG Medizinische Physik, Universitaet Oldenburg, Germany

One major problem of hearing impaired people is the 'Cocktail-Party-Effect', i.e., the reduced ability to understand speech in noisy environments, which cannot be fully compensated for by current hearing aids. The hearing system uses binaural information, i.e., interaural level and phase/time differences in addition to monaural cues to localize sound sources in the environment. Based on the localization result and on several monaural parameters, e.g. common onsets and common modulation, the listener forms acoustical objects and performs a significant noise reduction.

Noise reduction algorithms have been proposed that try to imitate the binaural information processing in the hearing system by using two microphones at both ears and a central processor. This setup allows for the use of interaural parameters to estimate the direction of sound sources and to attenuate unwanted sound source directions. However, in noisy environments the binaural parameters become noisy as well, reducing the accuracy of the sound source estimation and therefore reducing the maximum performance of the noise reduction algorithm. In this presentation, a localization algorithm is introduced, which uses a statistical description of the binaural parameters in order to perform a robust estimation of sound source incident directions in noisy environments. Possible Noise reduction schemes based on this localization algorithm are discussed.

11:45 AM Auditory grouping and attention

Chris Darwin, University of Sussex, UK

This talk will first review recent experiments on how listeners with normal hearing group and localise complex sounds and secondly discuss the implications of this research for the hearing-impaired. In particular, it explores the role of binaural cues in auditory grouping and the relationship between auditory grouping and mechanisms of auditory localisation. This relationship is a complex one since, paradoxically, the primary cue for human azimuthal localisation of speech (ITD) is surprisingly ineffective as a grouping cue (Culling, J. F. and Summerfield, Q. ,1995, J. Acoust. Soc. Am. 98, 785-797). However, listeners can use small differences in ITD to attend to one of two simultaneous speech messages. I will present new data on what structural properties of speech sounds enable them to be grouped and then separately localised by ITD. I will also review work on the relative importance of other cues (such as prosody and vocal-tract size) that listeners can potentially use to attend selectively to one of two spoken messages.

Saturday, August 26

SESSION EIGHT

OUTCOME MEASURES

Moderator: Arthur Boothroyd

5:15 PM Measured Satisfaction: What does it tell us?

Robyn Cox and Genevieve Alexander
University of Memphis

The field of hearing health care is marching steadily in the direction of evidence-based practice. Thus, there are increasing pressures for clinicians to produce data to demonstrate the effectiveness of the treatments they prescribe and provide. In the modern era the most convincing evidence of effectiveness is data that show the extent to which treatment has minimized the consequences of hearing impairment in the daily life of the hearing-impaired person.

Subjective evaluations of hearing aid fitting outcomes are arguably the most valid and time-efficient approach to gathering this type of data. Development and employment of self-report tools has been brisk in the past decade. Several variables contend for the position of pre-eminent outcome metric, including benefit, use, residual impact of the impairment, and satisfaction. There is a need for substantial additional research to explore the associations among outcome variables and their differential value in quantifying effectiveness. The paper will address this topic. At the last Arrowhead meeting, we described the development of, and early data from, the Satisfaction with Amplification in Daily Life (SADL) Scale. In this paper, we will present the results of a cross validation and construct validation study of the SADL completed with clients from thirteen Audiology private practices.

In addition, we will show data to illuminate the relationships between satisfaction and several other variables that are often implicated in the outcomes of rehabilitation involving hearing aids. These include pre-fitting expectations, hea ring aid cost, and technology levels. Finally the interim results of a current investigation exploring the interrelationships among aspects of personality, pre-fitting expectations, and post-fitting disability, handicap and satisfaction will be discussed. [Supported by the Department of Veterans Affairs RR&D Service]

5:50 PM Impact of digital labeling on outcome measures

Ruth Bentler
The University of Iowa

In 1996 the beginning of the digital hearing aid era began in the United States. This technological advance was heralded as a breakthrough in hearing aid technology. Earlier shortcomings of size and power consumption were resolved and the end product is now a true digital processor that can be used in hearing aids of any size and/or style. Marketers have touted this development as the qualitative equivalent of changing from LP records to CD sound recordings. Anecdotal claims of increased performance abound, but few laboratory studies have shown significantly increased benefit based on the digital processing alone, except with self-report measures. The question remains whether the subjects/listeners are responding to the prejudicing effects of "high technology" labels, or are really experiencing improved listening in situations that are not easily reproducible in the laboratory setting.

A total of sixty subjects participated in this study with equal numbers assigned to one of three groups. Each group was matched in demographic profile (age, hearing loss, previous experience, etc). Group A was fit with each of two dsp hearing aids for a month (cross-over design). Group B was fit with the same two hearing aids, but were told one was "conventional"; Group C wore the same hearing aid for two months, but were told that they were wearing a "conventional" hearing aid for one of the months. The two clinicians fitting the hearing aids used a script to describe the merits of digital technology over the "conventional" technology.

The results of this study indicate significant prejudice effects across both self-report and psychophysical measures of speech perception in noise. A clinician effect was also noted for several measures. These results will be presented and the implications for research design discussed.

6:25 PM Aspects of auditory ecology and psychoacoustic function as determinants of benefits from and candidature for non-linear processing in hearing aids

Stuart Gatehouse, MRC Institute of Hearing Research, Scotland
Claus Elberling and Graham Naylor, Oticon Research Center, Denmark

Recent years have seen an expansion of interest in factors outwith simple aspects of audibility as potential determinants of experienced disability following a hearing impairment and the benefits that accrue following hearing aid provision. Variables of interest include the distortions which are an almost inevitable accompaniment of sensorineural hearing loss (aspects of frequency, temporal and binaural processing) and the ways in which client-centered attributes such as personality, motivation, and cognitive function are influential. Aspects of signal-processing in hearing aids attempt to overcome, or at least compensate for, some of these non-linear distortions which accompany sensorineural hearing loss. The conceptual basis of such processing and field trials of wearable devices implicate some aspects of psychoacoustic function (e.g. dynamic range, frequency and temporal resolution, etc.) in benefit and candidature. However, other components of signal-processing attempt to allow listeners to function in this paper reports results from a within-subject crossover design of linear fittings, fast-acting WDRC, slow-acting AVC and hybrid compression rationales. The results show that aspects of speech identification in noise for amplitude compression systems are related to the psychophysical characteristics of impaired auditory systems, but play a minimal role in preference and perceived benefits. In contrast, simple characterisation of auditory ecology by either questionnaires or logging of physical aspects of the auditory environment do exert significant predictive leverage. It is argued that a holistic approach to hearing aid fitting and selection which embraces aspects of a listener’s lifestyle and requirements, as well as more traditional measures of psychoacoustic function in sensorineural hearing loss, can materially enhance understanding, benefit and candidature.

SUNDAY, AUGUST 27

SESSION NINE

ISSUES AND INNOVATIONS

Moderator: Michael Nilsson

8:00 AM LISTENERS WHO PREFER MONAURAL TO BINAURAL HEARING AIDS

Anne Strouse, Richard Wilson and Colleen Noe
VA Medical Center, Tennessee

Despite the theoretical superiority of binaural hearing, some individuals with hearing loss prefer monaural to binaural amplification. Five such patients have been identified. Based on audiometric findings, each patient originally had been fit with binaural amplification. Each reported after several months of hearing aid use that the hearing aids were not satisfactory, especially in the presence of background noise and that they performed better using one hearing aid rather than two. Four of the patients wore only the right hearing aid, whereas one patient could wear either the right or left hearing aid successfully but not both.

Previous reports of unsuccessful use of binaural amplification in elderly listeners consistently show an associated left-ear deficit in dichotic listening. In the evaluation of dichotic listening in elderly listeners, both free- and directed-recall conditions can be used to assist in differentiating performance deficits owing to cognitive stress (e.g., memory and speed of mental processing) from performance deficits unique to an auditory/structural disorder. In the free-recall task, stimuli are presented dichotically, and the listener reports both items. In the directed-recall task, the listener only reports the items in the directed ear that is cued prior to stimulus presentation. When recognition performance is poor in the free-recall condition, but improves substantially in the directed-recall condition, the problem is primarily in the cognitive domain. Memory and attention abilities are insufficient for successful performance when both ears must be monitored simultaneously in the free-recall condition.

In this report we will present (1) basic audiometric and dichotic listening data; (2) cognitive screening, hearing- aid outcome, and quality-of-life measures; and (3) monaural and binaural hearing-aid performance. We are in the process of evaluating different amplification strategies to use with these patients including speech in noise algorithms, multiple-microphone arrays, and FM technology. The results have implications regarding the evaluation of patients and the selection of appropriate amplification.

8:35 AM ACTIVE NOISE REDUCTION IN DIGITAL HEARING AIDS

Wouter A. Dreschler
Academic Medical Center, Amsterdam

In commercially available digital hearing aids new active noise reduction techniques have been introduced that take advantage of the specific differences between relatively steady noise signals and modulated signals like speech.

In this presentation data will be shown from three different clinical trials in which similar techniques have been used in 3, 4, and 14 frequency bands, respectively. With respect to the results for speech perception in noise, it is hard to obtain hard evidence of better performance measures due to the active noise reduction. However, the results of the subjective outcome measures are more favourable.

A problem in the interpretation is that details about the technical implementation of the noise reduction schemes are not available publicly. On the contrary, some manufacturers tend to discourage audiologists to measure the electro-acoustical properties themselves and they point to the danger of the dynamic behaviour of the signal processing schemes. However, dynamic responses can well be taken into account, given that the test conditions are specified in enough detail. In this respect the set of noises, proposed by ICRA, can be of help. ICRA noises provide a well-specified set of speech noises with different spectral shapes according to gender and vocal effort with different amounts of speech modulations according to the number of speakers. These well-specified noises have been used in measurements for the hearing aids involved in the clinical trials.

The sensitivity of different active noise reduction schemes is compared by measurements with ICRA noises with a varying ratio between unmodulated and modulated test signals: a modulated-unmodulated ratio. The results show characteristic and reproducible differences between the noise reduction schemes from different brands that are currently available on the market. It will be argued that this information may be important to understand the differences between the results in different studies.

9:10 AM AUTOMATIC CLASSIFICATION OF DIFFERENT LISTENING ENVIRONMENTS IN A GENERALIZED ADAPTIVE HEARING AID

Peter Nordqvist
Department of Speech, Music and Hearing, KTH, Stockholm, Sweden

This presentation describes an algorithm, which can be used in a hearing instrument for automatic classification of different listening environments. A hearing aid can utilize environment classification in several ways. One idea is to connect each listening environment with a pre-computed gain frequency response and then switch between the filters when the user is moving between different listening environments. Another possibility is to let the classifier supplement an existing AGC system with multiple lookup-tables, one for each listening environment. A third possibility is the let the classifier control the on/off switching of the directional microphone, the noise reduction, and the feedback suppression.

The focus in this article is on the classification. Is it possible to classify between different listening situations? Is it possible to implement the classifier in a hearing aid?
The classification is based on Hidden Markov Models (HMM), which is a standard method in speech recognition and speaker verification. The algorithm extracts a set of features every 6ms from the raw input sound. The vector of features is the input to a classifier, which consists of three layers. The first short-term layer is a vector quantizer (VQ), the second middle-term layer is based on HMMs, and the third long-term layer controls the switching between the HMMs.

The structure of the algorithm allows us to have different switching times between different listening environments, e.g. slow switching between traffic and babble and fast switching between traffic and speech.

Five listening environments have been used in a preliminary evaluation: real-life recordings of traffic noise, babble noise, speech, subway noise, and outdoor noise.
When the listening environments are presented alone, not mixed with another listening environment, the average short-term classification error, for all five listening environments, is 3.7 percent. The long-term layer has a low-pass filter behavior causing the long-term classification error to be close to zero before or after the transition between two listening environments.

The result indicates that it is possible to classify between different listening environments. Estimation of memory and power consumption indicates that it is possible to implement the algorithm in a digital hearing aid instrument.
A more detailed evaluation and a real-time demonstration of the algorithm will be presented.

SUNDAY, AUGUST 27

SESSION TEN

ENGINEERING ISSUES AND INNOVATIONS

Moderator: Steven Thompson

10:05 AM A BIOLOGICALLY INSPIRED DIRECTIONAL MICROPHONE CONCEPT FOR HEARING AIDS

R. N. Miles and S. Sundermurthy
Department of Mechanical Engineering, State University of New York

The incorporation of directional microphones in hearing aids poses a number of engineering challenges. Any acoustic sensor that responds preferentially to sound from a certain direction must be able to detect the gradient of the sound pressure which, unfortunately, becomes smaller as the overall size of the sensor is reduced. For sensors that are small relative to the wavelength of sound, as is the case with hearing aid microphones, the pressure gradient also possesses a 6dB/octave high-pass filter characteristic. The degradation in sensitivity and undesirable frequency response can be compensated for electronically but this compensation can result in unacceptable noise performance.

An alternative approach to electronically compensating the pressure gradient detected by a small directional microphone is suggested by our study of the mechanics of directional hearing in small insects. The animals we have examined do not have the means of performing electronic amplification or filtering also have rather modest neural signal processing ability, and yet, are highly adept at localizing sound sources. We have shown previously that the vibrations of the tympana in these insect ears are compensated by carefully designed mechanical structures, resulting in a nearly flat frequency response. In the present study we describe the design of a mechanically compensated, first order pressure gradient microphone that has a flat frequency response and excellent noise performance.

10:40 AM SIMULATION AND REAL-TIME IMPLEMENTATION OF A NEW OPTIMUM BEAMFORMING TECHNIQUE

Michael E. Lockwood, Mark E. Elledge, Kong L. Yang, Douglas L. Jones, Charissa R. Lansing, Robert C. Bilger, Bruce C. Wheeler, William D. O'Brian and Albert S. Feng
The Beckman Institute, University of Illinois at Urbana-Champaign

Microphone arrays have been shown to improve the intelligibility of a speech signal in noise for hearing-impaired listeners, and many signal-processing methods have been proposed for use with these arrays in hearing aids. These include methods using fixed weights, such as conventional delay-and-sum beamforming and superdirective weights, and those which employ adaptive weighting, including the well-known Frost algorithm. Fixed weight methods have the disadvantage of being unable to adapt to different situations, while adaptive methods cannot adapt quickly enough to effectively deal with changing auditory scenes, such as moving sources or rapid shifts in the frequency of a source. Using minimum-variance beamforming techniques, a new frequency-domain algorithm is presented which allows optimal filtering methods to be applied individually to different narrow frequency bands, thereby allowing simultaneous cancellation of multiple interfering sources whose spectra do not overlap simultaneously in time and frequency. This method calculates optimum filter weights for small blocks of data, allowing it to adapt extremely quickly to changes in target and interfering signals, and therefore it does not suffer from the problems associated with conventional fixed or adaptive methods as mentioned above.

With this technique it is possible to process signals from an array of two or more microphones in a manner which dramatically reduces the amplitude of signals originating away from a desired receive direction, even if the sources are relatively close to the desired receive direction. In addition to being computationally efficient, the algorithm, when used in a two-microphone system, was able to produce output which had a consistently positive intelligibility weighted SNR gain [Link & Buckley, J. Acoust. Soc. Am., 91, 1662] and in which the desired talker was noticeably easier to understand.

A two-channel version of the algorithm was tested on artificially combined anechoic signals with varying numbers of interfering sound sources. Performance comparisons with conventional approaches for binaural hearing aids are presented. Furthermore, the performance of a real-time system implementing the algorithm is discussed, and shows good robustness to room reverberation effects. Collectively, the results show that this algorithm may be the first building block for a binaural hearing-aid processing algorithm which suppresses off-axis interference in a highly directional manner.

11:15 AM Finite-State modeling for specification of non-linear hearing instruments

Arne Leijon and Peter Nordqvist
Royal Institute of Technology, Stockholm

Non-linear hearing instruments, by definition, adapt their signal-processing characteristics depending on the input signal. Standardized measurement procedures were designed primarily for linear hearing instruments and typically use test signals that may cause an advanced hearing aid to behave differently than it does in real life. By allowing real-life sound as test signals we can obtain a specification for any given sound environment. At least the following separate performance aspects need to be described:

1. The typical frequency response in a given sound environment
2. The degree and time-scale of the instrument's adaptation to different environments
3. The instrument's adaptation to different sound sources in a given environment, e.g. to different speakers and background noise sources
4. The instrument's (fast) adaptation to the modulation patterns of a particular sound source, e.g. to the stream of speech sounds from a given speaker.

Earlier specification methods have usually made some implicit assumptions about the internal design of the instrument. For example, regression analysis of input and output levels in each frequency band can describe the function of independent compressors in each band, but it can not show how the gain at one frequency is influenced by signals in other bands.

We present a pilot study using a more general approach: The hearing instrument is modeled as a finite-state machine that transforms the input log-spectrum data stream depending on a time-varying internal state of the instrument, which in turn changes depending on the input signal. The quantised internal states represent the total set of control parameters in an instrument with e.g. multi-channel compression and/or noise suppression.

Recorded input and output signals from the hearing aid are first analysed into streams of short-time spectra (in dB). A Hidden Markov Model (HMM) is then adapted, using standard training procedures, to describe the combined stream of input and output short-time spectra. If speech is used as test signal, each HMM state represents a class of speech sound, together with the typical hearing-aid processing of this sound class. Thus, the trained HMM can be seen as a compact representation of the recorded input and output test signals.

Results of this approach will be presented for simulated hearing instruments with exactly known adaptive properties as well as for a few commercial hearing instruments.

Poster Abstracts

Contributed posters have been divided into three groups of equal size. Posters will be displayed on the walls of the Conference Center Thursday through Saturday. The first group of posters will be displayed Thursday 8:00am to 10:00pm. The second group of posters will be displayed Thursday 8:00am to 10:00pm and the third group will be displayed Saturday 8:00am to 10:00pm.

POSTER GROUP A

Thursday 8:00AM – 10:00PM

PA1

How many channels are needed in wide dynamic range compression (WDRC) hearing aids?

Gitte Keidser and Frances Grant
National Acoustic Laboratories, Australia

In the past decade WDRC hearing aids have been released with an increasing number of channels, in which gain and compression can be independently adjusted. Introducing more channels means that the variation with frequency of a person's audible range can be better matched by the hearing aid. Whereas multi-channel WDRC hearing aids seem to have an advantage in theory, this remains to be proven in practice.

The National Acoustic Laboratories' new procedure for fitting non-linear hearing aids (NAL-NL1) prescribes gain and compression parameters for as many channels as desired. In a recent study conducted to evaluate the NAL-NL1 fitting formula, one of the aims was to determine whether NAL-NL1 was most effective when implemented in one, two, or four channels. Twenty-four subjects with flat and steeply sloping losses evaluated all three compression schemes in the laboratory and compared the single channel and two-channel schemes in a field test. In the laboratory the subjects completed a paired comparison test and a speech recognition test in a variety of situations and input levels. During the field test the subjects compared the single channel and two-channel schemes in individually selected everyday listening situations for 4-6 weeks.

In the laboratory most subjects showed no significant preference for either of the compression schemes. Of the few subjects who consistently selected one scheme over the others, the majority preferred the single-channel scheme. The speech recognition test in noise revealed no significant difference between the three schemes ($p = 0.93$). The field test, on the other hand, showed that all subjects with a steeply sloping loss, but one, selected the two-channel scheme. Among subjects with flat losses 62% selected the single channel scheme, and the remaining 38% selected the two-channel scheme. No factors have been found to discriminate between subjects who selected single channel or two-channel schemes.

PA2

Predictions of intelligibility and user preferences for amplitude-compressed speech based on the STI

Karen Payton, University of Massachusetts
Peninah Fine Rosengard, Harvard – MIT
Lois Braida, MIT

This presentation reports on relations between intelligibility scores, user preferences, and estimates of Speech Transmission Indices for speech processed by amplitude compression systems. Estimates of the effects of compression on the STI derived from changes in the modulation depth of sinusoidally modulated noise, e.g. Steeneken and Houtgast, [1980, J. Acoust. Soc. Am. 67, 318-326], may not be appropriate for amplitude-compressed speech.

Our approach is based, instead, on measured changes in the modulation spectrum of the speech signal itself [Payton \& Braida (1999) J. Acoust. Soc. Am. 106, 3637-3648]. This technique for estimating the STI has been shown to be capable of predicting intelligibility scores for noisy and/or reverberant speech. The modified STI can also take into account elevated thresholds as well as linear signal processing.

Data that illustrates an application of the same estimation procedures to amplitude-compressed speech will be presented. Three syllabic compression conditions were evaluated and tested in different background noise conditions. Four, independently compressed, frequency bands were used to compress the speech. Attack times and release times were fixed at 20 ms and 200 ms respectively. Target compression ratios were varied over the range reported in the literature as being beneficial to listeners for either improved understanding and/or improved listening comfort for speech in the presence of background noise. In all cases, each frequency band was compressed using the same target ratio. Two different types of background noise were considered: random speech-shaped noise and multitalker babble.

The speech materials were nonsense sentences, grammatically correct but semantically meaningless. Speech materials plus background noise were compressed and the NAL-R formula was used to determine output frequency-gain characteristics. Flat sensorineural hearing losses were simulated in normal-hearing listeners by the addition of continuous masking noise to the processed speech materials. Subjects were tested for both speech intelligibility and preferences across processing conditions. All conditions were also evaluated using the modified STI. A comparison of the intelligibility and preference scores to the STI will be presented. [This work was supported by NIDCD].

PA3

Effect of compression ratio on speech recognition in temporally complex background noise

Kumiko Boike and Pamela Souza
University of Washington

Everyday listening situations typically require listeners to distinguish speech in backgrounds of temporally fluctuating noise. Listeners with hearing loss have more difficulty in this situation than listeners with normal hearing, in part because they are unable to take advantage of the temporal "dips" in the background signal. Little is known about the interaction between the temporal properties of background noise and use of compression amplification. Because compression alters the properties of the signal, it can alter the relationship between the temporal fluctuations of the signal and noise. Previous work (Boike & Souza [2000], JSHR v 43, 456-468) suggested that increasing compression ratio has a detrimental effect on speech recognition in noise for hearing-impaired listeners but not for normal-hearing listeners. The current study further explores these findings.

This study examined the ability of normal-hearing and hearing-impaired listeners to recognize speech in temporally complex background noise as a function of compression ratio. Nine normal-hearing adults and nine adults with mild to moderately severe sensorineural loss participated. Speech recognition scores were measured for sentences presented in quiet and in three different background noise conditions (single-talker modulated noise, babble-modulated noise, unmodulated noise) that were spectrally similar but varied in the availability of temporal dips. The speech and noise were mixed at -4 and +2 signal-to-noise ratios for the listeners with normal hearing and hearing impairment, respectively. All signals were digitally processed by linear amplification and by compression amplification using compression ratios of 2:1 and 5:1. The attack time was 4 ms and the release time was 45 ms. The same frequency-gain response and presentation levels were used for each condition.

Normal-hearing listeners were able to take maximal advantage of temporal dips, demonstrating higher recognition scores for speech in a single-talker modulated background than for either the babble-modulated or unmodulated noise. The listeners with hearing loss did not show the same pattern, with poorer performance when noise was temporally complex (babble-modulated) than for speech in unmodulated noise. For both groups, performance decreased with increasing amounts of compression. Application of increasing amounts of compression did not alter the effects of background complexity on each group.

PA4

Effect of preferred volume setting on speech audibility for linear peak clipping, compression limiting and wide-dynamic range compression amplification

Pamela Souza and Virgina Kitch
University of Washington

Controlled studies using laboratory-based amplification systems have demonstrated that wide-dynamic range compression amplification can improve speech audibility, with corresponding improvements in speech recognition, over linear amplification. It is less clear whether this benefit is maintained in listening situations in which the listener controls the hearing aid volume setting. Variations in volume setting can significantly offset the theoretical advantage of improved audibility offered by compression amplification. The purpose of this study was to quantify speech audibility at the patient's preferred volume setting for peak clipping, compression limiting and wide dynamic range compression hearing aids. All participants had mild-to-moderate sensorineural hearing loss and were fit monaurally with a programmable behind-the-ear hearing aid. The hearing aid was programmed sequentially (in random order) with three amplification strategies: linear, peak clipping; compression limiting; and wide-dynamic range compression. In each case, target gain and output were selected using the DSL [i/o] (Cornelisse et al., 1995) prescriptive method and match to target was verified using probe microphone measures. For each amplification strategy, listeners were asked to adjust the volume of the hearing aid to maximize speech clarity. Test materials were speech passages presented at low, moderate and high input levels in quiet and in a background of multitalker babble. Overall and 1/3 octave band speech levels (gain and output) were measured at each preferred volume setting using a probe microphone system.

Results to date indicate that preferred volume settings were similar across the three amplification conditions. Listeners selected higher gain for speech in noise than for speech in quiet. Preferred gain varied systematically across input level, with significantly greater gain preferred for lower input levels. At high input levels listeners selected less gain for the wide-dynamic range compression hearing aid than for either of the other processing schemes. These differences in preferred gain may be due to the improved speech audibility offered by wide-dynamic range compression amplification.

PA5

Physiological Modeling for Hearing Aid Design

Ian Bruce, Eric Young and Murray Sachs
Johns Hopkins University

Perceptual models of impaired hearing are being used increasingly in hearing aid design. The most successful application has been compression schemes to compensate for loudness recruitment. Less fruitful have been attempts to counteract degraded cochlear filtering with spectral shaping. Perceptual models are developed from psychophysical measures that reflect both peripheral and central processing. More physiological detail may be required to describe the effects of a cochlear lesion on peripheral coding of speech.

For example, physiological data from hearing-impaired cats indicate that conventional hearing aid signal processing schemes do not restore normal auditory nerve responses to a vowel (Miller et al., JASA 101:3602, 1997) and can even produce anomalous and potentially confounding patterns of activity (Schilling et al., Hear. Res. 117:57, 1998). These deficits in the neural representation may at least partially account for poor speech perception in some hearing aid users. An amplification scheme has been developed that produces neural responses to a vowel more like those seen in normal cats and that minimizes confounding responses (Miller et al., JASA 106:2693, 1999).

A physiological model of the normal and impaired auditory periphery would provide simpler and quicker testing of such potential hearing aid designs. Details of a physiological model will be presented. Model predictions of vowel responses suggest that degraded cochlear filtering can indeed account for a good deal of the data from hearing-impaired cats described above. However, some response properties appear to result from physiological features that are not considered in perceptual models. In particular, auditory nerve responses to speech stimuli are very sensitive to wide-band nonlinearities in the basilar membrane mechanics known as two-tone suppression. These nonlinearities strongly affect how well auditory nerve fibers synchronize to specific formants at different stimulus intensities (Wong et al., Hear. Res. 123:61, 1998). Such factors will be important in generalizing the speech-processing algorithm described above to running speech. Model predictions of a prospective amplification scheme will be discussed. [Supported by NIDCD grants DC00109 and DC00023]

PA6

Perception of pure tones by listeners with and without a “dead region”

Martina Huss, Brian C.J. Moore, Thomas Baer and Brian Glasberg
University of Cambridge, UK

Hearing impairment may sometimes be associated with the complete loss of inner hair cells (IHCs) over a region of the basilar membrane. Such a region has been referred to as a "dead region" (Moore and Glasberg, 1997), and it can be defined in terms of the range of characteristic frequencies (CFs) normally associated with that region. Two different methods have been used to define the frequency range associated with a dead region, both based on masking (Thornton and Abbas, 1980; Moore et al., 2000). The assumption underlying both tests is that pure tones falling in a dead region are detected via the spread of excitation to intact IHCs with CFs outside the dead region.

It remains unclear how pure tones falling in a dead region are perceived in comparison to pure tones falling in a region where there are still functioning IHCs. The aim of the current study is to provide information on this topic. For all subjects, a test is administered to detect the presence of any dead region and define its limits (Moore et al., 2000). In addition, several aspects of tone perception are being examined.

In one experiment, subjects were asked to rate the quality of pure tones of various frequencies on a scale going from "Clear distinct tone" (1 on the scale) to "noise" (7 on the scale). For some subjects, there was a clear increase in rating as the frequency of a tone was increased from outside a dead region to inside it; tones within the dead region sounded more like noises. However, for other subjects, tones were rated as sounding tonal even when they fell somewhat inside a dead region; such cases were usually associated with very steeply sloping audiograms, where it was not possible to present audible tones with frequencies well inside the dead region. Also, some subjects without a dead region rated tones as more noise like when the tone frequency was moved into a region of greater hearing loss. Thus, the subjective impression of degree of noisiness does not seem to be a reliable indicator of the presence of a dead region.

In another experiment (in progress), subjects are being asked to make pitch matches between two pure tones presented alternately either to the same ear or to opposite ears. Preliminary data indicate that matches are more variable (indicating a less distinct pitch sensation) when the fixed tone falls in a dead region.

In a third experiment (in progress) listeners with highly asymmetrical hearing losses are asked to match the pitch of a pure tone falling in a dead region in one ear with that of a pure tone presented alternately to the other ear, not having a dead region at that frequency. Data should be available for presentation at the meeting.

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PA7

The effect of low-pass filtering on consonant identification for listeners with and without dead regions at high frequencies

Deborah A. Vickers, Brian C.J. Moore and Thomas Baer
University of Cambridge, UK

UKHigh-frequency cochlear hearing loss may be associated with "dead regions"; these are regions of the cochlea where there are no functioning inner hair cells and/or neurons (Moore and Glasberg, 1997; Moore, 1998). They are defined in terms of the characteristic frequencies normally associated with the damaged region. Dead regions can be diagnosed using psychophysical tuning curves (PTCs); when the tip of the tuning curve is shifted away from the signal frequency, this indicates a dead region (Thornton and Abbas, 1980; Moore and Alcántara, 2000; Moore et al., 2000). Recently, we have developed a simpler test, using a threshold-equalizing noise (TEN) designed to give equal masked thresholds at all frequencies for normally hearing subjects; abnormally high thresholds are indicative of dead regions (Moore et al., 2000).

There are several studies showing that listeners with high-frequency hearing loss often do not benefit from amplification of speech at high frequencies and some even show a degradation in performance with an increase in audibility of high-frequencies. One aim of the current study was to test the hypothesis that listeners without dead regions benefit from high-frequency amplification while listeners with dead regions do not. The second aim was to determine whether there was a benefit of amplifying frequencies falling just inside a dead region.

All subjects had high-frequency hearing losses. The presence or absence of dead regions was determined using PTCs and the test with TEN noise. VCV (vowel-consonant-vowel) stimuli were presented through headphones. Amplification according to the Cambridge formula (Moore and Glasberg, 1998) was applied for each subject using digital filtering to simulate listening through a hearing aid. The speech was either broadband (upper cutoff frequency 7.5 kHz) or was low-pass filtered with various cutoff frequencies. For listeners with dead regions, the cutoff frequencies were 0.8, 1.0, 1.2, and 1.4 times the low-frequency edge of the dead region. For listeners without dead regions, the cutoff frequencies were 0.8, 1.0, 1.2 and 1.4 kHz.

Preliminary results suggest that, for listeners without a dead region, performance progressively worsens as the cutoff frequency is reduced. Listeners with dead regions perform best for cutoff frequencies that are 1.2 or 1.4 times the low-frequency edge of the dead region and perform worse under the other conditions, including the broadband condition. These results have implications for fitting hearing aids. They indicate that listeners without a dead region should benefit from amplification over a wide frequency range. For listeners with a dead region, there is some benefit from amplifying frequencies just within the dead region, but amplification should not be applied for frequencies more than about 1.4 times the edge frequency of the dead region.

PA8

Auditory perception with sloping high-frequency hearing loss, and the limited benefit of frequency transposition

Hugh McDermott
University of Melbourne, Australia

For many people with sloping hearing loss, the amplification provided by conventional hearing aids can not restore full audibility at higher frequencies. Various frequency-lowering hearing aids have been developed in attempts to improve perception of speech and other sounds for these people. The AVR TranSonic is a body-worn, frequency-transposing aid that lowers all frequencies by a constant factor whenever the input signal is dominated by high frequency components. In a recent comparative study, 5 adults used the TranSonic in place of their conventional hearing aids for about 12 weeks. Significant improvements in speech perception were found with the TranSonic for 4 of the subjects. However, analysis of the results suggested that most of the improvements were related to the low-frequency amplification characteristics of the TranSonic, with only 2 subjects seeming to obtain benefit specifically from the transposition. A similar study is now being conducted with users of the ImpaCt aid, which employs a transposition function similar to that of the TranSonic, and is packaged in a behind-the-ear enclosure. Preliminary results indicate that performance of the ImpaCt and TranSonic are similar.

Some people with a very steeply sloping loss find conventional hearing aids unhelpful. Psychophysical studies on 5 adults with precipitous audiograms revealed abnormally steep loudness functions and anomalous tonal sensations at mid-to-high frequencies. Such characteristics may explain in part why most of those subjects rejected amplification. However, their scores on a speech perception test were close to the scores obtained by normally hearing subjects listening through a low-pass filter that simulated some of the effects of a steeply sloping hearing impairment. In an attempt to provide these subjects with additional information about higher frequency components of speech, recorded monosyllabic words were processed by an algorithm that continuously lowered all frequencies by a factor of 0.6. This algorithm is practical to implement in real time in a modern digital hearing aid. However, the frequency lowering did not improve open-set word recognition for the hearing-impaired subjects, even after a period of training. Although some improvements were obtained with the transposition by the normally hearing subjects (listening through a low-pass filter), similar improvements were also obtained when those subjects were trained and tested using nontransposed (but filtered) words. It is concluded that frequency transposition generally has limited potential to improve the perception of most people with sloping hearing loss. Alternatives, such as cochlear implantation, may provide greater benefit in many cases.

PA9

A Computational model for simulating basilar-membrane nonlinearity in subjects with normal and impaired hearing

Enrique A. Lopez-Proveda, Universidad de Castilla-La Mancha, Spain
Ray Meddis, University of Essex, UK

Psychophysical masking experiments show that hearing impairment implies not only an increase in hearing thresholds and in the width of the auditory filters, but also a loss of the nonlinearity associated with the cochlear compressive response. Loudness recruitment is a consequence of this loss that can not be overcome with linear hearing aids. The ideal hearing aid should, therefore, be able to compensate for the loss of nonlinearity as well as for the loss of sensitivity and sharpening of the auditory filters.

We are currently investigating on a computational model for simulating the results of masking experiments in human subjects with normal and impaired hearing. Our approach is based on adapting a digital dual-resonance nonlinear (DRNL) computational model of cochlear frequency selectivity for this purpose. The DRNL model was originally designed to simulate direct measurements of basilar membrane activity and has also proven to be able to simulate a wide range of auditory phenomena such as distortion products, two-tone suppression, as well as changes in bandwidth, best frequency and impulse response with signal level. It consists of two parallel pathways along which the stimulus is filtered: one linear and one nonlinear. The linear path consists of three cascaded elements: a gain control, a linear gammatone filter, and a low-pass filter. The nonlinear path consists of a compression function spliced between two identical linear gammatone filters. The model output is calculated by adding the signals coming o...

Preliminary results show that psychophysical masking data for subjects with normal and impaired hearing can be simulated by changing only the parameters of the compression function of the DRNL model, at least for a single frequency channel. Our final goal is to enhance the DRNL model so that, together with the forward masking model, provides us with a good quantitative description of hearing impairment for a range of frequency channels. The combined system will have a better ability to predict the benefit of any particular hearing aid and could be, therefore, a useful aid for designing and testing personalized nonlinear hearing aids.

PA10

Quantifying effects of the cochlear amplifier on temporal and average-rate information in the auditory nerve

Michael Heinz, MIT and Boston University
Steve Colburn and Laurel Carney, Boston University

The cochlear amplifier dynamically alters the tuning within the normal peripheral auditory system based on the spectral and temporal configuration of the stimulus, and is impaired or absent in many forms of sensorineural hearing loss. These findings suggest that the difference between normal-hearing and hearing-impaired listeners based on the cochlear amplifier is not simply a difference between systems with low and high thresholds, or narrow and broad tuning, or compressive and linear magnitude responses. Rather, the difference is between a static, broad, insensitive system and a fast, dynamic system that continuously changes (cycle by cycle at low frequencies) in response to the stimulus. This difference may be significant given that hearing-impaired listeners have particular difficulty understanding speech in temporally and spectrally varying backgrounds even with hearing aids.

The present study describes a general modeling approach that quantitatively relates the most significant auditory-nerve (AN) response properties associated with the cochlear amplifier to psychophysical performance. Quantitative methods combine phenomenological population models of the AN with statistical decision theory to evaluate performance limits imposed by the random nature of AN discharges. This approach, originally used by Siebert and Colburn in the late 1960's, has been extended in several ways. The use of computational AN models is important for evaluating hearing impairment because complex tasks are unable to be analyzed with the analytical models used by Siebert and Colburn. In addition, a new theoretical approach was developed to predict performance limits for psychophysical tasks that use random-noise maskers. These extensions were motivated by the fact that the cochlear amplifier has only a small effect on simple psychophysical tasks, consistent with model results predicting near-normal level discrimination of tones in quiet by hearing-impaired listeners.

Predictions were based on a computational AN model that included the effects of level-dependent tuning, level-dependent phase, compression, suppression, and fast nonlinear dynamics on the responses of high, medium, and low-spontaneous-rate AN fibers. The ability of temporal and average-rate information in the AN to account for human performance was evaluated for several psychophysical tasks for which the cochlear amplifier is believed to be significant (e.g., estimating auditory filters). Methods for quantitatively relating physiological response properties to psychophysical performance in masking studies will improve understanding of the significance of the cochlear amplifier in the normal ear. This knowledge will be beneficial for identifying useful forms of signal information that could be enhanced by preprocessing schemes to improve hearing-impaired performance.

PA11

Toward a clinical procedure for narrowband gap detection

Mary Florentine, Institute for Hearing, Speech and Language and Department of Speech-Language Pathology and Audiology, Northeastern University, Boston
Soren Buus, Communications and Digital Signal Processing Center, Department of Electrical Engineering, Northeastern University, Boston

This presentation describes the development of a clinical gap-detection test, which is simple, frequency specific, and reliable. A cued Yes-No Method of Maximum Likelihood (MML) procedure was compared to an Up-Down three-interval, two alternative forced-choice procedure. Results from five trained and seven naive normal listeners indicate that the Yes-No MML procedure yields valid and efficient estimates of narrowband gap-detection thresholds in 1-1.5 minutes per threshold. Results from 18 ears of 11 listeners with hearing losses of primarily cochlear origin indicate that the Minimum Detectable Gaps (MDGs) often were normal at low frequencies and somewhat elevated at and above 1 kHz. At all frequencies, the MDGs were normal in some listeners and elevated in others with no apparent relation to the pure-tone thresholds. This indicates that hearing losses of cochlear origin can reduce temporal resolution by different amounts in listeners with similar audiometric thresholds. This conclusion agrees with previous data showing that MDGs for broadband noise can differ between impaired listeners with similar audiograms (see Florentine and Buus, J. Speech & Hear. Res., 27, 449-455, 1984). [Work supported by NIH-NIDCD DC00187]

PA12

"Combination-sensitivity" as an aid to hearing pitch

Jagmeet Kanwal, Faye Chiao,
Georgetown University Medical Center

Recognition of the pitch of a complex sound is important for the perception of speech as well as musical sounds. For speech sounds, pitch may indicate the identity and/or the mood of the speaker whereas in music it is the fundamental variable in harmony and/or a melody. The auditory system in humans is specially designed to detect the pitch of any complex sound, yet the mechanism of this process is not well understood despite extensive psychophysical research on this topic. Neurobiologically inspired models of pitch perception are only beginning to be developed. Langner (1983, 1997) has proposed a neuronal correlation model that uses the periodicity of the sound envelop for determining pitch. Cartwright et al. (1999) have developed a mathematical model based on a generic nonlinear forced oscillator to compute the first pitch-shift effects. Both models conclude that extraction of pitch from a complex waveform can occur at the midbrain level so that higher levels of the brain can be used for computation We propose that pitch perception can be readily modeled using biologically-based neural networks that exploit the phenomenon of combination-sensitivity. Combination-sensitivity involves a nonlinear facilitation of neural responses to combinations of spectral components. Since natural sounds typically consist of several harmonics, combinations of these harmonic components can be used for computing the pitch of a sound. Combination-sensitive neurons are involved in processing complex sounds in virtually every major vertebrate group ranging from amphibians to mammals, including primates, and are known to be created at the midbrain level in mammalian species such as bats (Mittman and Wenstrup 1995). Yet, combination-sensitivity has not been extensively used to model the auditory system's capability for extracting pitch, timbre and other parameters of complex sounds. Algorithms based on the concept of combination-sensitivity may provide a robust method for computing pitch. Furthermore, by embedding the computational software for pitch perception along with filter banks in small electronic devices such as hearing aids, it may be possible to selectively extract and amplify natural sounds for the hearing impaired. The extracted signals could then be channeled to appropriate areas in the auditory cortex either through a conventional hearing aid, a cochlear implant or directly via chronically implanted electrodes in the brain. To initiate further research in this direction, we have begun to use commercially available software that allows us to directly modify the response properties and connectivity of model auditory neurons. Together with electrophysiological data, we believe this will pave the way to applications of combination-sensitivity in the development of sophisticated hearing aids in the future. Cartwright J.H.E., Gonzales D.L., Piro, O. (1999) Physical Rev. Letters 82:5389-5392. Langner G (1983) Experimental Brain Res. 52:333-355. Langner G (1997) Acta Otolaryngol Suppl. 532:68-76. Mittmann DH and Wenstrup JJ (1995) Hearing Res. 90:185-191. Supported in part by a NIDCD/NIH grant number DC02054 to J.K.

PA13

Tolerable hearing-aid delays estimated by effects on speech production

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

In previous work (Ear Hear. 1999;20;182-192) we examined the subjective disturbance produced by superimposition of bone-conducted sound and delayed sound through a simulated digital hearing aid as a function of delay time and degree of simulated hearing loss. Additional effects of delay would be expected on the basis of proprioceptive feedback, possibly leading to altered speech production. This effect was extensively studied in the delayed-auditory feedback (DAF) literature from the 1950's. However, the very long delays investigated there, and the elevated replay levels of the speech were not realistic for a hearing aid application.

Thirty-two normal-hearing subjects, (16 male, 16 female) were recorded reading short text passages in either a near-anechoic, or a small, reverberant office. Each subject wore bilateral behind-the-ear aids connected to Libby horns retained by foam earplugs. The aids were programmed either to be linear, or 3-channel, fast-acting, wide dynamic range compressors with a compression ratio of 2. The insertion gain of both types of aid was programmed to be 0 dB, flat across the audio bandwidth (0.2-6.5 kHz), for frontally presented speech with a free-field level of 65 dB SPL. Each aid had four delay settings of 7, 18, 28, and 40 milliseconds.

Sound arrived at the cochlea via two principal paths: (1) a direct path via bone conduction and leakage around the ear plug, producing a muffled timbre; (2) a delayed path via the hearing aid, imparting a brighter timbre to the sound. At long delays, the sounds from the two paths are distinctly discernible; at short delays, they blend with varying timbre. Two measures were recorded of the disturbance produced by the combination of the sound from the two paths. An objective measure was the reading rate of the subjects. A subjective measure was taken by asking the subjects to rate how disturbing they found the echo.

The results show that, for both environments, the rate of speech production was little affected by increasing delay up to about 30 msec, but decreased at a delay of 40 msec. In the near-anechoic room, the rated disturbance increased at a near constant rate with increasing delay, even from the shortest delay. In the office, the rated disturbance was little different at the two lowest values of delay, but increased at higher delays. There appeared to be little effect of the type of aid (linear or compression). [This project was funded by the Medical Research Council (UK). The 'Audallion' experimental digital hearing aids were provided by Audiologic, Boulder, Co. (USA)]

PA14

Signal processing in advanced hearing aids

Ole Dyrlund, GN ReSound

The recent generations of advanced analog and digital hearing aids offer a number of highly interesting signal processing functions, which significantly have increased user satisfaction. A variety of new advanced and even more beneficial functions are expected, when the next digital hearing aid generations are introduced in the coming years.

These functions can be split up into the following main groups based on the purpose of the signal processing:

- Signal processing to compensate for hearing loss of the patient (e.g. multiband wide dynamic range compression, spectral enhancement and frequency transposition / compression).
- Signal processing to improve the function of the hearing aid itself (e.g. feedback management, active feedback suppression, and microphone noise reduction)
- Signal processing to improve the communication abilities of the patient especially in noisy surroundings (e.g. directional systems, binaural processing and active noise canceling)
- Signal processing to enable adaptation to the acoustic environment (e.g. gain reduction based on signal dynamics / modulation and environment classification

The principles of typical signal processing functions within these categories will be outlined and examples of applications in existing and upcoming hearing constructions will be described. A main focus of the presentation will be an attempt to assess the potential contribution to user satisfaction from these advanced signal processing functions.

PA15

Costs and benefits of peak clipping in amplification for profound hearing loss

Frank Iglehart, Clarke School for the Deaf

Peak clipping, a common form of output limiting in hearing aids, has potential costs (distortion) and benefits (increased gain) to speech-perception ability. This study examined these costs and benefits for subjects with profound hearing loss under three conditions of clipped (by 15 dB) and unclipped speech in quiet.

Phoneme-recognition ability and sensation levels were measured for 16 subjects with profound hearing loss and, for comparison, two subjects with moderate hearing loss. First, speech stimuli were clipped to 15 dB below the highest instantaneous peak value and presented at highest comfortable level. The mean phoneme-recognition score for subjects with profound losses was 18.2%. Second, speech stimuli were presented unclipped with the same instantaneous peak level as clipped speech in the first condition. The mean score for the same subjects was 13.1%. Third, unclipped speech was presented at subjects’ highest comfortable level. The mean score was 17.1%. Comparisons of these scores show the benefit of increased gain available through the use of clipping and the lack of significant cost of distortion for hearing-aid users with profound losses. The benefit provided by clipping was equal to a 44% increase in channels of independent information (k = 1.44).Sensation levels were also measured. The mean level for subjects with profound hearing loss was 24.1 dB. For the two subjects with moderate hearing loss, the mean sensation level was considerably higher at 41.7 dB. Unlike the subjects with profound hearing loss, the phoneme-recognition scores for the two subjects with moderate loss were relatively unchanged across the three conditions. The relatively large sensation levels for these two subjects permitted a 15 dB attenuation of the signal without loss of phoneme recognition.

Under all conditions, vowels in consonant-vowel-consonant words were more easily recognized than either initial or final consonants, and initial consonants were more easily recognized than final consonants.

PA16

Biologically based speech signal enhancement

Leslie Smith, David Sterratt, Stirling Hearing Systems, University of Stirling, Scotland

We are drawing on knowledge of the brainstem nuclei processing to help design a component of a hearing aid that improves the signal to noise ratio (SNR) of speech and other harmonic sounds in background broadband noise. The first stage of the system comprises a bank of bandpass `cochlear' filters, simulating the movement of the basilar membrane.

The output of each of these filters is nonlinearly squashed to produce a signal related to the signal in the auditory nerve. This signal is passed through bandpass filters centred between 50 and 400 Hz, nonlinearly squashed and smoothed. The net result is a two-dimensional map of the amplitude modulation frequency in each bandpass filter similar to the outputs of AM-sensitive cells in the inferior colliculus. We assume the primary source of the AM is unresolved harmonics of voiced speech. We use this map to find the dominant fundamental frequency in the auditory input, and then selectively resynthesise the signal from the cochlear channels using the strength of the dominant fundamental frequency in each channel. The net result of the procedure should be to filter out channels that are not conveying much of the speech signal but that are conveying wide band noise.

Results to date have shown that the SNR of voiced speech can be improved with about 20 cochlear filters and 20 AM bandpass filters per cochlear filter. However the signals produced are do not seem any more intelligible than the original, noisy, versions. The problem seems to be that for short periods of time, the noise components envelope in each cochlear channel are such that they cause amplitude modulation in that channel at the dominant fundamental frequency. Consequently the channel is included for a very short period of time leading to a `frying' or `dripping water' sound. Our next aim is to combine this system with onsets, using common AM or fundamental onset to qualify channel selection. Eventually we hope to extend the system to a binaural system using interaural time and level differences. The approach we are taking shares some attributes of Computational Auditory Scene Analysis models. However, our goal is not as ambitious as a full auditory scene analysis, and in contrast to some models, our system is causal.

Currently we are in a software simulation phase, but eventually we hope to implement the system using Digital Signal Processing technology.

PA17

Speech enhancement and reverberation perception based on group-delay statistics

Hiroaki Nomura, Y, University of East Asia,
Y. Takahashi, and M. Toyama, Kogakuin University

Digital hearing aids have been developed using current digital technology. However the speech quality of a hearing aid is still not completely acceptable under reverberant and/or noisy conditions. This is because the human auditory system is not yet sufficiently understood for hearing aids to include a scheme for selective listening among competing sounds. Even with no power-spectrum changes included in the reception signal, reverberation effects will be auditory detected because the group delay due to phase distortion is also important perceptual cue for reverberation detection. We investigated the perception of reverberant effects and the dereverberation of speech according to the group-delay statistics of transfer functions in a reverberant field. This research can be applied to the development of an intelligent hearing aid.

The temporal effect of room reverberation on a speech sample appears in the waveform envelope. Reverberation sound buries the envelope troughs thereby, decreasing speech intelligibility. Thus even if power-spectrum changes due to reverberation remain, intelligible speech can be reconstructed by recovering the signal envelope from the reverberant speech. The modulation transfer function (MTF), which represents the transmission of the signal envelope in a linear system, was proposed to estimate the speech intelligibility under reverberant conditions. Increasing the magnitude of the MTF will improve speech intelligibility. The MTF can be derived from the impulse response of a reverberant space; however, this impulse response cannot be determined in practical situations where hearing aids are used. Blind-type methods of envelope recovery, including inverse filtering and cepstrum processing, have been investigated; however, dereverberation still has to be improved.

Perception of reverberation due to the temporal changes of the signal envelope can be estimated by using the group delay. The local fluctuations of the group delay from the linear phase of the transfer function increase in a reverberant space as the reverberation time becomes longer. We investigated a method for blind dereverberation by reducing the group-delay local variance of a reverberant speech signal while keeping the global phase trend. The reverberation effect due to the all-pass components can be reduced, although the power-spectrum changes due to the minimum-phase components will not be recovered.

PA18

Speech representation using the Hilbert envelope and a simple carrier signal

Michiko Kasama, Acoustic Consultant
MikioToyama, Kogakuin University

Auditory systems and hearing aids can now be developed because of the progress in signal processing. The characteristic of a hearing aid can be adapted to the frequency-dependent auditory range of a patient by using a signal processing technique. Furthermore, even profoundly deaf people can hear sound through cochlear implants. One of the next technical problems facing hearing aid development is detecting and reconstructing target speech from competing sound as a suitable input signal into the hearing aid. We have developed a method for reducing noise from noisy speech. This method is based on speech envelope reconstruction with speech-dependent or independent carriers.

When a person talks to someone directly or through a microphone in a noisy environment, we usually come closer to the microphone or speak louder than we normally would. We can thus assume that the peaks in the power spectrum record of a noisy speech signal are mainly due to the speech rather than the noise. Consequently, we should be able to synthesize the speech by extracting the high-energy speech components from the short-time Fourier spectrum of noisy speech. The number of extracted components are controlled frame by frame according to the signal-to-noise ratio. The speech quality is enhanced by 10 dB when the speech signal is reconstructed by extracting up to ten frequency components from noisy speech with an SDR of 0 dB. The temporal envelope of a vowel frame of the reconstructed speech is then analyzed by using the Hilbert transform. The fundamental frequency of a vowel frame is analyzed by using the auto correlation function. The vowel frame is expressed as a modulated sinusoidal wave whose envelope and carrier signal are the Hilbert envelope and a sinusoidal wave of the fundamental frequency, respectively. The reconstructed speech is intelligible and includes less noise. If the frequencies of the carrier signals in all the short frames are fixed (eg. 250 Hz), the reconstructed speech has no intonation but is understandable. The whole envelope obtained by low-pass filtering of the reconstructed squared signal recovered the modulation peaks and troughs where the noise was eliminated. The SDR of the whole envelope became higher by about seven times in the frequency band between 300 and 713 Hz, and 4 times higher in the band 713-1509 Hz. We confirmed that our developed method is an effective tool to recover temporal envelopes, which are important for cochlear implants, from noisy speech signals

PA19

Pilot experiments with a simulated hearing aid based on models of cochlear compression

Julius L. Goldstein, Washington University, Michael Valente, Washington University Medical School Roger Chamberlain, Peter Gilchrist, and Darko Ivanovich, BECS Technology

Cochlear mechanical response to sound normally provides rapid compression of signal levels, that is capable of adapting to a linear response. These useful nonlinear characteristics are degraded or absent in the impaired cochlea. Quantitative cochlear models were used to guide the design of multichannel bandpass-nonlinearity hearing-aid amplification that approximately restores lost nonlinear function (Goldstein, Hear. Res. 49, 39-60, 1990; 89, 52-68, 1995). For each channel the transducer characteristic is linear at low and high signal levels, and is smoothly joined by a compressive intermediate range. The transition from linear to compressive response at low sound levels is controlled by an adaptive compression threshold, whose quiescent level provides the desired gain for weak sounds. For sustained sounds at higher levels the compression threshold shifts upwards, as needed for effectively linear response. The adapted compression threshold is sustained through speech pauses in normal conversation, to minimize greater amplification of background than target signals. This dramatically improves the quality of compressively amplified speech in noise, with little loss of useful gain. Pilot psychoacoustic experiments with a design simulation were conducted to begin to assess the design benefits and develop fitting procedures. The hearing aid was simulated on a laptop running Matlab and excited with low predictability stimuli from the R-SPIN corpus (Bilger et al., J. Speech Hear. Res. 27, 32-48, 1984). An example stimulus sentence is "Miss White won't think about the crack." The subject must hear the final word in the presence of 12-speaker babble. Sounds were presented in a soundproof booth via insert earphones (4 kHz cutoff) driven by the laptop. Seven young normal-hearing subjects (ages 17-26 years) listened at a comfortable presentation level (~70 dB SPL), with various signal-to-noise ratios (SNR = 0, 2, 8 dB) and compression conditions (square root, cube root, low and high compression threshold). Five elderly subjects with mild to moderate hearing impairment were tested for the effects of hearing-aid input level at a fixed SNR of 8 dB, using fittings thought appropriate for their audiograms. For the normals, it was found that SNR is the major determinant of intelligibility over considerable changes in sound quality. The elderly subjects reported intelligibility scores similar to published results for advanced hearing aids (Valente et al., J. Am. Acad. Audiol. 9, 331, 1998), and judged the sound quality as equal to or better than their current aid.[Work supported at BECS by NIDCD SBIR Grant 1R43 DC04028-01, JL Goldstein, PI]

PA20

Real-time speech processing for digital hearing aids based on sinusoidal modeling

Peggy Nelson, University of Minnesota
Juan Carlos Tejero-Calado, University of Malaga, Spain
Janet Rutledge, University of Maryland

Digital hearing aids provide the opportunity for introducing new customized processing strategies for individual listeners with hearing loss. Recent attempts to customize digital processing algorithms have not been fully satisfactory in speech quality nor proven to improve speech understanding. Waveform parameterization models, such as wavelets or sinusoidal models, can be used in place of filter-based techniques to achieve speech processing. These models allow greater flexibility in the range of digital signal processing, and lend themselves to time-varying techniques. Presented here is a processing algorithm based on a sinusoidal speech model, implemented in real time, which identifies the important spectral peaks in overlapping time windows. The current application, which has been implemented on a TMS320C30 microprocessor, uses 7.5 msec analysis frames and 30 msec Hamming windows, leading to a 4 to 1 time overlap. A 256 point FFT is used to provide sufficient resolution for the speech sampled at 8.011 kHz. Synthesis is done using an inverse FFT. This base algorithm has been used to implement amplitude compression, frequency scaling, and spectral sharpening of speech. Preliminary listening tests indicate that listeners with moderate hearing losses showed benefit from the processing algorithm and positive judgements of sound quality. [This work was supported by the NIDCD (K08 DC00124, 1R03DC04125-01), NSF, and the University of Malaga]

PA21

In-the-ear vs. half-shell directional hearing instruments

Theresa Velde and Oleg Saltykov
Siemens Hearing Instruments Inc.

Directional hearing aids have enjoyed a comeback since the introduction of improved two-microphone and one-microphone directional systems in the late 1990's. One drawback of the directional hearing aids remains the implementation in BTE and ITE hearing aids. Customers frequently choose cosmetic considerations (e.g., ITC or CIC) over improved listening in noise (directional ITE or BTE). Additionally, the size of the ITE with a directional mic system can be a limiting factor in patient selection. Many individuals with small ears do not have the required space for a directional ITE. As a solution to these two obstacles, directional-mic systems are being implemented in half-shell designs. These new designs raise the question -- does a directional-mic system in a half-shell hearing aid provide the same benefit as a directional-mic system in an ITE hearing aid? While the half-shell directional hearing instrument may have the same or even better directional characteristics as the ITE directional instrument, the placement of the directional mics deeper into the concha may affect user performance. This study compares speech intelligibility in noise, as estimated by the Hearing in Noise Test (HINT), from hearing-impaired listeners fit with directional ITE and half-shell instruments. Ten hearing-aid users served as subjects. Binaural ITE and half-shell programmable directional hearing aids were built for each subject. The speech signal was presented through a loudspeaker at 0° azimuth. The background noise was presented through five loudspeakers at 30° and 330°. Testing was completed in a moderately reverberant room. The implications of the results on fitting directional hearing aids will be discussed.

PA22

Fixed polar pattern based adaptive directionality algorithm for two nearby microphones in endfire orientation

Fa-Long Luo, Jun Yang, Chas Pavlovic, Michael Nick, and Laurel Olson
GN ReSound Corporation, Redwood City

People can get benefits from the hearing devices with directionality which use the direction difference between the target signal and the noise. There are two types of directional devices, one is with fixed directionality or say with the fixed-polar pattern, the other is with adaptive directionality which can track the varying or moving noise. There have been many techniques for achieving directionality in both fixed mode and adaptive mode, however, most of these techniques can be neither immediately employed nor implemented in hearing aids mainly because of the limit of hardware size, computational speed, mismatch of microphones, power supply and other practical factors existed in hearing aids. The most common technique which is being used and can be realizable in hearing aids is directional microphone or dual-omnimicrophone system with some fixed polar pattern such as cardiod pattern or hypercardiod pattern, etc. For hearing-impaired people, there is an increasing demand for hearing aids Based on these problems, we have developed and implemented a more practical and effective adaptive directionality system for hearing aids with two nearby endfire orientation microphones. This adaptive directionality system is mainly based on an adaptive combination of two fixed polar patterns so as to make the null of the combined polar-pattern of the system output to be always toward the direction of the noise. The null of one of these two fixed polar patterns is at zero degree (straight ahead of subject) and the other's null is at 180 degree. These two polar patterns can be either cardiod or hypercardiod; we selected cardiod in our implementation. The first fixed polar-pattern can be implemented by delaying the front microphone signal with the value d/c (d is the distance of two microphones and c is the speed of sound) and subtracting it from the signal of the rear microphone. Likewise, the second polar pattern is implemented by delaying the rear microphone signal by d/c and subtracting it from the fr. The hardware implementation of this proposed system has been realized and various tests have been made. These test results have demonstrated the accuracy and effectiveness of this proposed scheme. The clinical tests are being performed and results will be reported at this meeting.

PA23

Hybrid adaptive beamformer (HAB) for improved speech recognition in noise

Gary Gibian, Scott Shaw, Walter Koroljow, and Andy LaRow, Planning Systems, Inc
Peggy Nelson and LaGuinn Sherlock, University of Maryland and University of Minnesota

An algorithm combining adaptive and fixed-weight beamforming has been evaluated with hard-of-hearing (HoH) listeners in a variety of environments using a four-microphone array and a wearable processor. Twenty-eight listeners with mild to moderate hearing losses were fit monaurally with behind-the-ear (BTE) hearing aids using the NAL-R fitting algorithm. Comparisons were made between listeners' speech reception thresholds (SRTs) in noise with a conventional BTE hearing aid using one of three inputs: a) conventional single-microphone, b) built-in dual microphone, and c) output of the 4-microphone hybrid adaptive beamformer (HAB) device with the array mounted on an eyeglass frame. The HAB processor was connected to the BTE hearing aid through the direct audio input (DAI) boot. Hearing in Noise Test (HINT) sentences were presented with single and dual noise sources in three room environments: a) sound booth, b) favorable classroom, and c) noisy hospital cafeteria. Listeners rated the three hearing aid systems and indicated their preference in paired comparison testing. HoH listeners' improvement in SRTs using the HAB ranged from 2 to more than 20 dB, depending on the individual, the environment and number of noise sources. Improvements over omnidirectional microphones, averaged over noise conditions, ranged from 8.7 dB in a classroom to 6.5 dB in a cafeteria. Similar averaged improvements over dual microphones ranged from 3 to 5 dB. Quality ratings indicated good acceptance of the HAB microphone systems. Listeners judged the HAB and dual-microphone systems to be quieter than the omnidirectional microphone. Approximately 80% of the listeners preferred the HAB system over omnidirectional, and approximately 70% preferred the HAB over the dual-microphone systems in paired comparisons. [The majority of this research was supported by grants 1 R43 AG 13515-01, 2 R44 AG 13515-02 and 5 R44 AG 13515-03 from the National Institute on Aging, for which the authors are very appreciative]

POSTER GROUP B

Friday 8:00AM – 10:00PM

PB1

Benefits and drawbacks of FM microphones as hearing aid accessories for adults

Arthur Boothroyd, Lexington RERC

Phoneme recognition in CVC words was measured as a function of speech level and speech-to-noise ratio in 12 hard-of-hearing and late-deafened adults. Hearing losses ranged from mild to severe. Threshold configurations ranged from flat to steeply sloping. Ages ranged from 55 to 80+ yrs. Performance was measured: a) under headphones in quiet, b) unaided in the sound field in quiet, c) aided in quiet and noise (spectrally matched to the talker's long-term-average spectrum), and d) aided with the use of an FM microphone situated so as to increase speech input by 15 dB without a change in noise input. Amplification was linear with compression limiting. Initial fitting was to NAL-R, with modifications according to subject response. As expected, unfavorable signal-to-noise ratios caused a dramatic drop of aided performance - most of which could be recovered with the use of the FM microphone - either alone or combined with input from the aid's own microphone. Compression limiting in the FM transmitter, however, reduced the FM benefit below its theoretical maximum. Laboratory tests were followed by a 2-week trial period in which subjects were asked to keep a diary indicating amount and circumstances of use, benefits noted, and problems encountered. The process ended with administration of a questionnaire that included not only perceived benefits and drawbacks but also probed the subject's lifestyle and needs. Subjects with steeply sloping hearing losses tended to perceive less benefit from the FM than did those with flat audiometric configurations. Perceived benefit tended to increase with decreasing hearing loss - partly because the FM was used for remote listening - providing, essentially, super-normal hearing. Problems included the inability to deal with multiple talkers, the persistence of noise interference for high noise levels (especially among those with poor high-frequency hearing), equipment complexity, and confusion about the exact purpose and optimal operation of the equipment. Some subjects with long-standing hearing loss had pre-established avoidance of the listening situations in which an FM link would be most useful. Following trial, the interest in purchasing FM equipment was low - even among those expressing very positive reactions. FM benefits in controlled test situations are considerable. Nevertheless, effective incorporation of this technology into the management of adult hearing impairment will require careful consideration of its limitations, subject needs, appropriate design, and education and counseling. [Research funded by NIDRR grant # H133E980010]

PB2

The sources of wind noise in hearing aids

Harvey Dillon, Inge Roe and Richard Katsch, National Acoustic Laboratories, Australia

Hearing aid users complain about the noise generated in their hearing aids by wind. This noise is caused by turbulence, but the source of the noise is little understood.

Measurements in a wind tunnel indicated that noise levels of 100 dB SPL at the input of a hearing aid are generated by even a light breeze. Visual observations using smoke trails indicate complex flow and turbulence patterns generated by the nose, cheek-bones and pinna. These observations are supported by laser doppler-velocimeter measurements of wind velocity around the pinna and in the concha.

The spectrum of wind noise was measured by mounting hearing aid microphones on KEMAR in the locations occupied by CIC, ITC, ITE and BTE hearing aids, while KEMAR was positioned within an acoustically deadened wind tunnel. The spectral shapes and levels varied strongly with the angle of incidence of the wind and with the type of hearing aid being simulated. As a generalisation, however, wind noise has a broad spectral shape with a low-frequency emphasis, and contains significant energy over the frequency range 20 Hz to 2000 Hz.

Comparison of the velocity measurements, the noise measurements, and the underlying theory regarding the generation of turbulence led to the following conclusions:

1. The turbulence sensed by hearing aid microphones is generated by the head, the pinna, the tragus, and the microphone inlet port.
2. Larger obstacles generate low-frequency noise and smaller obstacles generate higher-frequency noise.

3. The relative importance of the turbulence sources varies with wind direction and the type of hearing aid, but generally the physically larger obstacles generate more noise.
4. BTE hearings aids are much more prone to wind noise than the other three styles.
5. CIC hearing aids are generally less prone to wind noise, but this is not true for all wind directions.
6. Barriers, such as the tragus, can protect from wind noise for some wind directions, but generate wind noise for other wind directions.

The funding jointly provided by Oticon, Widex, Phonak and GN Resound to support this research is gratefully acknowledged.

PB3

Auditory acclimatization to amplified speech

Kevin Munro and Mark Lutman
University of South Hampton

Improvements in hearing aid benefit over time have been demonstrated in a number of studies using a variety of objective and subjective outcome measures. Parallel findings have been shown to occur in other psychoacoustic abilities such as intensity discrimination. Taken collectively, there is irrefutable evidence for the existence of auditory acclimatisation. However, the majority of recent studies have failed to demonstrate an acclimatisation effect. One explanation for the negative findings is that studies have used subjects with varying degees of hearing impairment, a mixture of unilateral and bilateral fittings, varying degrees of audibility and varying degrees of hearing aid use. The aim of the present study was to measure benefit in a more homogenous group of subjects who were making good use of a monaural hearing aid that provided a significant improvement in audibility.

Subjects comprised 16 new hearing aid users with a mean age of 70 years who had a symmetrical moderate high frequency sensorineural hearing impairment consistent with natural ageing. They were fitted with a monaural hearing aid that provided a mean insertion gain in excess of 30 dB at 4 kHz. The self-reported use of the hearing aid was typically 8-12 hours/day. The normally unaided ear was used as the control condition. The main outcome measure was the Four Alternative Auditory Feature (FAAF) test, which is a low redundancy speech test that is dependent on recognition of amplified high frequency acoustic cues. The material was presented at a maximum sound pressure level of 60 dB. Speech shaped noise was present at a level that resulted in an aided score of around 71% on the first test session. The subjects were tested at 3-week intervals over a period of 6 months. They were tested at two gain settings: fixed gain and a user adjusted gain selected on each test session.

The results show an initial benefit (aided minus unaided) score of 14%. There was no significant change over time on either gain setting. There was a significant increase of around 5% in the aided and unaided scores. However, this increase was also present in the control (normally unaided) ear and was attributed to procedural learning. The software provided feedback to the subject by informing them of the correct response after they had made their selection. The experiment has since been repeated without this feedback. The procedural learning effect was absent, but once again, there was no increase in benefit over time.

Close inspection of the published studies show that the four that revealed a significant improvement in benefit over time all achieved a mean initial benefit score of around 4-7%. It is possible that acclimatization is related to the degree of complexity of extracting the acoustic information from the background noise. We are presently collecting data in a third experiment using a variety of presentation levels and signal-to-noise ratios to test the hypothesis that auditory acclimatisation is measurable only under difficult listening conditions.

PB4

Validation of prescriptive technique to select saturation sound pressure level

Jill Preminger and David Cunningham
University of Louisville
Arlene Neuman, City University of New York

Recently, Dillon and Storey (1998) proposed a procedure for the selection of the SSPL in hearing aids. In this procedure the SSPL is determined based on pure tone thresholds, and not on individual loudness judgments. They validated this procedure in linear, single-channel hearing aids (Storey, Dillon, Yeend & Wigney, 1998), and found a very high correlation between the SSPL selected with the theoretical procedure and the SSPL selected based on listeners' subjective selections in the laboratory and in the real-world. Dillon & Storey (1998) proposed modifications to their SSPL selection procedure when fitting multichannel hearing aids, but this has not been validated.
The purpose of the present project was to validate Dillon and Storey's SSPL selection procedure for two-channel hearing aids.

Twenty experienced hearing aid users with mild to moderately-severe hearing loss listened to speech in quiet and speech in noise, while the SSPL of each channel of a two-channel digital hearing aid was varied. Subjects rated loudness, quality and clarity. The range of acceptable SSPL settings were determined from these subjective ratings. Statistical analyses determined whether the Dillon & Storey method using pure tone thresholds, or whether actual loudness judgments were better predictors of the measured SSPL settings.

PB5

Evaluation of initial procedures for fitting digital hearing aids

José I. Alcántara, Josephine Marriage and Brian C.J. Moore, University of Cambridge, England

Many hearing aids incorporate fast-acting compression acting independently in multiple channels. In fitting these hearing aids, an initial procedure is required to give a realistic "starting point" for the gains and compression ratios required for a particular hearing loss. The objective of this study is to assess the adequacy of three procedures for this purpose: (1) the Cambridge procedure for loudness equalisation, which aims to create equal loudness per critical band for typical speech stimuli (CAMEQ; see Moore et al., 1999); (2) The Cambridge procedure for loudness restoration, (CAMREST; see Moore, 2000); (3) The DSL I/O procedure (Cornelisse et al., 1995).

To date, twelve subjects with varying degrees of sensorineural hearing loss and hearing aid experience have been fitted with the Danalogic163D fast-acting 14-channel digital hearing aid. Each of these subjects has worn the aid configured using at least one of the three procedures for at least four weeks. The order of use of each procedure has been counter-balanced between subjects. All fittings have been based on real ear aided gains (REAGs) measured using a probe microphone system. Aid benefit has been evaluated after each period of use of a procedure using the APHAB questionnaire and speech reception threshold (SRT) measures in quiet and in two types of background speech-shaped noise; steady and modulated with the envelope of a single talker. The SRT is the signal-to-noise ratio (SNR) at which identification accuracy of the ASL sentences is 50%.
Preliminary results indicate that the CAMEQ procedure results in slightly lower (better) SNRs than the CAMREST procedure. The DSL I/O program results in higher (worse) SNRs than either of the Cambridge procedures. The differences are of the order of 2-3 dB. The SNRs for the modulated noise condition are lower than for the steady noise condition, for all three procedures, suggesting that subjects were able to exploit the temporal dips in the background noise to improve their speech intelligibility. Anecdotal reports indicate that subjects prefer the Cambridge programs to the DSL I/O program, for a variety of real-life situations.
References: Cornelisse, L. E., Seewald, R. C., and Jamieson, D. G. (1995). "The input/output formula: A theoretical approach to the fitting of personal amplification devices," J. Acoust. Soc. Am. 97, 1854-1864. Moore, B. C. J. (2000). "Use of a loudness model for hearing aid fitting. IV. Fitting hearing aids with multi-channel compression so as to restore "normal" loudness for speech at different levels," Br. J. Audiol. (in press). Moore, B. C. J., Glasberg, B. R., and Stone, M. A. (1999). "Use of a loudness model for hearing aid fitting. III. A general method for deriving initial fittings for hearing aids with multi-channel compression," Br. J. Audiol. 33, 241-258.

PB6

Acclimatization? Possible evidence seen in people with severe to profound hearing losses

Francis Kuk, Widex Hearing Aid Co.

Although the concept of acclimatization has been introduced for many years, researchers and clinicians disagree on the magnitude of this observation. One hypothesis for the mixed observations may be related to the interactions between the amount of "auditory deprivation" of the subjects and the ability of the hearing aids to provide the needed missing acoustic stimuli. In other words, acclimatization was not observed in some cases because the subjects never missed the acoustic stimuli used in the evaluation. Because patients with severe-to-profound hearing losses have frequently been under-amplified for low input sounds but over-amplified for high input sounds, this group of subjects may be ideal to examine this hypothesis.

A group of 20 subjects with a severe-to-profound degree of hearing loss was fit with the Widex P38 hearing aid, that among other features, utilize a low compression threshold at 20 dB HL, and high level compression above a conversational level in all three of its channels. Subjects' speech recognition score (as evaluated on the SPIN test at 50 dB SPL, 65 dB SPL, and 75 dB SPL (+10 SNR) was obtained with their own hearing aids, as well as with the P38 at the initial fitting, one month, and three-months post-fitting. No training was provided. Subjective questionnaires were also administered at one month and three months post fitting.

Results on SPIN test showed higher speech recognition scores with the P38 than the subjects' own hearing aids at the 50 dB and 65 dB SPL presentation levels at all evaluation intervals. Furthermore, the performance at one month ws significantly higher than the SPIN score at the initial fitting. Performance between the three and one month-post fitting intervals were similar. The difference seen at the 75 dB SPL presentation level was not significant.

These results suggest that acclimatization may account for the observation. Furthermore, it raise several interesting implications. First, is a 30-day trail period sufficient for hearing aid adjustment? Second, what is the clinical implication during the fitting process? Third, is the course of acclimatization different for different stimulus conditions (e.g., quiet versus noise)? These issues will be discussed.

PB7

Developing assistive listening system performance standards for people with hearing loss

Mathew Bakke, Faye Erickson and Harry Levitt, Lexington Center for the Deaf
Mary Rose McEnerney, Hackensack University Medical Center

The objective of this study was to establish guidelines for specifying the acceptable output characteristics of assistive listening devices for people with hearing loss. Fifty-nine adult listeners (49 with hearing loss and 10 without) listened diotically to sentence materials that were subjected to three different types of distortion: reverberation and background noise, internally generated induction loop noise, and peak clipping. Listeners provided ratings along a four-point scale (1. excellent, 2. good, 3. marginal and 4. unacceptable) as to the quality of the materials presented. A minimally acceptable criterion was selected and results for the listeners with hearing loss were compared with that criterion to arrive at minimally acceptable 1) output and dynamic range levels, 2) Speech Transmission Index (STI) level, 3) signal to noise ratio for internally generated noise, and 4) peak clipping. Results formed the basis for a set of recommendations submitted to the U.S. Access Board for the purpose of establishing standards for ALDs in public areas.

PA8

Optimal fitting of three algorithms in real-life noisy situations

Bas A. M. Franck, Jan Koopman, Wouter A. Dreschler, Academic Medical Center, The Netherlands

In order to improve speech intelligibility in noisy situations, different signal-processing schemes can be applied. In this study we investigated the optimal fitting of three algorithms. Background noise is attenuated by means of binaural noise suppression (Kollmeier et al., 1993). Specific speech segments are spectrally and temporally enhanced by means of spectral enhancement (Baer et al., 1993) and fast (phonemic) compression (Goedegebure et al., 1999). The purpose of the study is twofold: first to investigate the effects of several combinations of algorithms, and secondly to evaluate the reliability of a clinically more relevant fitting procedure. To study the effects of combination of algorithms, we applied a modified Round-Robin procedure (Studebaker et al., 1982). The second goal concerns the question how consistent a fast ‘hill-climbing’ method is, in order to predict the optimum of the three separate algorithms. This ‘Simplex’ procedure has been studied earlier (e.g. Neuman et al., 1987; Kuk and Pape.Six hearing-impaired subjects with cochlear hearing losses and six normal hearing subjects participated in this study. For both procedures, these groups were asked to evaluate two consecutive, differently processed sentences on aspects of speech intelligibility and listening comfort, separately. Sentences were presented in six conditions, defined by three binaurally recorded noise types and two signal-to-noise ratios.

Analysis of the Round-Robin data revealed that preferred settings of normal hearing and hearing-impaired subjects only differ significantly for spectral enhancement. For hearing-impaired subjects, the optimum for one of the algorithms changes when the other two are varied. The evaluation criterion had no effect on the preferred setting.

According to an analysis if variance, there are no differences between test and retest data for the Simplex procedure. However, variation of the initial setting influences the final estimate considerably. Most and largest deviations occur for the speech intelligibility criterion. Therefore, listening comfort seems most suitable for optimal fitting by means of the Simplex procedure. Surprisingly, it appears that most significant effects apply to the normal hearing group.

PB9

IEC fitting: New framework of hearing aid fitting based on computational intelligence technology, a user's preference for hearing

Hideyuki Takagi, Kyushu Institute of Design
Miho Ohsaki, Shizuoka University

We propose an interactive evolutionary computation (EC)-based hearing aid fitting method. Our proposed method optimizes hearing aid parameters using any sounds in our daily life according to how hearing aid users hear, which is completely different from conventional fitting methods.

The biggest and essential problem lays in conventional fitting methods is that an engineer or doctor who cannot perceive the hearing of a hearing impaired person must adjust a hearing aid with the clues from measured partial auditory characteristics. The second problem is that the conventional fitting is based on auditory characteristic measurement using pure tones or band pass noise. It is hard to believe that the best fitting using non-daily life sounds becomes the best for normal sound in our life; for example, it is easy to imagine that the uncomfortable sound pressure level of noise must be lower than that of music or speech. The third problem is that the conventional fitting based on pre-measured auditory characteristics cannot reflect users' preference that comes from a upper psychological layer than perception into the fitting.

Our proposed IEC Fitting fundamentally solves these problems. Since users cannot adjust their hearing aids in general but evaluate the processed sound, we combine user capability for evaluation and an optimization method for adjusting hearing aids. Most of optimization methods require gradient information of a searching landscape, but a psychological evaluation space, such as hearing landscape, cannot be differentiated. We adopt EC inspired by biological evolution as one of optimization methods requesting no gradient information. The IEC Fitting is an interactive EC system for hearing aid fitting that combines the EC search in a parameter space of a hearing aid and human evaluation based on his/her hearing preference.

Conventional loudness compensation based on a loudness function and new hearing aid filter with the IEC Fitting are compared. The input-output characteristics of our developed filter is formed by combining seven 3-D Gaussian functions, and the IEC Fitting adjusts the shape parameters of the Gaussian functions. In our experimental evaluation with three hearing impaired persons using clean speech, speech with multi-talker noise, and several music, the IEC Fitting was superior to the conventional loudness compensation on VCV syllable articulation test, preference test for speech sound quality with sign test, and preference test for music sound quality with sign test. We are now applying this method to the fitting for commercial hearing aids with several number of hearing impaired persons.

PB10

A low-complexity digital AGC circuit for dynamic range control of an A/D converter

Jos Leenen, Bert de Vries
Beltone Netherlands B.V.

Constraints on power consumption of the Analog-to-Digital Converter (ADC) in a digital hearing aid limit the dynamic range of ADC output to about 75 [dB]. This is a bit low relative to the dynamic range of a normal hearing person (about 120 [dB]) and one could argue that the range of interesting sounds around us extends at least over 90 [dB]. One way to increase the input dynamic range in a hearing aid would be to preprocess the incoming sound by a compressive AGC circuit prior to feeding the ADC. Such an AGC circuit would be realized in analog electronics which limits both programmability and accuracy.

We present a digital solution that accomplishes the same task, but, on Beltone's programmable digital DSP core, leaves open the possibility to adapt the system to a particular patient and/or product. The fundamental idea is to measure the power of the ADC output and feed back a bit that switches an analog premultipier. The premultiplier will shift the input signal by 25 [dB] or 0 [dB]. After A/D conversion, the premultiplication factor is accounted for in a digital compressive AGC, thus realizing an programmable ADC dynamic range with small computational overhead. Our presentation will include audio demonstrations.

PB11

New electronic technology for increased battery life in hearing aids

Joong-Seok Moon William Athas, Sigfrid Soli, Kisup Chong and, House Ear Institute
Nestoras Tzartzanis, Fjitsu Research Labs of America

Continual progress in the fabrication of integrated circuits has resulted in stupendous improvements in speed performance. Personal computers, workstations, file servers, set-top boxes, etc., have greatly benefited from the clock-frequency gains while power dissipation has continued to spiral upwards. Many emerging and existing applications for real-time computing systems, such as those of bioengineering, require only modest speed improvements but large reductions in operating power. Advances in digital signal-processing algorithms that provide superior user-level functionality comes at the cost of reduced battery life. This problem is extreme in hearing aids because of the stringent demands placed on size and weight.

For this presentation I will describe energy-recovery CMOS which is a new low-power technology that leverages the intrinsic high-speed nature of the integrated circuit chips to reduce power dissipation. In energy-recovery CMOS, circuit energies inside the integrated circuits are recovered and re-used, i.e., recycled. Normally these energies would be dissipated as heat. The key to recycling is a method for efficiently moving energy between internal circuits nodes without giving up the energy as heat. Improvements in circuit speed due to improvements in the underlying fabrication technologies directly translate into higher efficiency in energy transport and overall less power dissipation. This technology and approach has been tried out in a number of real-world computing problems. I will present simulation and laboratory results of experimental programmable chips that demonstrate the low-power advantages of energy-recovery CMOS

PB12

A prescription for maximizing the intelligibility of speech in the presence of various noises

Michael Fisher, Richard Katsch, Harvey Dillon, Teresa Ching and Gitte Keidser
National Acoustic Laboratories, Australia

One of the major challenges for the hearing-impaired person is understanding speech when background noise is present. Several studies have shown that hearing-aid users prefer different amplification characteristics for understanding speech in different background noise conditions. The NAL-NL1 hearing-aid prescription is based on the rationale of maximizing the predicted intelligibility of speech in quiet while constraining loudness to be equal or less than that perceived by normal-hearing people. This same rationale has been applied to speech in noise to obtain an optimal amplification characteristic that is dependent on the noise spectrum in addition to the speech spectrum and hearing loss.

The intelligibility of the speech in noise is predicted using a modified version of the Speech Intelligibility Index (SII) method (also known as the Articulation Index) and the loudness is predicted using a loudness model that allows for the effects of hearing loss. In quiet, the amplification characteristic matches that of the NAL-NL1 which at average input levels is similar to that prescribed by the NAL-RP formula.

In noise, less gain is generally prescribed at those frequencies where signal-to-noise ratio is worst, and more gain is generally prescribed at those frequencies where signal-to-noise ratio is best. The optimum gain-frequency response is, however also affected by upwards spread of masking and the decision never to prescribe negative gain at any frequency. At high input levels, speech intelligibility is maximised when the total loudness is lower than that perceived by normal-hearing people.

PB13

Development of a low power digital hearing processor

Neeraj Magotra, Frank Livingston, Mhamed Ibnabdeljaliel and Keith Guitierrez, Texas Instruments
Emily Tobey, Linda Thibodeau, Philip Loizou and Richard Wiggins, University of Texas at Dallas

Over the past decade there has been a shift in the hearing aid industry towards commercial hearing aids with digital signal processing (DSP) capabilities. However, in order to meet the size and power requirements, the current devices are application specific integrated circuit (ASIC) solutions. Constantly upgrading these ASICs can become a costly process and also detract from the goal of developing digital speech processing strategies that provide maximum benefit to the end user – the hearing impaired individual. By switching to a commercially available programmable DSP processor, hearing aid companies could significantly reduce their costs and increase their potential for reaching a larger portion of the hearing impaired population with a flexible, effective and affordable solution.This paper describes the application of TIs family of fixed-point DSPs as fully programmable, low power, binaural, digital hearing processors. Specifically, the TMS320C54X DSP chip allows researchers to explore new algorithms while providing the portability required for laboratory as well as real world testing and final implementation. It can implement a real-time system capable of processing two input speech channels at a minimum of 20 KHz sampling rate for each channel and driving a stereo headphone output. It could provide for the implementation of custom algorithms thus providing the hearing impaired subject with a device tailored to his or her hearing loss.

Furthermore, code developed on this DSP chip will be compatible for porting to Texas Instruments new low-power fixed-point DSP chip – the TMS320C55X. Its core has a power consumption of 0.05mW/MIP, the best in the industry. This device also has improved MIPS capabilities that would enable the real-time implementation of even more complex digital speech processing algorithms for the hearing impaired. Researchers at Callier are studying the feasibility and effectiveness of this approach of providing a programmable, customizable and effective solution for the hearing impaired.

PB14

Adaptive feedback cancellation as a system identification problem

Marion Schabert and Walter Kellermann
University Erlangen-Nuremberg

Feedback is a persistent problem in hearing aids. The hearing aid output signal quality should remain unaffected by any feedback cancellation method and the effectiveness of the method should be independent of the fast changing conditions of the feedback path. The goal in employing a feedback cancellation scheme is to provide more gain to people with severe hearing loss without the annoying occurance of howling.
Feedback cancellation in hearing aids uses an adaptive filter to estimate the transfer function of the feedback path. The hearing aid output signal is filtered by adaptive weights producing an estimate of the feedback signal. As no direct access to the actual feedback signal itself is possible, the microphone signal has to serve as the desired signal for the adaptive algorithm. Subtraction of the estimated feedback from the microphone signal results in the error signal which is the essential quantity for algorithms based on the method of least squares.

In this work the reasons for insufficient performance of this method are derived. We investigate the applicability of the normalized least mean squares (NLMS) algorithm and the recursive least squares (RLS) algorithm for feedback cancellation in digital hearing aids. The latter, in general, is computationally more demanding but shows faster convergence properties. Especially two problems implied by the closed loop structure given for hearing aids prevent successful system identification: a) the error signal to be minimized by the adaptive algorithm is strongly disturbed by the recorded source signal, i.e. it comprises the difference between the actual feedback and the estimated feedback and additionally the source signal b) correlation between the source signal and the feedback signal or hearing aid output signal, respectively.

The second problem accounts for a deviation between the optimum set of filter weights approached by the adaptive algorithm and the actual feedback path. The first problem forces the choice of a small step size and thus slow convergence. Especially this low feedback power relative to the power of the desired signal - acting as an interferer for the system identification - prevents the application of a straightforward system identification structure to the problem of feedback cancellation..

Simulations using real data illustrate the differences of the two algorithms and the performance of adaptive feedback reduction in general.

PB15

Suppression of feedback oscillation: Why the simple methods don't work

Steve Thompson, Knowles Electronics, LLC

Feedback oscillation may be considered a good thing or a bad thing depending on the system in which it occurs. In musical instruments, both wind instruments and bowed string instruments, the tone is produced by feedback oscillation. The quality of the musical instrument is determined by the ease of producing a stable and controllable musical note. In this context, feedback oscillation has been studied at great length to learn and understand the ways of the instrument craftsman in optimizing the design of the musical instrument.

In hearing aids, of course, feedback oscillation is always undesirable. The purpose of this paper is to apply the understanding of feedback oscillation learned from musical acoustics to the problem of feedback oscillation in hearing aids.

Most attempts to use feedback theory to reduce feedback oscillation in hearing aids have been based on a simple linear theory. The linear explanation of feedback oscillation is that an oscillation will occur when the feedback signal has magnitude of at least unity and is in phase with the input signal. Under these conditions, the feedback oscillation will grow to infinity. From this theory, all that is necessary to avoid an oscillation a shift of the phase of the feedback signal to keep it from adding to the input. However, attempts to implement a phase shift to suppress feedback oscillation have been uniformly disappointing.

The reason is that feedback oscillation is inherently a large signal phenomenon, subject to a significant nonlinearity. This nonlinearity is a necessary part of the process since it keeps the oscillation finite. However it also allows the system to be far more versatile in using the feedback energy to generate and maintain the oscillation. Changing the phase of the transfer function is generally not sufficient to eliminate feedback, but will only change the feedback frequency. Robust methods to reduce the tendency for feedback oscillation are 1) reduction of the magnitude of the feedback through isolation or gain reduction (the traditional approach), and 2) adaptive estimation and cancellation of the feedback signal as is currently done in some DSP hearing aids.

Note to reviewers: This paper is offered for the session on "Hearing Aids: Feedback and Fitting." You will note that it does not fit exactly within the guidelines for that session. However it is intended to motivate the value of DSP based feedback cancellation systems. In that context it may provide a useful introduction to the subject.

PB16

Band-limited feedback cancellation for reducing feedback oscillation in hearing aids

Shawn X. Gao, Sigfrid D. Soli, Daniel J. Freed, House Ear Institute
Chi Hsiang-Feng and Abeer Alwan, University of California, Irvine

Most of the reported feedback cancellation methods use adaptive filters to track the feedback signals and cancel them using wideband approaches. However, widely-used wideband adaptive feedback cancellation for hearing aids does not provide satisfactory performance in terms of oscillation suppression and sound quality. In this research, the fundamental problems of wideband feedback cancellation will be investigated theoretically and experimentally. A band-limited feedback cancellation algorithm is developed.

In adaptive feedback cancellation, the signal from the external sound source acts as an interference signal. When a wideband feedback canceller is adopted, a bias will be introduced in the feedback path estimation if the interference signal is correlated with the feedback canceller input, which is often the case. Since the feedback canceller is adaptive, the non-zero interference signal will introduce a noisy coefficient adaptation which results in distortion at the hearing aid output. Using the coherence function to characterize the nonlinear distortion caused by the filter coefficient adaptation, we showed that the distortion is significant when the bias and coefficient variation are large. From the minimum-mean-square-error (MMSE) analysis, we also showed that the bias is significant in the frequency-band where most of the external signal energy is concentrated. We conclude that distortion is apparent in the frequency bands where most of the external signal energy is located, which is at low frequencies for speech. The sound quality and intelligibility of the output speech are compromised.

In addition, the time-varying behavior of the adaptive feedback canceller makes oscillation suppression difficult because the adaptive filter always functions better in the band where large energy exists. This means that the adaptive feedback canceller operates on oscillation frequencies only when the energy of the oscillation components in the adaptive filter input and error is comparable to or greater than the peak energy of the signal from the external sound source. If the external sound has significant energy in the bands where oscillation frequencies are not located, the oscillation suppression ability of the feedback canceller will be compromised.

Utilizing the characteristics of feedback oscillation, we develop a band-limited adaptive feedback cancellation algorithm, which has better cancellation efficiency, convergence behavior, and output sound quality than widely-used wideband algorithms.
Convergence analysis of feedback cancellers with the wideband and the band-limited algorithms is conducted, and the performance is evaluated using computer simulations and subjective evaluation, which is based on a digit hearing aid prototype with a portable digital signal processor.

PB17

Predicting memory for hearing aid orientation information and treatment outcomes

Judith Reese, Theresa Hnath-Chisolm, Harvey Abrams and Cathy McEvoy, University of South Florida

Cognitive styles, general cognitive abilities, and memory self-efficacy are evaluated as predictors of 100 new hearing aid users' memory for hearing aid orientation (HAO) information and hearing aid outcomes. The study is a prospective, nonrandomized, non-experimental survey of hearing aid recipients. Results of multiple regression analysis will indicate whether certain aspects of cognitive function, in addition to abilities, should be considered in hearing aid delivery.

One hundred new hearing aid wearers serve as participants. The cognitive tests and clinical protocols that comprise the research battery are administered over the course of the three routinely scheduled visits. Participants are included on the bases of the following criteria: age 55 or older; 2) adult onset hearing impairment; 3) at least a mild, high-frequency, sensorineural hearing impairment (pure tone average of 30 dB HL or more for 1000, 2000 and 4000 Hz in the better ear); 4) no previous hearing aid use; 5) adequate cognitive skills as determined by the Mini-Mental State Exam (Folstein, Folstein, & McHugh, 1979); 6) normal or corrected-normal vision by self-report; 7) manual dexterity adequate for writing and keyboard use by self-report; 8) no known life-threatening disease as determined by chart review (e.g., terminal cancer, hepatic encephalopathy, end stage cardiac, pulmonary or renal disease); and, 9) no known neurological or psychiatric deficiencies as determined by chart review.

The following measures are used: (1) MMSE (Folstein, Folstein, & McHugh, 1979);(2) subtests of the Wechsler Memory Scale - Revised (Wechsler, 1987); (3) the Mill Hill Vocabulary Test (Raven, 1965); (4) selective attention task (Stroop, 1935); (5) perceptual attention tasks (Salthouse, 1991); (6) Hearing Handicap Inventory for the Elderly (Ventry & Weinstein, 1982); (7) Abbreviated Profile of Hearing Aid Benefit (Cox & Alexander, 1995); (8) Audiologist Counseling Effectiveness Scale for the Elderly (Taylor, 1993); (9) Satisfaction with Amplification in Daily Living (Cox & Alexander, 1997); (10) self-report of hearing aid use; (11) Cognitive Styles Analysis (Riding, 1991); (12) self-report of cognitive styles; (13) prediction of task-memory self-efficacy; (14) subscales of the Metamemory in Adulthood questionnaire (Dixon & Hultsch, 1984); and, (15) a Hearing Aid Orientation Inventory (HAOI). These measures are administered in paper-and-pencil format with the exception of the CSA, which is a computer-presented test.

Findings: None to date. Data collection is nearly complete. A convenience sample of one-hundred three veterans have initiated the study with seven dropouts. Over fifty veterans have completed the study. It is expected that data collection will be finalized by mid-June. Hence, results will be available for discussion in August.

PB18

Hearing loss, hearing aids and depression in the elderly

James McCartney and Larry Meyers
California State University, Sacramento

Preliminary results will be reported on an ongoing study investigating hearing loss, hearing aids and depression in an independent living subgroup of long term care residents in the greater Sacramento area. The subjects comprise an aged (65+), independent and assisted living, long term care population who have received pure tone and speech (NU-6) tests in a quiet environment (<38 DBA), APHAB, HHIE, three standard screening tests of depression (CES-D, BDI-II and GDS) and a standardized diagnostic interview (DIS-IV). The Diagnostic Interview Schedule (DIS) was developed at the request of the National Institute of Mental Health. The goal was to have a reliable and valid instrument capable of being administered by trained lay individuals for an Epidemiological Catchment Area program which was initiated in the early 1980's. The DIS was developed at Washington University in the Department of Psychiatry and incorporates historical paper and pencil tests along with a formalized clinical interview using DSM-IV criteria in the latest version.

PB19

Measuring the value of hearing aid outcome

Harvey Abrams, VA Medical Center

Introduction: As a result of the growth in healthcare consumerism, the use of outcome data to demonstrate the efficacy of audiologic intervention has taken on increasing importance. Current clinical outcome measures (e.g. speech recognition testing, AI, APHAB, COSI, etc.) do not directly address the issue of “value”, i.e. are the benefits worth the cost? Among health economists, this question is often addressed by measuring an individual’s “willingness-to-pay (WTP)” for goods and services.Purpose: The purpose of this study was to measure the value placed on aspects of self-perceived hearing aid benefit utilizing a WTP approach.

Method: Questionnaires were mailed to 300 patients fit with hearing aids during a two-month period at a VA Medical Center. The questionnaires consisted of the Abbreviated Profile of Hearing Aid Benefit plus a question concerning household income and a question concerning how much the respondent would be willing to pay for each hearing aid, given the perceived benefits experienced with the hearing instruments.

Results & Preliminary Discussion: Step-wise multiple regression analyses were utilized to determine if hearing aid benefit was a significant predictor of the amount individuals would be willing-to-pay (WTP) for their hearing aids. Separate analyses were conducted for the APHAB Global Benefit score and for each of the four subscale scores: ease of communication (EC), reverberation (RV), background noise (BN), and aversiveness (AV). In addition to the benefit score, a second predictor variable, income category, was input into each regression model. Results revealed significant positive correlations between WTP and Global, EC, RV and BN benefit, with income group serving to increase the predictive strength of the regression model for the subscale scores. These findings indicated that within this population, as the amount of benefit increased in any of these domains, individuals were willing to pay more for their hearing aids. On average, individuals were willing to spend $22.35, $8.71, $8.03, and $6.85 for each point increase in Global, EC, RV, and BN benefit, respectively. These results suggest that a slightly greater value is associated with improving ease of communication than with improving listening in reverberant conditions. Further, both EC and RV benefit appear to have a greater value than that associated with improving listening in background noise. The relative values associated with improvements in listening conditions may have implications for hearing aid design and fitting.

PB20

The relation between audio-visual speech perception and reported benefit on the hearing aid performance inventory (HAPI)

Maureen Coughlin and Larry Humes
Indiana University

Evidence exists that hearing impaired listeners may rely more heavily on visual cues to aid in speech understanding than normal hearing listeners. Given that speech perception is enhanced when supplied with multi-modal cues, and that hearing aid benefit is directly correlated with improved speech understanding, it seems reasonable to assume that the ability to utilize multi-modal cues is a factor in hearing aid benefit. The purpose of this ongoing study is to determine whether the ability to utilize visual information effectively improves speech-understanding performance in noise and ultimately results in a higher rating of perceived benefit on the Hearing Aid Performance Inventory (HAPI) questionnaire.

Twenty elderly hearing-impaired subjects will participate in this project. The subjects were part of a larger study examining the factors of success associated with hearing aid use. All subjects had been fitted with the same hearing aids and therefore represent a sample of hearing aid users who have been wearing the same hearing aid circuitry for at least one year. Aided speech understanding in noise is being assessed using the CID Everyday sentences test at one presentation level (70 dB SPL, +2dB SNR) in three different conditions: audio only, visual only and audio-visual combined. Correlational analysis of speech recognition scores (CID Everyday Sentences test) and subjective reports of hearing aid benefit (HAPI) will be conducted. Results will be discussed in terms of benefit factors associated with hearing aid use and multi-modal speech perception.

PB21

The influence of measurement method on indicators of spatial hearing, speech understanding in noise and soft sounds hearing

Greg Flamme, University of Iowa

A disconcerting feature of current hearing aid research is the rarity with which promising new technologies are empirically shown to obtain better performance than their conventional counterparts. This presentation will describe a measurement

artifact that could obscure substantial treatment differences. Measurement theory shows that test scores contain trait, method, and unique/random error components. It is possible that the inability to observe significant treatment differences is due to a large method component in the outcome measures.
This presentation will describe the magnitude of trait, method, and error components in a group of nine audiometric measures, and the correlates of method factors. The psychophysical measures include pure tone thresholds, the Connected Speech Test, and a test of localization in noise. The self report measures include a modified version of the Hearing Screening Inventory, the Profile of Hearing Aid Benefit, and the Localization Abilities in Typical Environments questionnaires. The significant-other report measures were modified forms of the self report measures.

In this presentation, it will be noted that method components were negligible in some measures (e.g. accounting for less than 5% of the variance), yet quite high in others (e.g. accounting for 46% of the variance). Unique variance (i.e. the summed effects of measurement error and systematic variance associated neither with method nor the putative trait) accounted for between 7% and 54% of the variance in this group of measures. These results imply a need for re-consideration of how hearing abilities and hearing aid outcomes are assessed.

For example, typical estimates of effect size and statistical power are founded on classical test theory, which presumes that all systematic variance in a score is related to the trait. If a score is greatly impacted by measurement method, the minimum number of subjects required to detect a given treatment effect could be underestimated when using common equations. Consequently, important treatment effects could be missed if method effects are not controlled.

In summary, this presentation will describe the psychometric characteristics of nine measures of hearing and discuss the practical implications of non-trait variance in measures of hearing ability.

PB22

Total Hearing Health Care: Benefits from supplementing hearing aid fitting with group aural rehabilitation

Theresa Hnath-Chisolm, University of South Florida
Harvey Abrams, Barbara Mokotoff and Rachel McArdle, VA Medical Center, Florida

The cornerstone of comprehensive hearing health care management for individuals with adult-onset hearing loss is the use of well-designed and well-fit hearing aids. However, despite recent advances in hearing aid technology and fitting strategies there are still individuals whose quality-of-life remains negatively impacted as a result of their hearing losses. These individuals may benefit from participation in an aural rehabilitation (AR) program. In the current health care milieu it is important to demonstrate that the addition of such a program would result in greater improvements in quality-of-life than occurs through the use of hearing aids alone. Further, it would be important to demonstrate that any additional improvements from AR can be obtained at a reasonable cost. The purpose of this ongoing study is to compare the relative treatment efficacy and relative cost-effectiveness of hearing aid alone use (HA-alone) to that of hearing aid use combined with a short-term counseling-based group aural rehabilitation program (HA+AR).
At the conclusion of this study data will be available from 100 participants, both male and female, with acquired sensorineural hearing loss. Participants are fit with the most appropriate hearing aid technology and randomly assigned to one of two treatment groups: HA-alone or HA+AR. Numerous outcome data are being collected (e.g., MOS-36-Veteran's version; Hearing Handicap Inventory-short form (HHIE-S),

Communication Profile for the Hearing Impaired (CPHI), Communication Oriented Satisfaction Index (COSI), Speech Perception in Noise (SPIN), etc.) at four points in time: pre-intervention, immediately, 6-months, and 1-year post-intervention. Preliminary analysis of pre- to immediately post-intervention data for 40 participants indicates that the primary advantage to participation in this short-term group aural rehabilitation program is in the affective components of a person's acceptance and adjustment to the hearing loss. Methods derived from health economics to compare the relative cost-effectiveness of obtaining this differential treatment effect will be presented. [This work is supported by a VA Rehabilitation Research and Development Grant]

POSTER GROUP C

Saturday 8:00AM to 10:00PM

PC1

Perception of voiceless fricatives by normal-hearing and hearing impaired children and adults

Andrea Pittman, Patricia Stelmachowicz
Boys Town National Research Hospital

The perceptual strategies of young normal-hearing (NH) children and hearing-impaired (HI) adults have been found to differ from those of NH adults (Nittrouer & Crowther, 1998; Nittrouer, Crowther, & Miller, 1998; Doherty & Lutfi, 1996, 1999). These differences are likely due to a lack of experience with speech perception for the children and the presence of hearing loss for the adults. Hearing-impaired children present a unique problem in that hearing loss and immature perceptual skills may interact to produce perceptual weighting strategies that differ from all other groups.

This study examined the perceptual weighting strategies and performance-audibility functions of 11 moderate- to moderately-severe HI children, 11 age-matched NH children, 11 moderate to moderately-severe HI adults, and 11 NH adults. The purpose was to (a) determine the perceptual weighting strategies of HI children relative to the other groups, and (b) to determine the audibility required by each group to achieve a criterion level of performance. Stimuli were four nonsense syllables (/us/, /uS/, /uf/, and /uT/). The vowel, transition, and fricative segments of each nonsense syllable were identified along the temporal domain and each segment was randomly amplified within each syllable during presentation. Point-biserial correlation coefficients were calculated using the amplitude variation of each segment and the correct and incorrect responses for the corresponding syllable.

Results showed that for /us/ and /uS/, all four groups heavily weighted the fricative segments during perception whereas the vowel and transition segments received little or no weight. Performance-audibility functions were constructed for these segments and the mean audibility levels required to achieve > 70 % recognition were calculated for each group. Results showed that the HI children and adults required similarly low levels of audibility relative to the NH adults and children who required significantly higher levels. A decision theory approach, used to confirm the audibility criteria for each group and phoneme, yielded similar results. The present study revealed: (a) similar perceptual weighting functions for the HI children relative to the other groups; and (b) lower audibility requirements for the HI groups relative to the NH groups when perceiving the fricative segments of /us/ and /uS/.

PC2

Initial segment slowing effects in improving speech intelligibility for older adults

Satoshi Iwasaki, Hoshino Tomoyuki and Watanabe Takahiro, Hamamatsu University School of Medicine
Kondo Katsuhumi, Yamaha Corporation

The delayed processing ability of the central auditory system in older adults results in an increasing difficulty in understanding rapid speech. It is reported that elderly listeners had significantly better comprehensive performance at 140%-150% time-expansion than at the normal rate. Our hypothesis was that the slowing of initial part of a sentence would improve comprehension for older adults, because such adults have particular difficulty with initial encoding processes in part due to loss of auditory sensitivity. The purpose of this study was to investigate the effect of time expansion of the initial part (700ms in length) of a sentence composed of 5 or 6 syllables, on speech intelligibility for hearing-impaired older adults.

In experiment I, the entire sentence was expanded at one of three rates of speech ; fast (11 molar/s), normal (9 molar/s), or slow (6 molar/s). The effect of speech rate was then evaluated by scoring recall for each syllable in the sentence. In experiment II, a segment of the sentence (700ms in length) was expanded 150%, 170%, or 200%, and the pattern of expansion (A: initial 700ms expanded, B: initial 700ms expanded gradually, C: 700ms expanded from the 500ms mark) was varied.

The recall score at the slow speech rate was significantly ($p<0.0001$) high compared to the scores obtained under other speech conditions in experiment I. The score for the first syllable was the highest. Conversely, the average score for the second syllable was lowest at all the speech rates. Two patterns of expansion A and B, in which the initial 700ms of the sentence, the first and second syllables included, was expanded, improved performance in experiment II. Compared with A, B, where the rate of speech was changed in two steps, produced a more intelligible performance. In contrast, expansion of speech part way through a sentence (pattern C) was less effective.

The delay of time-expanded speech is a problem for the utility of hearing devices. Compared with the speech expansion systems reported previously, the length of speech expansion is small in our method. Therefore it may be possible that the initial segment slowing system is effective in improving speech intelligibility for hearing-impaired older adults as hearing device.

PC3

Speech recognition in noise at higher-than-normal levels: Decreases in scores and increases in masking

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

When signal-to-noise ratio is held constant, speech recognition in noise by normal-hearing and hearing-impaired listeners decreases as signal levels increase above those of conversational speech (e.g., Studebaker et al., 1999). There are several issues that remain unresolved with regard to the detrimental effects of high signal and noise levels on speech recognition. For example, the magnitude of the reduction in speech recognition at high levels varies substantially, perhaps due to differences in speech materials and/or masker spectra (e.g., Dubno et al., 2000). Also, the mechanism responsible for the deterioration of speech recognition at high levels is not yet known. To address these questions, word recognition was measured for normal-hearing subjects in nine conditions, corresponding to all combinations of three signal-to-noise ratios and three speech-shaped masker levels; pure-tone thresholds were also measured at each masker level. An additional noise was always present to equate subjects' threshold. Word recognition in speech-shaped maskers declined slightly but systematically as a function of speech level with the greatest change occurring in the most advantageous signal-to-noise ratio. When subjects' thresholds in the speech-shaped maskers were incorporated into predictions of word recognition using the articulation index, declines in word recognition with increasing speech level were accurately predicted. Thus, although the results confirmed that word recognition in speech-shaped maskers decreased at high levels when signal-to-noise ratio was held constant, deterioration in speech recognition was attributed to increasingly high thresholds in the speech-shaped maskers. [Work supported by NIH/NIDCD]

PC4

Integration Efficiency for Speech Understanding Within and Across Sensory Modalities

Ken Grant, Walter Reed Army Medical Center
Steven Greenberg, International Computer Science Institute, Berkeley

The ability to understand speech relies in part on our capacity to integrate spectro-temporal information from different frequency regions of the speech spectrum. This is especially true for multichannel hearing aids and cochlear implants where speech information is divided into separate spectral bands and subjected to different types and degrees of signal processing. The time frame over which this integration occurs may reflect different levels of processing: one which operates over relatively short time spans (ca. 50 ms) and is involved in detailed phonetic analysis of the signal and another operating at a more abstract level of syllable length units where the time span is about 250 ms. Because spoken conversation normally involves face-to-face interaction resulting in both auditory and visual information being used to decode the speech signal, it is important to determine whether the time frames for spectro-temporal integration and the efficiency at which integration proceeds is the same for auditory-only speech presentations and for auditory-visual speech presentations, and whether integration efficiency is compromised by hearing impairment. In this paper, we discuss the relative efficiency of these processes by comparing the ability of normal-hearing and hearing-impaired subjects to integrate narrow bands of speech when presented underauditory-only and auditory-visual conditions. Specifically, nonsense syllables (/a/-consonant-/a/) spoken by a female talker were filtered into two or four 1/3-octave wide bands of speech using an FIR filter whose slope exceeded 100 dB/octave. The four-band auditory condition consisted of filter passbands of 298-375 Hz, 750-945 Hz, 1890-2381 Hz, and 4762-6000 Hz presented concurrently. Two additional auditory conditions were made by combining either bands 1 and 4 (the two fringe bands) or bands 2 and 3 (the two middle bands). For auditory-visual conditions, subjects viewed a video image of the talker presented synchronously with either the two fringe bands or the two middle bands. A sixth condition consisting of visual-only speech recognition was also tested. Integration efficiency was determined by using Braida's Prelabeling Model of Integration [Quart. J. Exp. Psych., 43, 647-677 (1991)] to predict subject responses in the four-band auditory condition as well as for the two auditory-visual conditions. Comparisons of within-modality (auditory only) and across-modality (auditory-visual) integration efficiency will be discussed relative to optimal strategies for speech understanding. [This research was funded in part by the National Science Foundation]

PC5

Spectral integration across frequency for normal-hearing and hearing-impaired listeners

Jennifer Lentz, Marjorie Leek, and Laura Dreisbach, Walter Reed Army Medical Center

Distortions imposed by a hearing loss may affect a listener's ability to integrate spectral information, particularly when there is unequal loss across frequency. It is unclear whether the spectral shaping imposed by a hearing aid can fully compensate for such a loss in spectral integration. To evaluate whether sensorineural hearing loss affects across-frequency spectral integration, normal-hearing and hearing-impaired subjects were asked to discriminate between multi-tone complexes with differing spectral shapes. The stimuli were the sum of six tones spaced equally on a logarithmic frequency scale ranging from 200 to 3000 Hz. Listeners were asked to detect a spectral change in which half the components were increased in level and the other tones were decreased in level.

Thresholds were estimated for the amount of component level change necessary for detection, and the influence of different spectral regions (spectral weights) was estimated using a correlation technique as described by Richards and Zhu ("Relathan the equal-SPL condition. These results suggest that hearing-impaired listeners are able to make across-frequency comparisons in a manner similar to normal-hearing listeners. However, when the spectrum is shaped according to equal SL's across a sloping hearing loss, listeners may not be able to take full advantage of across-frequency comparisons. [Work supported by NIH (DC00626)]

PC6

Effects of Talker Variability and Lexical Difficulty on Spoken Word Recognition

Sumiko Takayanagi, Donald D. Dirks, Anahita Moshfegh, P. Douglas Noffsinger, and Stephen A. Fausti, UCLA School of Medicine

For most traditional word-recognition tests, phonetically or phonemically balanced word lists are produced by a single talker with only modest consideration for the phonological and lexical characteristics of the items. Evidence suggests however that word recognition depends on numerous talker-, listener-, and stimulus- related characteristics.

The purpose of this study was to examine the effects of lexical difficulty and talker variability on word recognition among four groups of listeners; native listeners with normal hearing or hearing impairment (moderate sensorineural hearing loss) and non-native listeners with normal hearing or hearing-impairment. Lexical difficulty was assessed by comparing word recognition performance between lexically "easy" and "hard" words based on frequency of occurrence in language and the structural characteristics of similarity neighborhoods formalized in the Neighborhood Activation Model (Luce, 1986; Luce & Pisoni, 1998). The ability of listeners to accommodate trial-to-trial variations in talkers (perceptual normalization) was assessed by comparing recognition scores for a single talker condition to those obtained in a multiple talker condition (words produced by a large number of talkers).

Seventy-five "easy" and 75 "hard" words were recorded by 10 talkers (5 males and 5 females) with voices whose fundamental frequencies ranged from 71 to 232 Hz. An up-down adaptive procedure was used to determine the sound pressure level for 50% correct performance. The experimental design consists of four variables: 2 within-subject variables - lexical difficulty and single vs. multiple talker conditions; and 2 between-subject variables - native vs. non-native status of English, and normal vs. hearing-impaired status of hearing.

The results demonstrated that the estimation of 50% performance was observed at lower (more sensitive) sound pressure levels for "easy" than "hard" words for each subject population. For both "easy" and "hard" words, non-native listeners with normal hearing required an average of ~4dB intensity for 50% performance than native normal hearing listeners. Non-native hearing-impaired listeners required an average of ~4dB greater intensity for 50% performance than hearing-loss matched native hearing-impaired subjects.

The current results suggest that among diverse groups of listeners (native and non-native listeners with normal hearing and hearing impairment), both the degree of acoustic-phonetic detail in the signal and the phonological and lexical nature of the stimulus materials ("easy" versus "hard" words) influence word recognition. In addition word recognition is also affected by the listeners prior experience with a talkers' voice (single vs. multiple talkers). Both lexical and instance-specific factors should be considered in the development of speech recognition tests. Funded by grants from the Department of Veterans Affairs Rehabilitation R&D

PC7

Factors effecting vowel discrimination and identification for listeners with normal and impaired hearing

Diane Kewley-Port, Indiana University

Vowel sounds, high in energy and proportionally longer than consonants, are highly salient portions of ordinary speech. Although it is generally recognized that vowels are perceived relatively well compared to consonants under a variety of listening conditions, the factors that effect discrimination and identification of vowels by hearing-impaired listeners are not well understood. This presentation will summarize recent research on abilities to discriminate and identify vowels over a wide range of experimental conditions. As a baseline, thresholds for discriminating a small change in vowel formants in isolation have been measured under optimal conditions. For young listeners with normal hearing (YNH), formant thresholds for synthetic vowels are best described as constant over formant frequency at 0.11 Barks. As uncertainty increases, such that one of eight vowel formants may be presented on any given trial, discrimination degrades by about a factor of two (0.24 Barks).

Another YNH group and a group of elderly hearing-impaired (EHI) listeners with mild-to-moderate sloping sensorineural hearing loss participated in both a discrimination and an identification experiment. At 70 dB SPL, the EHI listeners demonstrated comparable discrimination for F1 (0.22 Barks), but had a factor of 5.5 times worse performance over F2 (0.62 Barks). Identification of the synthetic vowels by EHI listeners was also impaired. This identification performance was correlated with both the reduced ability to discriminate vowels as well as with reduced audibility.

New research has examined how formant discrimination is degraded in more linguistically relevant stimuli that have longer phonetic context, including those identified as meaningful sentences. Increasing phonetic context from isolated vowels to syllables degrades discrimination performance, but additional phonetic context had reduced effects such that formant discrimination DLs in three-word phrases, nine-word sentences and sentences with the identification task were not statistically different. Thus a norm for formant discrimination in more ordinary sentences was estimated as 0.28 Barks. An analysis of the production of English vowels found closely spaced vowels to be about 0.56 Barks apart. That is, YNH listeners have a good match between production and perception in which they can discriminate vowel formants with a resolution two times smaller than formant differences in closely spaced vowels. This contrasts with the EHI listeners whose discrimination Dls for F2 exceeded the close-vowel formant differences. Performance by listeners on these synthetic vowels will be compared with those from nearly natural stimuli currently be tested. [Research supported by NIHDCD-02229.]

PC8

Low-pass-filtered word recognition in steady-state and interrupted maskers for adults with normal and impaired high-frequency hearing

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

High-frequency auditory nerve fibers respond to lower-frequency speech presented with sufficient intensity. Due to their high onset response synchrony, phase-locking, and coherence in phase, high-frequency fibers may be better at encoding low-frequency timing cues than low-frequency fibers. Thus, high-frequency hearing loss, in addition to limiting audible high-frequency speech information, may affect the encoding of lower-frequency speech cues. Also, because recovery from forward masking is more gradual at lower sensation levels and lower frequencies, individuals with high-frequency loss may be especially disadvantaged when listening to speech in temporally-varying maskers.

This study measures low-pass-filtered speech recognition in interrupted and steady-state maskers to assess the role of high-frequency fibers in processing temporal information in speech. Speech and maskers are selected and filtered to assure equal speech audibility for subjects with normal and impaired high-frequency hearing. Thus, any differences in speech recognition between subjects with normal and impaired high-frequency hearing will not be due to differences in speech audibility between groups, but may be consistent with the hypothesis that high-frequency fibers are useful in encoding lower-frequency speech. Here, subjects hear words presented in one of two spectrally-shaped-noise maskers: a) steady-state noise, and b) noise modulated by a 10 Hz square wave to create an interrupted noise. Both the speech and noise are presented at a higher and a lower intensity.

It is hypothesized that the importance of high-frequency fibers for understanding lower-frequency speech will be more pronounced for conditions that are affected by temporal resolution abilities, as is the case for interrupted maskers. Thus, the silent periods in interrupted maskers will benefit listeners with high-frequency hearing loss less than listeners with normal high-frequency hearing. In addition, it is hypothesized that at the lower speech level, contributions from the tails of high-frequency fibers will be reduced, resulting in smaller differences in speech recognition between listeners with normal and impaired high-frequency hearing. [Work supported by NIH/NIDCD]

PC9

Feature extraction in human speech recognition

Jont Allen, Miriam Furst, Larry Saul, and Mazin Rahim, AT&T Labs Research

In 1955 George Miller and Patricia Nicely explored the human speech code and discovered that a 5 channel feature code is used to extract phonemes from speech. Their \emph{ mutual information } analysis of the confusion matrices of filtered noisy speech showed that the 5 channels were virtually independent of each other, accounting for the robustness of speech communication quantified by Miller, Heise and Lichten in 1951. While the 1955 analysis was a major step forward, it did not lead to practical results, since there are no known methods for extracting these features from the speech. This weakness is the basis of the present research program, to \emph{define the signal processing required to robustly extract features from narrow bands of speech}. Our emphasis is to work in narrow bands, starting from critical bands. In this paper we discuss the first step of this process, which is to measure (a) the JND for the detection of speech in noise and (b) the JND for residue pitch given narrow bands of speech in noise. We have been investigating these features with the ultimate goal of trying to build a speech recognition machine based on distinctive feature signal processing.

Probably the most information bearing and robust of the Miller and Nicely features is ``voicing." It is no coincidence that pitch perception is widely recognized as contributing to ones identity as a speaker, providing the listener with the tags need to extract speech under adverse conditions. We have started by exploring the ability of normal hearing listeners to distinguish ``residue" pitch in narrow bands of noisy speech. Specifically we measured the threshold SNR for a speaker identification (pitch) task relative to the threshold SNR for a speech detection task. The method reduced the bandwidth and signal to noise ratio until the performance of the subject was 50% above chance level. Given these performance measures, our goal is to build an auditory model of detection of the speech band in background noise.

Our results show that using a 40 Hz speech band, the speakers can detect speech at an SNR of about 0 dB, and for 400 Hz bands, at -8 dB SNR. These results are, to first approximation, independent of talker, listener, the speech sample, and level. In the second experiment we looked at the discriminabilty of speakers for narrow band speech at adverse SNRs. In this case our preliminary results show that to detect residue pitch from a 0.75 oct band, centered at 1 kHz, the listener must hear the speech at a level approximately 10 dB greater than the speech detection threshold. We shall summarize our results on modeling these results. Robust feature extraction would be useful to hearing impaired listeners, and could supplement, or even replace, current hearing aids for the severely hearing impaired.

PC10

A statistical-decision-theory-based model for predicting speech intelligibility

Hannes Muesch, Northeastern University and GN ReSound, and Soren Buus, Northeastern University

This presentation introduces a new model for predicting speech intelligibility. The model, which we call the Speech-Recognition Sensitivity (SRS) Model, is based on Statistical Decision Theory. It aims to predict speech-recognition performance from the long-term average speech spectrum, the masking excitation in the listener's ear, the linguistic entropy of the speech material, and the number of response alternatives available to the listener. Unlike Articulation-Index (AI) models, the SRS model can account for synergetic and redundant interactions among spectral bands of speech. The SRS model also accounts for effects of linguistic entropy and number of response alternatives on intelligibility scores without resorting to the empirically determined ad hoc transformations employed by AI models. The effects of linguistic entropy are modeled by an entropy-dependent central noise, which modifies the listener's identification sensitivity to the speech. The effect of the number of response alternatives on the test score is a direct consequence of using Statistical Decision Theory. The SRS model also appears to predict how the effect of linguistic entropy varies with the filter condition and how linguistic entropy and language proficiency interact with signal-to-noise ratio.

The SRS model was tested by fitting it to data from the literature and to consonant-discrimination data collected in our laboratory. Normal listeners' ability to discriminate among 18 consonants was measured in 58 filter conditions using two test paradigms. In one paradigm, listeners chose among all 18 stimuli. In the other, response alternatives were restricted to the correct response and 8 consonants that were randomly selected from among the 17 incorrect response alternatives. Most filter conditions included one or more sharply filtered narrow bands of speech. Results show that, depending on the combination of bands used, listeners' performance in multi-band conditions falls short of, equals, or exceeds the performance expected from multiplication of the error rates in the individual bands. The performance advantage in multi-band conditions increases with the average band separation. The SRS model provides a good fit to the data for both response paradigms. The best-fitting model parameters are in good agreement with parameters used to fit data from the literature.

PC11

Effects of hearing loss and linear amplification on the relative importance of acoustic cues to vowel intelligibility

Sarah Hargus Ferguson, Diane Kewley-Port, and Larry E. Humes, Indiana University

The current study focused on the relationship between acoustic properties of vowels and their identification by elderly listeners with sensorineural hearing impairment, and the effects of hearing aid amplification on this relationship. To achieve speech tokens that varied naturally in intelligibility, a male audiologist read sentences under two speaking style conditions: (1) speaking as he would in everyday conversation (i.e., conversational speech) and (2) speaking as though he were talking to a hard- of-hearing person (i.e., clear speech). Monosyllabic words were excised from these sentences for presentation to nine elderly hearing-impaired subjects for vowel identification. Words were presented at 70 dB SPL in a background of 12-talker babble (signal/babble ratio = +3 dB) under two conditions, unaided and aided. In the aided condition, speech and babble were amplified with a response simulating the effects of a linear hearing aid with a frequency-gain characteristic prescribed by the Desired Sensation Level (DSL) formula. Vowel intelligibility was compared for the two speaking styles and also correlated with acoustic measurements to assess the relative importance of spectral, durational, and dynamic cues to vowel perception in both the unaided and aided conditions.

In contrast with previous research on the effect of clear speech, and also with data obtained from normal-hearing subjects on these materials (Ferguson & Kewley-Port, 1999), speaking style had no significant effect on overall vowel intelligibility for the hearing-impaired subjects. For many vowels, in fact, the conversational tokens were more intelligible than the clear tokens, in both the unaided and aided conditions. This suggests that some "clear speech" strategies may be detrimental for hearing-impaired listeners. In addition, the relationship between acoustic properties and vowel identification for the hearing-impaired subjects was different from that observed for normal-hearing subjects in the same task. This abnormal relationship was observed for both the unaided and aided conditions, suggesting that restoring signal audibility does not necessarily eliminate the effects of hearing loss on speech perception. [Supported by the American Speech-Language-Hearing Foundation, NIHDCD-02229, and NIHDCD-00012]

PC12

Using signal detection theory to evaluate differences between normal and impaired auditory performance on a frequency discrimination task

Lisa Huettel and Leslie Collins
Duke University

Although many individuals with hearing aids have relatively good speech recognition in quiet environments, their ability to understand speech in noisy environments degrades much more quickly than their normal-hearing counterparts. It has been hypothesized that this is partially a result of decreased sensitivity to sound in various regions of the frequency spectrum, and partially a result of degradation in frequency resolution capabilities. Standard signal processing strategies employed by analog hearing aids can only address sensitivity by applying simple amplification and compression strategies, and cannot adequately address frequency resolution and signal interaction. The advent of digital hearing aids provides an opportunity to implement more complex signal processing techniques in the next generation of hearing aids.

We are interested in determining which signal processing strategies will offer the greatest benefit to impaired listeners. The first step is to determine how various impairments affect the way acoustic signals are processed or encoded by the auditory system. Models provide a way in which impairments can be studied in isolation and compared to the behavior of a normal auditory system. In addition, by examining the responses at successive "stages" of the model, the type of information lost can be identified and localized. Finally, with a better understanding of what signal information is used by a normal auditory system but is lacking in an impaired system, signal processing techniques that take advantage of the remaining capabilities of an impaired system can be implemented in a digital hearing aid. These new designs can then be evaluated quickly, using models, to identify the most promising and to determine upper bounds on performance expectations.

We have investigated the differences between normal and impaired auditory processing for a frequency discrimination task by analyzing the responses of a computational auditory model using signal detection theory. Impairments were simulated by removing the compressive non-linearity and broadening the auditory filters. Just noticeable differences in frequency were determined both as a function of frequency and stimulus intensity. Two detectors, the temporally-based optimal processor and another, sub-optimal, processor based on spike count, were implemented. An analysis of the differences observed in normal and impaired model responses and performance will be provided along with some possible implications for hearing aid design.

PC13

Perception of sine wave modeled speech by hearing-impaired adults

Rupa Balachandran, Harry Levitt, Arlene Neuman, Patrick Nye and Maryrose McInerney
City University of New York

Sine wave modeling is a parametric method of modeling speech. In this process, the peaks in the speech spectrum are computed, selected and then speech is resynthesized using one sine wave for each peak in the speech spectrum. The number of sine waves used in the reconstruction can be varied. The current study is designed to determine the minimum number of sine waves at which there is no difference in the naturalness between sine wave and natural speech as perceived by hearing-impaired listeners. Subjects are required to listen to a natural token and sine wave processed tokens of VCV syllables. They are asked to pick the token that sounds more natural and then rate the difference between the two tokens. There are two different methods of peak selection under study. In the first method the speech spectrum is unmodified, In the second method the spectrum is boosted in the high frequency region to compensate for the roll off in the spectrum, before the peaks are selected. Listeners also compare the intelligibility of Sine wave tokens processed with and without spectral lift. All stimuli are filtered to meet NAL-R targets in a 6cc coupler, and subjects are tested at their most comfortable listening level. This is an on going study and preliminary results suggest that subjects are sensitive to the different methods of peak selection. For the first method there seems to be little or no difference to the perceived naturalness of speech as more sine waves are added to the signal. In the second method subjects are able to readily differentiate between natural and sine wave speech and require more sine waves to reach a point of no difference in naturalness between the two. Overall there is very little or no difference in the perceived naturalness after 20 sine waves. Also the sine waves processed without lifting the spectrum are judged to being more intelligible than their counterparts whose speech spectrums have been lifted.

PC14

The Symphonix Vibrant®Soundbridge - and alternative treatment modality for sensorineural hearing loss

Th. Lenarz, D. Gnadeberg, K. Ambjornsen,
Medical University of Hannover

During the last three years of clinical investigation the partially implantable hearing device Symphonix Vibrant soundbridge (VSB) has proven to be an alternative treatment modality for patients suffering from bilateral sensorineural hearing loss. Especially for patients having medical problems (e.g. otitis externa) with wearing an ear mold the soundbridge provides a valuable alternative to conventional hearing aids.

The system consists of two parts: the externally worn audio processor (AP) and the implanted vibrating ossicular prosthesis (VORP). The AP comprises a battery, a microphone, a signal processor, a magnet for retroauricular fixation and a transmitting coil. The VORP receives the electrical signal transcutaneously and vibrates the ossicular chain, using the so-called floating mass transducer (FMT). The transducer is coupled to the long process of the incus with a special crimper and without any changes to the structure of the ossicles. The coupling of the FMT to the incus is an important step during the surgery to guarantee maximum energy transfer and long-term stability.

The major advantage of this device as compared to conventional hearing aids is the ability to transfer frequencies up to 10 kHz and to leave the external ear canal open. An upgrade to implement new signal processing algorithm can easily be performed by exchanging the AP without the need for further surgery.Since February 1997, the Symphonix soundbridge has been implanted at the Medical University of Hannover in 34 patients, the first 19 during the clinical study.The benefit of the system for the patients was analysed by an extensive audiometric test battery. To determine the subjective contentment the soundbridge users were asked to complete questionnaires (e.g. APHAB).No complications, such as a deterioration of hearing or conductive hearing loss caused by the surgery or the mass loading of the transducer were observed in the long-term.The patients commended the natural sound quality, the lack of feedback, the absence of occlusion and distortion and the improved speech understanding in noise. The results of the Freiburger monosyllabic word test and the Goettinger sentence test in quiet and in noise (10dB S/N) improved with the soundbridge, especially after the upgrade from a two-channel analog to a three-channel fully digital audio processor.For detailed analysis the patients were subdivided into three groups according to their pre- and postoperative aided situation (monaural/binaural).First results of intraoperative and postoperative measurements to determine the quality of the coupling will be shown.

PC15

Versatile rehabilitation technique for DSP-driven cochlear implants: Speech-processing strategy using 'FIR' filter bank or 'FFT' algorithm

Ahmed Ben Hamida University of Sherbrooke

Cochlear implant is a hearing prosthesis dedicated to restore at least partially the hearing for profound or total deafness disability. In order to gain in flexibility and performance, recent design of these apparatus could be driven by a Digital Signal Processor 'DSP'. The use of a host 'DSP' enables to gain in handiness thanks to its programming tools. It gives, on the other hand, an opportunity to benefit of different options facilitating clinical trials and adjustments, particularly for achieving the optimal apparatus adaptation. Most of the conceived systems were based on research undertaken by several university groups, and were supported by different companies throughout the world.

Rehabilitation technique for deafness healthcare by cochlear implant system has been considered among its basic needs. Hence, in the present work, we try to offer a versatile rehabilitation technique offering more possibilities by using two complementary speech-processing
strategies, which could be implemented on the DSP-board of the prosthesis. These two numerical strategies were conceived around a temporal approach utilizing a filter bank model, which was based on Finite Impulse Response 'FIR' filters, and a spectral approach utilizing a Fast Fourier Transform 'FFT' algorithm. It is interesting to test these two strategies especially during apparatus adjustments, in the purpose of achieving the optimal adaptation.

By using the spectral strategy based on FFT, it was possible to evaluate sounds' spectrum with great flexibility. The sounds' spectrum would be necessary for determining several frequency bands that would be associated to the specific stimulation channels.. All parameters would be programmable thanks to a provided experimental environment permitting to adjust the apparatus relatively to patient comfort.

The temporal approach based on adjustable 'FIR' filter bank would be similar to the spectral approach since the designed FIR filters would be associated to stimulation channels, and with the same flexibility, especially in choosing filters' bands. After processing sounds by the external sound analyser, appropriate numerical data would be transmitted to the internal micro-stimulator through a communication link based on inductive coupling. The main functions assured during internal processing permitted to determine with great flexibility the stimulation current level to generate at each specified channel as well as stimulation rhythm.

PC16

Towards TICI (totally implantable cochlear implant) alternate powering and placement strategies for behind the ear (BTE) cochlear implant

Koushik K. Narayan
Sri Venkaeshwara College of Engineering, India

Cochlear Implants are recently being used for restoring the sense of hearing to profoundly deaf people. The goal of researchers in the recent times, is to make this system totally implantable. Once this is realized , it would lead to increased comfort for the implantees and would be cosmetically superior to non-TICI systems. Cochlear Ltd. have developed a Behind The Ear implant in which conventional belt worn processors are done away with and the implant system out side the ear is confined to the space between the pinna and the skull. Since these implants are powered totally using external means (using Zinc batteries), this system cannot be further miniaturized and thus TICI becomes an impossibility here.

This paper proposes alternate powering techniques for BTE so that it can be made totally implantable. The proposed system is powered by a chargeable photo-cell which is placed behind the skin present on the head region just behind the pinna.. This cell is charged by irradiating this region (behind the pinna just outside the skull) with low wavelength light. This is done as low wavelength light has greater penetrating power. The cell has to be placed as close to the surface as possible as the rate of transfer of power decreases as the square of the cell placement depth. Use of high energy radiation is limited here due to safety concerns. This cell can be designed to give 50-60 milliwatts which is the typical power consumption for the BTE implant. Due to practical reasons, power for the implant has to be supplemented by other means. The paper further proposes a Transcutaneous transformer, which can be fed through external means. This transformer steps down the power supplied and charges an adjacently placed behind the tympanic membrane (if it does not rupture during the surgery)or anywhere in the external auditory meatus. Thus the proposed system is a definite forward step towards TICI. Practical transformer charging/irradiation schemes are proposed. The paper further examines the viability of use of short bursts of high energy radiation for charging. A comparison between the conventional system and proposed system is presented.

PC17

Simulation of binaural intelligibility improvement through spectro-temporal manipulation of noise correlation

John Culling, Cardiff University

Competing voices are better understood when the voices emanate from different directions and are heard binaurally. An understanding of the underlying mechanism of this effect is vital to the appropriate design of binaural hearing aids. It is widely presumed that the binaural masking level difference (BMLD) acts as a simple laboratory model for this improved intelligibility. However large differences exist between the tasks of tone detection and of speech recognition; the latter requires a spectro-temporal pattern to be recovered and interpreted. Contemporary theories of the BMLD suggest that the relevant cue is a reduction in interaural correlation in the signal compared to the non-signal interval. These theories therefore imply that a spectro-temporal pattern of noise correlation could give the illusion of speech embedded in noise. The present experiment tested this inference. Speech was mixed with pink noise at a fixed adverse signal-to-noise ratio in the NoSo and NoSpi conditions. The correlation and intensity of the NoSpi mixture was measured as a function of time and frequency. Three further stereo stimuli were derived from this stimulus. A separate noise was manipulated in order to recreate the spectro-temporal patterns of (1) correlation, (2) intensity and (3) both correlation and intensity. Recognition of words encoded in this way was negligible in conditions (1) and (2), but significantly better in conditions NoSo, and (3). Condition NoSpi was substantially more intelligible than all others. The fact that recognition was better in condition (3) than in (1) or (2) suggests that the spectro-temporal pattern of correlation is a useful cue for speech recognition, but must be combined with monaural intensity information to produce measurable intelligibility. On the other hand, recognition performance in the NoSpi was not emulated in (3), so a complete account of performance in the NoSpi condition was not achieved.

PC18

Benefit of binaural hearing for speech intelligibility with multiple speech competitors

Monica Hawley, Ruth Litovsky, and H. Steven Colburn, Boston University

We are often required to listen to speech in environments that include other competing sources. A common complaint of hearing impaired listeners is the inability to understand speech in these "noisy" environments. Eight listeners with symmetric sensorineural hearing loss were tested in non-speech psychoacoustic tasks that measured their sensitivity to binaural information and in a speech intelligibility task with multiple speech competitors, in order to compare benefit of binaural information in the speech task to the results from the other psychoacoustical tasks. Five of the eight listeners (those with the largest losses) wore their hearing aids during the sound-field testing. The speech intelligibility task required subjects to repeat full sentences (HINT sentences: Nilsson et al., 1994) in the presence of one, two or three other sentences that were played from either the same loudspeaker as the target or from loudspeakers separated from the target loudspeaker. All loudspeakers were located in the fronl difference and speech intelligibility in complex environments, suggesting that for these listeners, the small advantage of separation was not simply an insensitivity to binaural information altogether. Low frequency hearing loss was associated with the inability to use binaural hearing in complex wideband tasks of masking level difference and speech intelligibility with competitors, but not with narrowband interaural difference tasks.

PC19

Talker separation and sequential stream segregation: Gender and ascending/descending paradigm effects

Carol Mackersie and Derek Stiles, San Diego State University
Tammy Prida, VA Medical Center, San Diego

The purpose of this study was to examine details of sequential stream segregation as predictors of the intelligibility of male and female speech in the presence of a competing talker of opposite gender.
Eleven adults with sensorineural hearing loss repeated pairs of sentences spoken simultaneously by a man and a woman. Performance was measured in terms of the percentage of words correct. Sequential stream segregation was measured using the method described by Rose and Moore (1997) in which a fixed 1000 Hz tone [Tone A] was alternated with a tone whose frequency varied [Tone B]. The tone sequences were presented in an ABA_ pattern. The varying tone B changed in either an ascending or descending pattern starting at a frequency either lower than (100-200 Hz) or higher than (2000-3000 Hz) the fixed tone. The frequency difference between the tone A and B was decreased, until listeners indicated they could no longer perceptually separate the two tones (fusion threshold).
Based on group data, there was no significant difference between fusion thresholds for the ascending and descending series. Analysis of individual listener data, however, showed significant differences between ascending and descending fusion thresholds for 7/11 of the subjects. Fusion threshold as measured in the "ascending" task was found to predict intelligibility of the male talker, but not the female talker. Conversely, fusion threshold measured using the "descending" task was found to predict intelligibility of the female talker, but not the male talker. In both cases, higher speech perception scores were associated with lower (better) fusion thresholds.

These finding are consistent with the idea that the dynamic process involved in streaming might also underlie listeners' ability to use intonation contours to facilitate the perceptual separation of two simultaneous talkers. In addition, findings suggest that the importance of streaming in the perceptual separation of talkers depends on the nature of the information provided by the changing pitch stream.

PC20

Selective listening by using a pair of microphones and the interaural signal-envelope correlation

Kiyoaki Terada, Masumi Tamura, and Mikio Toyama, Kogakuin University, Hachioji-shi, Japan

Selective listening is an important function for binaural hearing. If we can implement a binaural auditory function which includes selective listening in hearing aids, signal quality should be much improved in subjective terms, particularly in conditions with a lot of reverberation or noise. The precedence effect is one of the most important binaural properties for perception of sound in a reverberant space. The human auditory system is easily able to focus its attention on the direct sound in such a situation by inhibiting reverberant sounds. A two-point microphone system has been used to investigate models of binaural selective listening by many researchers. Most of the processing schemes are based on interaural cross-correlation: however, it turns out that the precedence effect can not be described well by using interaural correlation.

We have investigated a selective listening system which is based on the interaural signal-envelope correlation, which is closely related to the precedence effect. The signal envelope is an important cue for binaural properties including the precedence effect. The precedence effect depends on the envelope of the signal over time rather than on the power spectrum of the signal. We thus introduce the interaural correlation of the signal envelope instead of correlation of the signals in themselves, so that the precedence effect can be taken into account. Speech can be enhanced by the selective listening system in the presence of strong reverberation by following these procedures: 1) sub-band filter the signals, 2) detect the sub-band signal envelope and generate a pulse train that represents the signal envelope, 3) calculate the short term envelope-pulse cross-correlation between every pair of sub-band signals, 4) determine the weighting function that corresponds to the envelope correlation, 5)synthesize full-band signals through overlap-frame processing.
The precedence effect occurs clearly for speech signals, although for noise signals the effect is not marked. The correlation between the two-point

microphone signals estimated by using the signal envelope pulse-trains is able to nicely represent this difference in the precedence effect, in terms of the inhibition of initial echoes. The weighting function obtained in our selective listening system can thus be extended to include the precedence effect. We will discuss the effect of changing the signal processing parameters and location of the pair of microphones on the quality of the speech signal. We would be able to obtain almost the same speech enhancement effect by using a pair of gradient-type microphones at a single point in the listening space, instead of a pair of microphones separated by some distance. This selective listening system has two input signals and one output signal. This research can therefore be applied to the development of hearing aids that include a selective listening property.

PC21

Effects of relative amplitude and fundamental frequency differences in the perceptual separation of competing voices in listeners with hearing loss

Kathryn Arehart and Julie Swensson, University of Colorado at Boulder

Even when wearing hearing aids, listeners with hearing loss have difficulty understanding one talker in the presence of a competing talker. The listener must perceptually separate the parts of the acoustic signal that belong to the talker from the parts of the acoustic signal that belong to the competing talker. A better understanding of the mechanisms underlying this difficulty in listeners with hearing will contribute to our understanding of the limitations of current amplification systems as well as suggest strategies for their improvement.

When normal-hearing listeners are asked to identify a target vowel in the presence of a competing vowel, their ability to identify the target improves significantly when the two vowels have different fundamental frequencies. This effect is greatest when the target vowel is lower in level than the competing vowel, supporting the idea that the target is perceptually separated by cancellation based on the harmonicity of the competing vowel (deCheveigne, 1997, J. Acoust. Soc. Am. 101, 2839-2847).

Previous research in our laboratory showed that listeners with hearing loss are not able to use differences in fundamental frequency in the perception of competing voices as well as normal-hearing listeners and that the overall ability to understand simultaneous speech is significantly reduced. Our results suggested the hypothesis that listeners with hearing loss do not benefit from fundamental frequency differences at low target-to-background ratios and so may not be able to carry out the harmonic cancellation process.

We compared the ability of five normal-hearing listeners and eight listeners with cochlear hearing loss to identify a target vowel in the presence of a competing vowel when the double vowels had the same fundamental frequency (0 semitones) or different fundamental frequencies (2 semitones). The amplitudes of the vowels were also manipulated to create five different target-to-background levels (-10, -5, 0, +5, +10 dB). Listeners with cochlear hearing loss showed the same benefit from fundamental frequency differences at low target-to-background ratios as listeners with normal hearing. This result suggests that listeners with cochlear hearing loss are able to carry out the spectro-temporal processing necessary to separate the periodicities of competing vowel sounds. This allows them to both determine the multiplicity of sound sources when there are fundamental frequency differences as well as to give them significant improvement in their ability to correctly identify target vowels. However, the overall correct identification rates of the listeners with cochlear hearing loss are substantially lower than the rates in listeners with normal hearing across all conditions. This suggests that the representation of the vowels in the peripheral auditory system of listeners with hearing loss are degraded relative to the representations of vowels in the auditory system on normal-hearing listeners.

Conference Attendees

Name	Affiliation
Harvey Abrams	Audiology & Speech Pathology Service
Jayne Ahlstrom	Medical University of South Carolina
Jose I. Alcantara	University of Cambridge
Silvia Allegro	Phonak AG
Jont Allen	AT&T Labs Research
Abeer Alwan	University of California Los Angeles
Ture Andersen	University Hospital of Odense
David Anderson	Georgia Tech
Kathryn Hoberg Arehart	University of Colorado
Thomas Baer	University of Cambridge
Carlos Baez	Decibel Instruments, Inc.
Matthew Bakke	Lexington Center for the Deaf
Rupa Balachandran	City University of New York
Sharba Bandyopadhyay	Center for Hearing and Balance
Lucille Beck	Department of Veterans Affairs
Ahmed Ben Hamida	University of Sherbrook
Ruth Bentler	University of Iowa
Kumiko Boike	University of Washington
Arthur Boothroyd	City University of New York
Lars Bramslow	Oticon A/S
David Brown	Gennum Corporation.
Ian Bruce	Johns Hopkins University
Sam Burchfield	University of Tennessee
Pamela Burton	Seimens Hearing Instruments
Soren Buus	Northeastern University
Juan Carlos Tejero Calado	
Caron Caraway Grant	
Laurel Carney	Boston University
Teresa Ching	National Acoustic Laboratories
Theresa Hnath Chisolm	University of South Florida
Thomas Ulrich Christiansen	Technical University of Denmark
Leslie Collins	Duke University
Chris Conger	Resistance Technology Inc.
Kelly Consoer	Medtronic
Mary Cord	Army Audiology & Speech Center
Leonard Cornelisse	dspfactory, Ltd
Maureen Coughlin	Indiana University
Robyn Cox	University of Memphis
Brian Csermak	Gennum Corporation
Huanping Dai	GN ReSound
Alex Darbut	Resistance Technology Inc.
Chris Darwin	University of Sussex
Harvey Dillon	NAL
Amy Donahue	NIDCD/NIH
Wouter Dreschler	Academic Medical Center
Judy R. Dubno	Medical University of South Carolina
Ole Dyrlund	GN ReSound
Brent Edwards	RxSound
Claus Elberling	Oticon Research Centre

Conference Attendees

Name	Affiliation
Almudena Eustaquio Martin	Universidad de Castilla-La Mancha
Xiaoling Fang	SONIC Innovations
Stephen Fausti	VAMC Portland Oregon
Brenda Fedor	SONIC Innovations
Sarah Hargus Ferguson	Indiana University
Michael Fisher	National Acoustic Laboratories
Gregory Flamme	University of Iowa
Mary Florentine	Northeastern University
Anna Forsline	GN ReSound
Todd Fortune	Telex Communications
Bas Franck	Academic Medical Centre
Dan Freed	House Ear Institute
Bob Fretz	Resistance Technology Inc.
John Fryer	Department of Veterans Affairs
Miriam Furst Yust	AT&T Research Labs
Shawn Gao	House Ear Institute
Stuart Gatehouse	MRC Institute of Hearing Research
Gary Gibian	Planning Systems Inc.
Julius L. Goldstein	Washington University
Stephen Gonzenbach	VA New York Harbor Healthcare System
Ken W. Grant	Walter Reed Army Medical Center
Melanie Gregan	Starkey Laboratories
Gustav F. Haas	Auditory Rehabilitation Engineering
Martin Hansen	Widex ApS
Frances Harris	University of Arizona
Jorg Haubold	GEERS Hörakustik
Monica Hawley	MIT
Simon Haykin	McMaster University
Mark Hedrick	University of Tennessee
Michael Heinz	Hearing Research Center
Johan Hellgren	Linköping University
August Hernandez	Seimens Hearing Instruments, Inc.
Maria Ho	Texas Instruments
Volker Hohmann	Universitaet Oldenburg
Inga Holube	Siemens Audiologsiche Technik GmbH
Ben Hornsby	Vanderbilt University
Amy Horwitz	Medical University of South Carolina
Tomoyuki Hoshino	Mamamatsu University School of Medicine
Zezhang Hou	Audia Technology
Lisa Huettel	Duke University
Karen Hull	Hearing Research Center
Martina Huss	University of Cambridge
Frank Iglehart	Clarke School for the Deaf
Tatuya Ito	RION Co. Ltd.
Satoshi Iwasaki	Hamamatsu University School of Medicine
William Johnson	Threepenny Electronics Corporation
Douglas Jones	University of Illinois at Urbana-Champaign
Craig Jordan	NIH/NIDCD/DER
Jagmeet Kanwal	Georgetown University

Conference Attendees

Name	Affiliation
Min Chi Kao	National Chiao Tung
James M. Kates	AudioLogic
Michiko Kazama	Kogakuin University
Gitte Keidser	National Acoustic Laboratories
Michael Kenworthy	Technology Commercialization Corp.
Diane Kewley Port	Indiana University
Mead Killion	Etymotic Research
Hee Nam Kim	Starkey Korie Ltd.
David Kirkwood	The Hearing Journal
Virginia Kitch	University of Washington
Francis Kuk	Widex Hearing Aid Company
Akiko Kusumoto	Oregon Graduate Institute
Chi M. Lam	National Acoustic Laboratories
Matthias Latzel	INTERTON Hoergeraete GmbH
Jos Leenen	Beltone
Arne Leijon	KTH-TMH
Jennifer Lentz	Army Audiology & Speech Center
M. Charles Liberman	Eaton-Peabody Laboratory
Michael Lockwood	University of Illinois
Enrique Lopez Poveda	Facultad de Medicina, Edificio Benjamin Palencia
Carl Ludvigsen	Widex ApS
Lynn Luethke	NIDCD/NIH
Thomas Lunner	Technical Audiology
Fa Long Luo	GN ReSound
Carol MacKersie	San Diego State University
Gil Magilen	Hearing Centers Network
Neeraj Magotra	Texas Instruments
Alhassane Maiga	Starkey Laboratories
Josephine Marriage	University of Cambridge
Norm Matzen	Etymotic Research
Rachel McArdle	Bay Pines VA Medical Center
James McCartney	California State University, Sacramento
Hugh McDermott	The University of Melbourne
Chandler McDowell	SONIC Innovations
Nick Michael	GN ReSound Corporation
Ronald Miles	SUNY Binghamtom, Dept. of Mechanical Engineering
Joong Seok Moon	House Ear Institute / USC
Brian Moore	University of Cambridge
Ludwig M. Moser	Bayerische Juliu-maximilians Universität
Hannes Muesch	GN ReSound Corporation
Kevin Munro	University of Southampton
Melanie Nakisa	Royal National Institute for Deaf People
Koushik K. Narayan	Sri Venkateswara College of Engineering
Graham Naylor	Oticon Research Centre
Anna Nebelek	University of Tennessee
Igor Nebelek	University of Tennessee
Victor Nedzelnitsky	National Institute of Standards and Technology
John Nelson	Widex
Peggy Nelson	University of Minnesota

Conference Attendees

Name	Affiliation
Arlene Neuman	Graduate Center, CUNY
Cheryl Neylon	GN ReSound
Dr. Donald Nielsen	Central Institute for the Deaf
Suzanne Nielsen	Spouse of Dr. Donald Nielsen
Michael Nilsson	SONIC Innovations
Hiroaki Nomura	University of East Asia
Peter Nordquist	Department of Speech Music and Hearing
Manette Nousak	Nousak Audiology & Hearing Aid Center, Inc.
Regina Nuzzo	Stanford University Statistics Department
Anna O'Brien	Bernafon AG
Cherish Oberzut	GN ReSound Corporation
Miho Ohsaki	Shizuoka University
Bob Oliveira	Hearing Components
Laurel Olson	GN ReSound Corporation
Jane Opie	InSonus Medical, Inc.
Chaslav Pavlovic	GN ReSound
Karen Payton	University of Massachusetts at Dartmouth
Mette Boel Pedersen	Oticon
Shawn Perman	GN ReSound
Andrea Pittman	Boys Town National Research Hospital
Angela Pool	Siemens Hearing Instruments, Inc.
David Pool	Decibel Instruments, Inc.
J. Kristian Poulsen	Microtronic
Thomas Powers	Siemens Hearing Instruments, Inc.
Jill Preminger	University of Louisville, School of Medicine
David Preves	Songbird Hearing, Inc.
Sunil Puria	RxSound Corp / Stanford
Tina Purtee	California State University, Northridge
Christine Rankovic	Articulation, Inc.
Vishakha Rawool	Bloomsburg University, Dept. of Audiology & Speech Pathology
Judith Reese	University of Florida, Department of Veterans Affairs
Carol Rogin	Hearing Industries Association
Tove Rosenbom	Oticon
Peninah Rosengard	HST – Division of Harvard University and MIT
Ayaskanta Rout	Purdue University
Jay Rubinstein	University of Iowa
Melissa Ruscetta	University of Pittsburgh
Janet Rutledge	National Science Foundation
B M Ryals	James Madison University
Allen Ryan	UCSD/SOM – Surgery-Otolaryngology
Carl Sandrock	Phonic Ear, Inc.
Marion Schabert	Telecommunications Institute I, University Erlangen-Nuremberg
Arthur Schaub	Bernafon AG
Jon Schmidt	Starkey Laboratories
Ahmad Shamsoddini	GN ReSound Corporation
Bob Shannon	House Ear Institute
Don Shaver	Gennum Corporation
Scott Shaw	Planning Systems, Inc.
Adnan Shennib	InSonus Medical, Inc.

Conference Attendees

Name	Affiliation
Gae Won Shim	Starkey Korea, Ltd.
Sang Don Shim	Starkey Korea, Ltd.
Helen Simon	Smith-Kettlewell Eye Research Institute
Leslie Smith	Stirling Hearing Systems/Stirling University
Sig Soli	House Ear Institute
Pamela Souza	University of Washington
Paul Spaanderman	Microtronic
Patricia Stelmachowicz	Boys Town National Research Hospital
David Sterratt	Stirling Hearing Systems/Stirling University
Michael Stone	University of Cambridge
Ann Strouse	VA Medical Center
Christopher J. Struck	GN ReSound Corporation
Joanna Swiers	Microtronic
Hideyuki Takagi	Kyushu Institute of Design
Nubuo Takahashi	Ehime University
Yoshinori Takahashi	Kogakuin University
Sumiko Takayanagi	UCLA School of Medicine
Jagdish Tandon	
Kiyoaki Terada	Kogakuin University
Linda Thibodeau	University of Texas at Dallas
Steve Thompson	Knowles Electronics, Inc.
Emily Tobey	University of Texas at Dallas
Tim Trine	Starkey Laboratories, Inc.
Danielle Tryanski	Starkey Laboratories, Inc.
Chris Turner	University of Iowa
Rosalie Uchanski	Central Institute for the Deaf
Aart Van Halteren	Microtronic
Dianne VanTasell	Starkey Laboratories, Inc.
Therese Velde	Siemens Hearing Instruments, Inc.
Andy Vermiglio	House Ear Institute
Deborah Vickers	University of Cambridge, Department of Experimental Psychology
Thomas Victorian	Starkey Laboratories, Inc.
Marcel Vlaming	Beltone Netherlands
David Wayne	Desert Microtechnology Associates, Inc.
Theo Wesendahl	auric Hörsysteme GmbH & Co. KG
Soren Westermann	Widex
Nathaniel Whitmal	Worcester Polytechnic Institute, ECE Department
William Woods	Starkey Laboratories, Inc.
Thomas Worrall	AudioLogic, Inc.
Min Xu	Medtronic, Inc.
Hirofumi Yanagawa	Chiba Institute of Technology
Jun Yang	GN ReSound Corporation
Jerry Yanz	Micro-Tech Hearing Instruments, Inc.
Eric Young	Johns Hopkins University
William Yund	VA Medical Center
Hans Zwart	Auric Hearing Systems