IHCON 2002

INTERNATIONAL HEARING AID RESEARCH CONFERENCE 2002

AUGUST 21 - 25, 2002

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### Scholarship Recipients

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<td>Sharba Bandyopadhyay</td>
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<td>Cynthia Compton</td>
<td>Gaullaudet University</td>
<td>Audiology</td>
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<td>Michael Epstein</td>
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<td>Rachael Frush</td>
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<td>Anita Haravon</td>
<td>City University of New York</td>
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<td>Michael Heinz</td>
<td>The Johns Hopkins University</td>
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<td>Eric Hemmeter</td>
<td>Washington University, St. Louis</td>
<td>Electrical Engineering</td>
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<td>Paula Henry</td>
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<td>Masato Hishitani</td>
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<td>Martina Huss</td>
<td>University of Cambridge</td>
<td>Experimental Psychology</td>
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<td>Lorienne Jenstad</td>
<td>University of Washington</td>
<td>Audiology</td>
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<td>J. Brandon Laflen</td>
<td>Purdue University</td>
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<td>Dawna Lewis</td>
<td>University of Nebraska</td>
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<td>Sheng Liu</td>
<td>University of California, Irvine</td>
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<td>Michael C. Orr</td>
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<td>Pritesh Pandya</td>
<td>University of Texas, Dallas</td>
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<td>Melissa Ruscetta</td>
<td>University of Pittsburgh</td>
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<td>Susan Scollie</td>
<td>University of Western Ontario</td>
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<td>Jennifer Sowards</td>
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<td>Thomas Stainsby</td>
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<td>Chin-Tuan Tan</td>
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<td>Rebecca Warner</td>
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<td>Audiology</td>
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<td>Jason White</td>
<td>Washington University, St. Louis</td>
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Daily Schedule

Wednesday, August 21

5:00 PM          Welcome Social
6:00 PM          Dinner
7:30 PM          Welcome Remarks
7:45 PM          Keynote Address
8:45 PM          Discussion
9:00 PM          Evening Social

Thursday, Friday & Saturday, August 22-24

7:00 AM          Breakfast
8:00 AM          Morning Session A
9:45 AM          Poster Session
11:10 AM         Morning Session B
12:20 PM         Lunch
5:15 PM          Evening Session
7:00 PM          Dinner
8:20 PM          Social/Poster Session continues

Sunday, August 25

7:00 AM          Breakfast and Checkout
8:00 AM          Morning Session
9:45 AM          Break
10:05 AM         Morning Session continues
12:00 PM         Adjournment (buses leave for airport with box lunches for passengers)
PROGRAM SUMMARY

WEDNESDAY, AUGUST 21
WELCOME AND KEYNOTE ADDRESS
7:30PM-8:45PM

Welcome Remarks: Sig Soli
Pat Stelmachowicz

KEYNOTE ADDRESS

H. Steven Colburn Making use of multiple microphones in hearing aids and cochlear implants

THURSDAY, AUGUST 22
SESSION ONE
8:00AM-9:45AM

AUDITORY PHYSIOLOGY AND HEARING AIDS
Moderator: Eric Young

Brenda Ryals The functional significance of hair cell regeneration on auditory perception and vocal production

Monica Hawley Coding of sound in the human central auditory system: Spatiotemporal patterns of fMRI activity

Ian Bruce Physiological modeling for hearing aid design

POSTER SESSION A 9:45AM – 11:00AM
THURSDAY, AUGUST 22
SESSION TWO
11:10AM-12:20PM
GROWTH OF RESPONSE AND LOUDNESS
Moderator: Jont Allen

Michael Heinz  Activity growth rates in auditory-nerve fibers following noise-induced hearing loss
Mary Florentine  Cochlear hearing loss reduces the dynamic range for loudness: Implications for hearing aids

SESSION THREE
5:00PM-7:00PM
SIGNAL PROCESSING: COMPRESSION
Moderator: Robert Morley

Karolina Smeds  Is normalization of overall loudness an appropriate goal for hearing aid prescription?
James Kates  Advances in digital dynamic-range compression
Peter Blamey  Adaptive dynamic range optimization (ADRO)
FRIDAY, AUGUST 23
SESSION FOUR
8:00AM-9:45AM
CLINICAL ISSUES
Moderator: Lucille Beck

Gitte Keidser
What amplification characteristics do hearing impaired listeners want?

Richard Seewald
Hearing aid selection procedures for children: Report of a collaborative study

Eric A. Durant
Hearing aid fitting with a genetic algorithm

POSTER SESSION B 9:45AM – 11:00AM

SESSION FIVE
11:10AM-12:20PM
CLINICAL ISSUES
Moderator: Lucille Beck

Catherine Palmer
Functional and physiological change subsequent to hearing aid use

Mead Killion
Hearing-aid fidelity ratings, 25-band accuracy scores and compression characteristics

SESSION SIX
5:00PM-7:00PM
OUTCOME MEASURES
Moderator: Robyn Cox

Larry Humes
The secrets of hearing-aid “success” in elderly adults

Mary Pat Moeller
Developmental indices of hearing aid benefit: Insights from a longitudinal study
Stuart Gatehouse  Beyond segmental intelligibility – celebrating the complexity of the auditory world

SATURDAY, AUGUST 24

SESSION SEVEN
8:00AM-9:45AM

SPEECH PERCEPTION

Moderator: Dianne Van Tasell

Judy R. Dubno  Listening in the dips: Masking of speech by interrupted noise and detection of tones in forward masking

Jont Allen  Speech intelligibility and future hearing aids

Henrik L. Olsen  Fast or slow compression for optimal speech recognition in fluctuating noise for normal and hearing impaired listeners

POSTER SESSION A  9:45AM – 11:00AM

SESSION EIGHT
11:10PM-12:20PM

DEAD REGIONS IN THE COCHLEA

Moderator: Dianne Van Tasell

Brian C.J. Moore  Effects of low-pass filtering on the intelligibility of speech in noise for people with and without dead regions at high frequencies

William Woods,  “Dead regions”: Test results and predictions from cochlear models
SESSION NINE
5:00PM-7:00PM

SIGNAL, PROCESSING

Moderator: Harvey Dillon

Stefan Launer  Digital signal processing for "real-life" hearing instruments

Björn Hagerman  A method to measure the effects of noise reduction algorithms using simultaneous speech and noise

Michael A. Stone  Perceptual constraints to hearing aid processing algorithms

SUNDAY, AUGUST 25
SESSION TEN
8:00AM-9:45AM

DIRECTIONAL MICROPHONES

Moderator: Sig Soli

Arlene Neuman  A newly developed in-situ measurement system: The directional hearing aid analyzer

Todd Ricketts  Frequency effects and directional hearing aid benefit

Wouter A. Dreschler  The effectiveness of adaptive directionality by dual microphones in a digital hearing aid

BREAK 9:45 AM - 10:05 AM
SESSION ELEVEN
10:05AM-12:00NOON

ALTERNATIVE THERAPIES FOR SENSORINEURAL HEARING LOSS

Moderator: Jay Rubenstein

Christopher Turner  Combining electric and acoustic hearing for severe high-frequency hearing loss

Patricia Leake  Plasticity in the developing auditory system induced by electrical stimulation with a cochlear implant

Fan Gang Zeng  Treatment of auditory neuropathy: cochlear implants or hearing aids?

ADJOURNMENT  @

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Making use of multiple microphones in hearing aids and cochlear implants

H. Steven Colburn and Nathaniel J. Durlach, Hearing Research Center and Department of Biomedical Engineering, Boston University, USA

Since spatially distributed sensors often provide significant advantages in analyzing the environment, it is not surprising that such sensor arrangements are often found in both natural and artificial sensory systems. In general, these advantages arise in two ways. First, multiple sensor locations provide multiple "looks" at the environment and enable the receiver to choose a look that is relatively optimal for a particular purpose. Second, the signals received at the different locations can be combined using appropriate processing to give a "synthetic look" that is superior to that which can be achieved with any single sensor. In principle, hearing aids and cochlear implants can exploit both of these paths to provide the impaired listener with superior performance in such tasks as auditory localization and the perception of target sources in backgrounds of interference. However, the extent to which these advantages can be realized depends not only on the availability of multiple microphones, but also on the availability of adequate processing of the microphone signals, either internal (i.e., in the listener's auditory system) or external (i.e., in the electronic processing of the microphone outputs prior to stimulation of the listener's auditory system).

In this talk, we review our state of knowledge, and comment on needed research, in the area of multimicrophone hearing aids and cochlear implants. Although a variety of configurations will be considered, attention will be focused exclusively on acoustical sensing, auditory stimulation, and the use of no more that two microphones (with the microphones located at the ears). No attention will be given to visual aids or to the use of more than two microphones and array processing of the outputs of these microphones (although such systems are becoming increasingly practical as the appropriate technologies are becoming available). The case of single-microphone input with unilateral stimulation is the normal monaural hearing aid case and provides the natural reference for the bilateral cases. The cases with left and right microphones, with either one or two ears stimulated, are the primary cases of interest. In contrast to the case in which both ears are stimulated (the conventional bilateral case), in the case in which two microphones are used for sensing but only one ear is used for stimulation, additional external processing (a "smart" processor) is considered.

The primary conclusion of this review is that research and development is needed in at least four areas: (1) investigation of basic abilities of listeners to combine inputs from left and right stimulation; (2) development of prosthetics designed to exploit residual binaural abilities with particular attention paid to the coherence of the stimulus waveforms to the right and left sides; (3) development of processing schemes that provide processing of the bilateral inputs for unilateral stimulation; and (4) comparative tests of systems of different types. [Work supported by NIDCD.]
SESSION ONE

AUDITORY PHYSIOLOGY AND HEARING AIDS

Moderator: Eric Young

8:00 AM  THE FUNCTIONAL SIGNIFICANCE OF HAIR CELL REGENERATION ON AUDITORY PERCEPTION AND VOCAL PRODUCTION
Brenda Ryals, James Madison University and Robert J. Dooling
University of Maryland

As far as is currently known, birds provide the only animal model where it is possible to restore hearing sensitivity through renewed sensory cell regeneration. Fortuitously, birds also offer a unique animal model of vocal learning and production. Thus we have the singular opportunity of examining the influence of a "new" auditory periphery on complex auditory perception as well as vocal learning and production in a mature vertebrate. The question of whether a "new" auditory periphery results in sufficient functional recovery that the animal can perceive, learn, and produce complex acoustic communication signals has enormous health relevance as research efforts focus on finding ways to trigger hair cell regeneration in the mammalian auditory system. Understanding the fine details of hearing recovery in birds may also tell us something about how the bird ear functions and, at the same time, add to our knowledge of plasticity in both peripheral and central auditory nervous system structures and the nature of sensorimotor interfaces. Developing an animal model to track the time course of auditory and vocal recovery may have particular significance for the effective use of auditory prosthetic devices, such as cochlear implants, for the severely hearing impaired (Tyler, 1993).

The purpose of this presentation will be to review a set of experiments we have performed over the past 5 years to determine the influence of hair cell regeneration on complex auditory perception and vocal production. All of the experiments involved behavioral conditioning in a small Australian parrot, the budgerigar. First, we tested whether, following hair cell loss and regeneration, a bird could discriminate among, and reliably classify, complex vocal communication signals. Absolute thresholds, intensity difference limens (IDLs), frequency difference limens (FDLs), the discrimination between natural and synthetic contact calls, and the identification of contact calls were all measured before and after injection of kanamycin in adult budgerigars. In the second set experiments we examined the effect of hair cell loss and regeneration on vocal production. Budgerigars were trained by operant conditioning to produce different contact calls to two different LEDs before and after injections of kanamycin.

Results show that both auditory perception and vocal production are disrupted when hair cells are damaged or lost but that these behaviors return to near normal over time. Precision in vocal production recovered completely, well before recovery of full auditory function. Thus, at least in birds, even limited recovery of auditory input soon after deafening can support full recovery of auditory perception and vocal precision.
8:35 AM  Coding of sound in the human central auditory system: spatiotemporal patterns of fMRI activity
Monica Hawley, University of Maryland School of Medicine and Jennifer Melcher, Massachusetts Eye and Ear Infirmary, Harvard Medical School

We are examining how fundamental sound features (e.g., intensity, temporal envelope, frequency, bandwidth) are coded in the spatial and temporal patterns of population neural activity in the human central auditory system. Traditional microelectrode recordings capture the fine time-pattern of neural spiking in individual neurons, but do not necessarily provide a good assay of spatio-temporal coding in neural populations. In contrast, functional magnetic resonance imaging (fMRI), the technique chosen for our studies, provides an indicator of population activity (over a spatial scale of millimeters and a time-scale of seconds). And, unlike microelectrode recordings, it can be applied routinely to human listeners. In this talk, we will review how sound is represented in the spatio-temporal patterns of fMRI activity in humans. We will illustrate how the activity patterns are transformed across structures from cochlear nucleus through auditory cortex, interpret the patterns in terms of neural response properties, and relate the activity patterns to sound perception. [Supported by NIDCD, American Tinnitus Association, Tinnitus Research Consortium]

9:10 AM  Physiological modeling for hearing aid design
Ian Bruce, Eric Young, and Murray Sachs
Johns Hopkins University

Physiological data from hearing-impaired cats suggest 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 reduces confounding responses (Miller et al., JASA 106:2693, 1999). A physiologically accurate model of the normal and impaired auditory periphery would provide simpler and quicker testing of such potential hearing aid designs. Details of such a model, based on that of Zhang et al. (JASA 109:648, 2001), will be presented. Model predictions suggest that impairment of both outer and inner hair cells contribute to the degraded representation of vowels in hearing-impaired cats. The model is currently being used to develop and test a generalization of the Miller et al. speech-processing algorithm described above to running speech.
[Supported by NIDCD grants DC00109 and DC00023]
SESSION TWO

GROWTH OF RESPONSE AND LOUDNESS

Moderator: Jont Allen

11:10 AM Activity growth rates in auditory nerve fibers following noise induced hearing loss

Michael Heinz, Murray Sachs and Eric Young
Johns Hopkins University

Loudness recruitment imposes a significant constraint on the design of hearing-aid algorithms, however, the physiological correlates of loudness and recruitment are still not well understood. Physiological experiments were designed to characterize the growth of auditory-nerve (AN) activity with sound level for a range of stimuli that are commonly used in psychophysical studies of loudness. The present study directly tests the hypothesis that AN-fiber rate-level functions should be steeper in an impaired ear.

Rate-level functions were measured in 1-dB steps for various stimuli in AN fibers with best frequencies (BFs) up to 10 kHz. Data were collected from both normal-hearing cats and cats in which a narrowband noise centered at 2 kHz was used to produce high-frequency hearing loss. The stimuli included BF tones, 1- and 2-kHz tones (frequencies near the corner of and within the hearing loss), broadband noise, and a brief speech token (\textit{besh}).

The shapes of rate-level functions for BF tones demonstrated the standard diversity as a function of threshold for normal-hearing cats, i.e., low-threshold fibers showed sharp saturation while higher-threshold fibers typically showed sloping saturation. Two-line fits to each sloping-saturation rate-level function characterized the low-level and high-level slopes, while single-line fits were made for sharply saturating responses. In general, normal-hearing response growth showed variations across stimuli and BF that can be accounted for by basilar-membrane compression and suppression effects associated with normal outer-hair-cell (OHC) function.

Response growth in many impaired AN fibers was quite similar across the various stimuli, in contrast to normal response growth, and these fibers rarely demonstrated sloping saturation. These AN fibers typically had broad tuning curves that lacked a well-defined tip region, as expected with OHC loss. In contrast, some impaired AN fibers showed much shallower slopes to BF tones than normal fibers. These fibers tended to have tuning curves that included a well-defined tip, presumably resulting from mixed IHC and OHC damage. Overall, the slopes of rate-level functions for impaired AN fibers were not consistently steeper than the low-level slopes for normal AN fibers. The similarity of low-level slopes in normal and impaired rate-level functions is inconsistent with the idea that loudness recruitment represents a steeper growth of loudness near threshold. However, this physiological result is consistent with recent psychophysical evidence (Buus and
Florentine, 2002) suggesting that the rate of loudness growth near threshold is similar in normal and impaired listeners. [Supported by NIDCD].

11:45 AM COCHLEAR HEARING LOSS REDUCES THE DYNAMIC RANGE FOR LOUDESS: IMPLICATIONS FOR HEARING AIDS
Mary Florentine and Seren Buus
Northeastern University

A commonly held belief in audiology and hearing science for the past 60 years is that loudness grows more rapidly than normal near the elevated thresholds of listeners with cochlear hearing losses. This belief is based on the untested assumption that loudness at threshold is the same in normal listeners and in listeners with cochlear hearing losses. This assumption—together with the common observation that intense sounds usually have near-normal loudness in listeners with cochlear hearing losses—has led to the classic concept of recruitment.

Recent evidence against this pervasive notion comes from innovative psychoacoustical experiments, which use loudness matches between a tone and four- or ten-tone complexes to obtain highly reliable measurements of the rate of loudness growth near threshold [Buus, Mitsch & Florentine, J. Acoust. Soc. Am. 104, 399-410, 1998]. Results for listeners with hearing losses of primarily cochlear origin reveal a normal rate of loudness growth near threshold [Buus and Florentine, J. Assoc. Res. Otolaryngol., 3, 120-139, 2001], which agrees with recent knowledge about basilar-membrane mechanics. These data also indicate that loudness at threshold is greater than normal when threshold is elevated by a cochlear hearing loss. In other words, listeners with cochlear hearing losses do not show recruitment in the sense of an abnormally rapid growth of loudness near threshold. Rather, they have softness imperception, because they cannot hear some low loudnesses that are audible to normal listeners. This means that listeners with cochlear hearing losses not only have a reduced dynamic range of audible sound levels, they also have a reduced dynamic range of loudness. This new understanding of loudness perception in listeners with cochlear hearing losses has at least two clear implications for the design of hearing aids. First, complete restoration of normal loudness in people with cochlear hearing losses is probably impossible. Second, to make loudness perception as normal as possible in listeners with cochlear hearing losses, one must use a compression ratio less than that needed to map the normal dynamic range into their reduced range of audible SPLs. These implications are consistent with clinical experience and lead to a new understanding of consumer complaints. Sounds that are easily tolerated by normal listeners may be annoyingly loud if they are amplified to be audible for a listener with cochlear hearing loss. This is what many patients have been trying to tell us for years.
[Supported by NIH/NIDCD grant R01DC02241.]
IS NORMALIZATION OF OVERALL LOUDNESS AN APPROPRIATE GOAL FOR HEARING AID PRESCRIPTION?

Karolina Smeds, KTH Royal Institute of Technology, Sweden, Gitte Keidser and Harvey Dillon, National Acoustics Laboratories, Australia, Justin Zakis, University of Melbourne, Frances Grant and Christopher Brew, National Acoustics Laboratories, Australia

The study investigates a rationale underlying most prescriptive methods for nonlinear, WDR, hearing aids. Prescriptions are typically based on some sort of loudness normalization. Most methods are based on the assumption that loudness should be normalized at each frequency (loudness density normalization), and hence that overall loudness should be normalized. The NAL-NL1 method, which has speech intelligibility maximization as the basic underlying rationale, aims at normal overall loudness, though does not normalize loudness at each frequency. Maybe people don't actually prefer normal loudness.

The study consists of two parts, one laboratory test and one field trial. Normal-hearing subjects and hearing-impaired subjects (both with and without hearing aid experience) participate. In the laboratory test, subjects watch and listen to video recordings representing listening situations that vary in presentation level and in type (both speech and non-speech situations). For the hearing-impaired subjects the sound files are filtered according to the NAL-NL1 fitting method, which for most situations gives normal overall loudness. The subjects rate loudness and interest for the eleven listening situations and adjust the presentation level to their preferred loudness. Preliminary results indicate that neither the normal hearing nor the hearing-impaired subjects prefer normal loudness. They generally prefer less than normal loudness, especially in high level situations and in uninteresting (typically non-speech) situations.

In order to study this in the subject's everyday listening situations, a field trial is carried out. The subjects are fitted with a digital research hearing aid. For the normal-hearing subjects, the hearing aid is used as a linear hearing aid, and the insertion gain set to 0 dB across frequency. For the hearing-impaired subjects, the hearing aid works as a slow-acting WDR hearing aid with compression in three channels, and the insertion gain is set according to NAL-NL1. The hearing aid has a volume control, and the subjects' task is to adjust the volume control to preferred loudness in everyday listening. After the adjustment, the subject presses a button on the hearing aid, which makes the hearing aid log acoustical information about the listening situation, calculated loudness, and volume control setting. We are currently collecting data for the field study, and the results.
relating input signal statistics, preferred gain, and calculated loudness will be presented at the conference.

If hearing-impaired subjects' preferences for overall loudness on average deviate from normal, the finding will have consequences for the design of prescriptive methods for hearing aids.

5:50 PM

ADVANCES IN DIGITAL DYNAMIC RANGE COMPRESSION
James M. Kates
Cirrus Logic / AudioLogic (USA)

Multi-channel dynamic-range compression is an important function in modern hearing aids. One objective in designing a digital compressor is to match the frequency resolution of the digital system to the resolution of the human auditory system. The frequency-analysis tools commonly used in digital signal processing, such as the discrete Fourier transform (DFT), provide constant-bandwidth frequency resolution. The frequency resolution of the human auditory system, however, is more accurately modeled by a filter bank having a frequency-dependent bandwidth; the auditory bandwidth is nearly constant at low frequencies but becomes proportional to frequency as the frequency increases.

An approximation to auditory frequency analysis is provided in most multi-channel compression systems. In a compression system having many channels, a fast Fourier transform (FFT) is often used to provide the frequency analysis. Because of the uniform frequency spacing of the FFT, a large transform size is needed to provide adequate frequency resolution at low frequencies; the resolution is then finer than that needed at higher frequencies, so FFT bins are combined to give analysis bands with a resolution closer to that of the ear. The large transform size, however, has an undesired side effect in that the compressor will have an excessive processed signal delay. Further problems can result from the limitations on the number of processing operations per second available in the digital hearing aid. Compromises made to get the processing to run on the hearing-aid DSP can lead to implementations having excessive non-linear signal distortion, and can also result in large amounts of ripple in the compressor frequency response.

In this presentation, digital frequency warping is used as the basis for a new compressor design. Digital frequency warping replaces the uniform digital frequency scale with a frequency scale that is stretched at low frequencies and squeezed at high frequencies to more closely match auditory perception, thus eliminating the frequency-resolution conflict inherent in FFT systems. The new compressor gives frequency resolution on an auditory critical band frequency scale, and reduces signal delay, frequency-response ripple, and non-linear signal distortion. In this presentation, the new compressor algorithm is described and compared to approaches currently in use. Frequency-response, signal delay, and distortion measurements are then presented to illustrate the advantages of the new compressor. Processed sound files will also be presented.
[Work performed under contract to GN ReSound]
ADAPTIVE DYNAMIC RANGE OPTIMIZATION (ADRO)

Peter Blamey, Lois Martin, Christopher James and David MacFarlane
Cooperative Research Centre for Cochlear Implant and Hearing Aid Innovation, Australia

The adaptive part of ADRO continuously monitors the output levels of a sound processor in multiple narrow frequency bands, and slowly adjusts the gain in each band to keep its output level within the input dynamic range for the following stage of processing. In the case of hearing aids and cochlear implants, the following stage of processing is the listener's auditory system, and the dynamic range is defined by the threshold and maximum comfortable level for each frequency band. Optimization occurs independently in each band according to an audibility rule which ensures that sounds can be heard, and a comfort rule that ensures that sounds are not too loud. A third rule which limits the loudness of background noise in each band has also been investigated. Flexible fitting software allows the audiologist to measure in-situ thresholds and comfort levels, and to adjust audibility, comfort, and noise targets for individual listeners.

The differences between ADRO, linear gain, and multiple broadband compression schemes with various fitting prescriptions will be illustrated by detailed measurements of the long-term spectrum of the output signals relative to listeners' thresholds, maximum comfortable levels, and discomfort levels.

Experimental studies have shown significantly improved speech perception in quiet and in noise for adults and children using ADRO with cochlear implants, compared to the same processor without ADRO, especially at low input levels. The ADRO processor is also reported to be more comfortable and preferred over the non-ADRO processor in patient questionnaires. A laboratory study of hearing aid users has produced compatible results comparing ADRO to a linear hearing aid fit with frequency-specific maximum power output limiting. A field trial comparing ADRO to a 3-channel compression scheme using the NAL-NL1 fitting rule in a commercially-available hearing aid is in progress. The preliminary results will be reported at the conference.
WHAT AMPLIFICATION CHARACTERISTICS DO HEARING IMPAIRED LISTENERS WANT?

Gitte Kedser, Frances Grant, Christopher Brew, Harvey Dillon and Scott Brewer, National Acoustics Laboratories, Australia

For many years, hearing aids did nothing but amplify sounds based on a single gain-frequency response. At that time, the focus of clinicians and researchers was to find the amplification characteristic that would work best for the hearing aid user in a wide range of listening environments. A typical recommendation was a response prescribed from threshold or supra-threshold levels that made speech audible and comfortable across a wide range of frequencies. It was also found that hearing aid users could benefit from relatively more gain in the high than in the low frequencies to avoid upward spread of masking from the stronger low frequency components of speech and some background noises. Such recommendations were found to compensate well for the individual hearing loss in general listening situations. However, with the introduction of more sophisticated features in hearing aids, such as multiple channels, compression, multiple memories and speech/noise detectors, it is now possible to change the gain-frequency response at independent frequencies and for independent input levels, and depending on the acoustic input. This development has resulted in the introduction of new prescriptions and device-specific recommendations that largely ignore the findings of early research and that in some instances vary greatly in outcomes. Further, there is little evaluation data available for the new recommendations though it seems to be a simple question to ask what amplifications hearing impaired listeners want?

In the past decade, several laboratory and field tests have been conducted at the National Acoustic Laboratories (NAL) that each provides some information about the hearing aid user's choice of amplification in a wide range of listening environments. This talk combines this empirical data to demonstrate that 1) between a loudness normalization and a speech intelligibility maximization rationale, the latter is preferred as a base response for general listening. In particular when the two rationales result in very different targets; 2) simple linear and non-linear gain changes from a fitted base response can compensate for changes in the acoustic input; 3) more gain is preferred for understanding speech in noise than to reduce the annoyance of background noise; and 4) in a given situation the inverse of the preferred compression ratio linearly increases as the compression threshold decreases. That is, the hearing impaired listener selects to have a given range of input levels presented within the same (narrower) range of output levels.
HEARING AID SELECTION PROCEDURES FOR CHILDREN: REPORT OF A COLLABORATIVE STUDY

Richard Seewald, National Centre for Audiology, Canada, Teresa Ching and Harvey Dillon, National Acoustic Laboratories, Australia, Jane Joyce, National Centre for Audiology, Canada, Louise Britton and Susan Scollie, National Acoustic Laboratories, Australia

It is widely accepted that infants and children who have hearing loss should be fitted with amplification as soon as possible. In recent years, several hearing aid selection procedures have been proposed for the fitting of modern non-linear hearing instruments in children. To date, there is no conclusive evidence to support the universal application of one of these procedures over the alternatives. Further, it can be assumed that even the best of the existing procedures will require further development as we continue to learn more about this important clinical problem. As proponents of the two most widely used prescriptive methods in children (DSL[i/o] and NAL-NL1), the National Acoustic Laboratories of Australia and the National Centre for Audiology in Canada are collaborating to investigate the amplification requirements of infants and children.

This presentation will report the findings of the first in a series of collaborative studies on pediatric hearing instrument fitting procedures.

The purpose of this study was to compare performance and the preferences of children who were fitted with the DSL[i/o] and NAL-NL1 procedures. Twenty-four children at each site (48 in total), aged between 6 and 18 years, with hearing losses ranging from mild to severe, served as subjects. Each child was fitted binaurally with new digital dual-channel, dual-memory, wide dynamic range compression instruments. The measures used included multiple-level speech perception tests in quiet and in noise, paired comparison tests, loudness ratings, parental and classroom teacher observations and ratings, and the opinions and ratings of the children. Two eight-week home trial periods were employed during which the children wore their hearing aids set to each of the two prescriptions. Assignment of prescriptions was counterbalanced across the 48 subjects. In this way, half of the children were assigned to NAL-NL1 for the first trial and half to DSL[i/o] first. During a third home trial both prescriptions were programmed into the hearing aids and the children were permitted to freely switch between the two programs. All trials were double-blind with one experimenter at each site setting the hearing aids and the second experimenter administering the testing and interviews. The children, parents and teachers were all blind to the hearing aid settings on trial. The results on all measures will be presented and discussed along with a description of our plans for future collaborative projects.

[This work was funded by the Oticon Foundation]
HEARING AID FITTING WITH A GENETIC ALGORITHM

Eric A. Durant, Milwaukee School of engineering, Gregory Wakefield, University of Michigan, Dianne VanTassel and Martin Rickert, Starkey Laboratories, Inc.

The signal processing parameters of today's complex hearing aids are usually set via prescriptive fitting algorithms accessible through the hearing aid manufacturer's fitting software. Once the initial setting of the parameters has been accomplished, however, there is usually a second phase wherein the parameters must be further adjusted to approximate more closely the user's preferred settings as the user becomes accustomed to wearing the hearing aid. Because there are so many adjustable signal processing parameters, and because their effects co-vary, the simultaneous readjustment of all necessary parameters is a complex multidimensional task that is, at best, tedious, error-prone, and time-consuming. An automated, efficient, and effective means for searching a multidimensional hearing aid parameter space is therefore required.

Research on parameter search procedures for fitting hearing aids has typically used mathematical optimization theory, adapting search approaches such as the simplex method (Neuman et al., 1987). As the number and complexity of hearing aid parameters increase, the shortcomings of these methods, such as becoming stuck at local optima and not being able to combine attributes from multiple settings, become burdensome. The genetic algorithm (GA) is a search procedure that borrows many concepts from biology, including natural selection and genetic crossover and mutation. The GA maintains a population of solutions (hearing aid parameter sets) and repeatedly replaces the least fit solutions with the offspring of better performing solutions. In the work reported here, the GA was applied to hearing aid fitting via development of an efficient procedure to determine the relative quality of solutions in the population by repeatedly asking the patient to select the better of two alternatives.

Two experiments were conducted with eight normal hearing and eight hearing-impaired subjects. In the first, three parameters were varied to control a normalized least mean squares (NLMS) filter for cancellation of acoustic feedback. In the second, six parameters were varied to fit slow-acting three-band dynamic range expansion. In both experiments, the GA was running in a portable experimental hearing aid platform that was configured to enable subjects to make paired comparisons of hearing aid parameter sets in their normal acoustic environments.

In the feedback cancellation experiment, all subjects converged efficiently to parameter sets that produced optimal feedback cancellation. A follow-up experiment showed that subjects' initial preferences varied little upon retesting. The results for the expansion system were also positive, but highlighted some problems and suggested changes that might be made to improve performance when fitting more complicated parameter sets.
11:10 AM  **FUNCTIONAL AND PHYSIOLOGICAL CHANGE SUBSEQUENT TO HEARING AID USE**  
Catherine Palmer, University of Pittsburgh, Shelley Myer, Veterans’ Administration Medical Center, John Durrant, University of Pittsburgh, Charles Nelson, Veterans’ Administration Medical Center, and Melissa Ruscetta, University of Pittsburgh

The purpose of the investigation was to quantify the extent and rate of auditory perceptual learning subsequent to hearing aid use across conditions of quiet and noise in a carefully defined group of subjects. Degree of hearing loss, subsequent return of audibility, age of subjects, signal processing schemes, and amount of hearing aid use during the study were controlled. Both experimental groups (trained and untrained on the specific task) revealed a significant improvement in percent correct on a nonsense syllable task over time as compared to the control group. The trained group revealed more improvement than the untrained group for high frequency sounds in both quiet and noise (both conditions were considered inaudible prior to amplification) and for low frequencies in noise (this condition may have been defined as audible for many of the subjects prior to amplification). In the condition that was defined as audible prior to amplification (low frequency sounds in quiet) and therefore no change was expected, there was no difference in change over time between the untrained and trained subjects. There was no significant correlation between hours of use and gain in percent correct over time. The training paradigm will be described and future research related to training tasks will be outlined. Currently, we are measuring the mismatched negativity (MMN) associated with these tasks in order to define a potential physiological correlate to the functional learning that was found in the first part of the experiment.

11:45 AM  **HEARING-AID FIDELITY RATINGS, 25-BAND ACCURACY SCORES AND COMPRESSION CHARACTERISTICS**  
Mead Killion, David Preves, Ronald Scicluna, and Patricia Niquette, Etymotic Research, Inc.

Since the 1970’s, Consumer’s Union has used a 21-band accuracy score based on Steven’s Mark 6 Loudness Tables, to rate high fidelity loudspeakers. In 1979, Killion showed excellent correlation between subjective ratings of three groups of judges and calculated 25-band accuracy scores. In experimental recordings of available digital hearing aids on KEMAR using live jazz and classical groups, a recent digital hearing design using electronic damping and CORFIG equalization rated highest in both fidelity ratings and dollar value ratings by several groups of subjects. Some of the other digital aids, whose accuracy rating was only 20 points lower, were rated 40 to 50 points lower on subjective
judgments. Measures of distortion (clipping), Speech Transmission Index (for pumping), and Perceptual Analysis Measurement System (quality) were obtained to help explain these results.

Friday, August 25

Session Six

Outcome Measures

Moderator: Robyn Cox

5:15 PM

The Secrets of Hearing Aid “Success” in Elderly Adults

Larry Humes, Nathan Amos and Dana Wilson
Indiana University

How is a successful hearing-aid outcome determined? What outcome measures should be used and when should they be obtained? What factors are associated with individual differences in hearing-aid “success”?

These and other questions will be addressed in this presentation through a review of recent research completed in our laboratory. A multidimensional model of hearing-aid outcome measures will be presented and the outcome measures associated with each dimension identified. Factors or variables associated with individual differences in the performance of elderly hearing-aid wearers along each dimension will also be identified. It is hoped that identification of such factors will ultimately lead to the optimization of performance along each dimension and, ultimately, to greater hearing-aid “success” for the hearing-impaired elderly, both individually and collectively. [This work was supported, in part, by NIA.]

5:50 PM

Developmental Indices of Hearing Aid Benefit: Insights from a Longitudinal Study

Mary Pat Moeller
Boys Town National Research Hospital

It is well established that infants with mild to moderately-severe hearing loss have the potential to benefit significantly from amplification. It is less clear how to select the best hearing aid characteristics for infants and how to determine if a selected fitting yields optimal benefit for language learning. The fitting process would be enhanced by: a) better information on the specific impact of hearing loss on infant learning and b) identification of markers signaling hearing-related development that is “off course.”

These issues are currently being investigated in a longitudinal study designed to explore the role of auditory experience in children’s early language development. This presentation will include a description of the longitudinal study and tools used to track developmental outcomes in typical and hard of hearing infants. Early phonological and
lexical-grammatical development of these two groups of infants will be compared. Case examples will be used to illustrate some specific influences of audibility on early development. Preliminary results suggest the potential for developmental indices of hearing aid benefit. Implications for fitting hearing aids in young infants and the need for focused developmental measures will be considered. [This word was supported by NIDCD]

6:25 PM

BEYOND SEGMENTAL INTELLIGIBILITY — CELEBRATING THE COMPLEXITY OF THE AUDITORY WORLD

Stuart Gatehouse

MRC Institute of Hearing Research (Scottish Section), Royal Infirmary, United Kingdom

We hear sounds around us all the time, deriving from multiple sound sources at multiple locations occurring at varying points in time. When we hear a sound which has salience, we shift our attention, eyes and head towards the source, and we listen carefully. We comprehend the sound, and often participate in communication, principally in the form of dialogue. The auditory systems, and deficits in its function, are integral to the cascade between hearing, listening, comprehending and communicating. Traditional audiological research pays little attention to the complexities of human communication. In elderly listeners with SNHL, for the perception of the segmental intelligibility of a single talker in a background of steady-state noise presented monaurally over headphones, individuals’ measures of the audibility of the speech-signal (perhaps augmented by measures of frequency and temporal resolution) exert a high degree of predictive leverage. In contrast, for listeners in real rooms with a variety of reverberation characteristics, containing multiple sound sources (some of which are talkers and some of which are non-speech) which listeners are required to locate, attend to, and switch attention between, then such measures are much less predictive. Despite the fact that the vast majority of SNHL is primarily cochlear in origin, the interaction between sensory and cognitive aspects of hearing must exert a material influence on the extent to which listeners function in real environments on perceptually relevant tasks, and hence on disability and on the benefits of intervention. Performance measures in the laboratory or clinic usually test segmental intelligibility of a single voice, whose spatial position and spectral/temporal characteristics are static and predictable, in a single noise (usually steady state or at best speech-like babble), which is again static and predictable. In the self-report domain we access a richer set of communication environments, though still predominantly ignoring the three-dimensional and temporally dynamic aspects of the auditory world. This communication reports from a series of experiments which represent the start of a research program to investigate auditory abilities, disabilities and the benefits of hearing aids in more complex acoustical environments whilst undertaking perceptually relevant tasks. They include:

1) The development of a self-report scale, the Speech hearing, Spatial hearing and Quality of hearing (SSQ) which attempts to tap into elements of disability including selective attention, switching attention, appreciation of distance and movement, and aspects of identification and sound source segregation. The data show systematic relationships with impairment and likely difficulty of the circumstances. Strikingly, they show the highest correlations with an independent measure of hearing handicap for items which tap into switching and sustaining attention, and aspects of distance and movement (that is the perceptionally, spatially and temporally dynamic aspects of
listening) rather than the traditional disability items. The correlations survive partialling with age, hearing level and auditory lifestyle and are not evident in elderly listeners without sensorineural hearing loss.

2) A performance measure of selective attention based upon a combined speech and music task confirms that the rank ordering of reports on the SSQ with measured ability to sustain attention. Thus listeners report of attentional abilities appear to reflect a processing capability rather than some external intervening influence.

3) Repeat administration of the SSQ after experience of (predominately) linear amplification shows both systematic benefits and disadvantages on individual SSQ items.

4) A combined test of localization and speech identification ability in noise for wide-band and low-pass speech stimuli in diffuse noise background and noise backgrounds where the noise has a single perceived location (equivalent to noise with low and high interaural correlation) demonstrate significant effects of both binaural hearing and auditory attention on speech intelligibility, even in a relatively simple segmental task.

The results provide an initial framework to justify the potential importance of auditory abilities that are achieved beyond the level of the cochlear in both attentional and spatially and temporarily dynamic aspects to the problems that hearing impaired people experience in everyday listening and the extent as to which the provision of amplification might elevate those disabilities. They point to some potential pitfalls in amplification strategies which might compromise real world listening ability. A challenge for the future is to gain an understanding based on evidence in both self-report and performance domains of the effects of sensorineural hearing loss and the benefits of differing amplification rationales.
LISTENING IN THE DIPS: MASKING OF SPEECH BY INTERRUPTED NOISE AND DETECTION OF TONES IN FORWARD MASKING

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

When listening to speech in interrupted or modulated noise, audible speech information during momentary improvements in signal-to-noise ratio provides substantial increases in speech recognition relative to performance in steady-state noise. However, individuals with hearing loss, even those with relatively mild loss, benefit less from these improvements, a finding that may be explained in two ways. First, elevated thresholds limit the improvement in signal-to-noise ratio that may be realized when the noise level momentarily decreases. Second, the benefit derived from modulated or interrupted noise may be related to the recovery from forward masking, i.e., the recovery of a response to a supra-threshold signal from prior stimulation by a masker. Further, recovery from forward masking may be related to subgroups of auditory neurons with different thresholds. In addition to recovering more slowly relative to high spontaneous-rate (SR) fibers, low-SR fibers have higher thresholds, larger dynamic ranges, smaller effective response areas, and are better able to preserve timing information and amplitude modulation. These characteristics may provide low-SR fibers with an increased resistance to the effects of masking; loss or inactivity of low-SR fibers, as has been observed at high characteristic frequencies in older animals, may make older listeners more susceptible to these effects.

Few studies have focused on age-related differences in benefit from interrupted noise and the effects of reduced audibility due to older listeners' slightly elevated quiet or masked thresholds. Moreover, although there is considerable evidence of age-related changes in various measures of temporal resolution, little is known about recovery from forward masking in older subjects. Thus, it remains unclear whether recovery from forward masking may contribute to older subjects’ reduced benefit from interrupted maskers. To explore these factors, two experiments were conducted whereby simultaneous- and forward-masked thresholds and the masking of speech by steady-state and interrupted noise were measured in younger and older subjects with normal but not identical audiograms. An additional low-level broadband noise was always present to equate audibility across subjects despite differences in quiet thresholds. Results will focus on separating age-related and threshold-related effects, associations between speech recognition and masked thresholds, and differences between predicted and observed benefit from modulated maskers. [Work supported by NIH/NIDCD]
In this talk I will review what I have learned about how humans decode speech. This science depends on error analyses of spoken communication, and the use of statistics, information theory, and psychophysical methods. Applications include improved automatic speech recognition, hearing aids and cochlear implants.

The main points are:
1) we decode phones based on “binary-features,” such as VOT,
2) these binary-features are language and phone dependent, but independent of the syllable, word, or meaning (i.e., context),
3) phones are recognized independently of their word context;
4) many words are correctly recognized once a uniquely defining combination of phones has been uttered, and usually before the word is complete;
5) language (context) models can only correct unambiguous errors;
6) robustness to noise and filtering depends on a healthy cochlea.

The above points are based on the following experimental observations:
1) CV confusion matrices are block diagonal;
2) phones may be modeled as being derived from independent binary-features;
3) we must use nonsense words to make independent models of syllables;
4) ERP recordings show that the recognition process proceeds at the phoneme rate;
5) we know how language is limited in its ability to correct errors;
6) From years of observations of the cochlear-damaged hearing impaired, we know that there is a natural robustness to noise and filtering given a normal cochlea.

There are two categories of “independence models” that may be used to analyze perception errors:
i) product of error probabilities and
ii) product of recognition probabilities.

The first is useful in parallel processing, such as multi-channel inputs. This is the case of binary-feature extraction based on a tonotopic (cochlear) array of inputs. The second is useful in sequential processing, such as in consonant-vowel-consonant (CVC) models. Both types of model were introduced first by Fletcher.

After WWII, George Miller and colleagues at the Harvard Psychoacoustics Laboratory, extended Bell research. They measured the recognition trade-off between task entropy, chance and the signal to noise ratio (Miller, Heise and Lichten, 1951). Miller and Nicely were the first to convincingly identify perceptual binary-features from confusion matrices.

The robustness of speech, which is the most amazing thing about the speech code, has been rarely studied since the 1950's. It seems to me that the next breakthrough in hearing aids must be in the area of robust binary feature extraction, analysis and presentation.
FAST OR SLOW COMPRESSION FOR OPTIMAL SPEECH RECOGNITION IN FLUCTUATING NOISE FOR NORMAL AND HEARING IMPAIRED LISTENERS
Henrik L. Olsen
Institute of Clinical Neuroscience, Karolinska Institutet, Sweden

Although fast-acting compression systems are integral components of modern hearing aids, research results have not consistently demonstrated their benefit over conventional linear amplification. This study intends to examine two factors that may have contributed to this inconsistency in results: audibility of speech cues and alteration of temporal information. We have therefore measured the effect of presentation level and compression characteristics on compression benefit using a simulated three-channel compressor. The peaks of the noise in different octave bands were matched between each compression condition and of the linear reference at three different sensation levels: 5, 15 and 25 dB. Nine different combinations of time constants were tried together with two different compression ratios. An acoustical analysis revealed that the speech-to-noise benefit through the compressor systems ranged from 5 dB for the fastest compressor towards 0 dB for the slowest compressor. Twenty hearing-impaired listeners and 5 normal-hearing listeners participated in the study. Speech recognition was measured with sentences in fluctuating speech shaped noise for all systems. Additionally, a steady state speech noise was measured for the linear system.

Results shows that only one-third of the hearing-impaired listeners had positive or no benefit from fast-acting compression across all presentation levels. The normal-hearing listeners had the largest compression benefit, around 3.5 dB. It was found that a negative compression benefit often was associated with poor speech recognition in noise using linear processing. On the other hand, the listeners with positive compression benefit had very close to normal speech recognition in noise using linear processing and therefore good suprathreshold hearing.

The effect of changing the compression characteristics i.e. time constants and compression ratio revealed different effects on the hearing-impaired listeners. The group of listeners with positive compression benefit showed successively decreasing benefit when the time constants were increased. The same tendency was seen for the normal-hearing listeners and the effect was predicted from acoustical analysis. For the group with worse negative compression benefit the reverse effect was observed.

The result questions the general clinical prescription of compression where the degree of compression uses to be proportional to the degree of the hearing threshold. We have indications that a good suprathreshold hearing is required to benefit from fast-acting compression. For the group of listeners with poor suprathreshold hearing a slow-acting compressor is needed in order to avoid further degradation of speech recognition.
SESSION EIGHT

DEAD REGIONS IN THE COCHLEA

Moderator: Dianne Van Tasell

11:10 AM  EFFECTS OF LOW-PASS FILTERING ON THE INTELLIGIBILITY OF SPEECH IN NOISE FOR PEOPLE WITH AND WITHOUT DEAD REGIONS AT HIGH FREQUENCIES

Thomas Baer, Brian C.J. Moore and Karolina Kluk
University of Cambridge, United Kingdom

People with high-frequency sensorineural hearing loss differ in the benefit they gain from amplification of high frequencies when listening to speech. Using vowel-consonant-vowel (VCV) stimuli in quiet that were amplified and then lowpass filtered with various cutoff frequencies, Vickers et al. (2001) found that the benefit from amplification of high-frequency components was related to the presence or absence of a cochlear dead region at high frequencies. For hearing-impaired subjects without dead regions, performance improved with increasing cutoff frequency up to 7.5 kHz (the highest value tested). Subjects with high-frequency dead regions showed no improvement when the cutoff frequency was increased above about 1.7 times the edge frequency of the dead region. The present study was similar to that of Vickers et al. but used VCV stimuli presented in background noise. Ten subjects with high-frequency hearing loss, including eight from the study of Vickers et al., were tested. Five had dead regions starting below 2 kHz, and five had no dead regions. Speech stimuli at a nominal level of 65 dB were mixed with spectrally matched noise, amplified according to the “Cambridge” prescriptive formula (Moore and Glasberg, 1998) for each subject and then lowpass filtered. The noise level was chosen separately for each subject to give a moderate reduction in intelligibility relative to listening in quiet. For subjects without dead regions, performance generally improved with increasing cutoff frequency up to 7.5 kHz, on average more so in noise than in quiet. For most subjects with dead regions, performance improved with cutoff frequency up to 1.5 – 2 times the edge frequency of the dead region, but hardly changed with further increases. Analyses of the audibility of the stimuli were conducted using the method of calculating the articulation index (AI) described by Moore and Glasberg (1998). The results showed that application of the “Cambridge” formula to the subjects with dead regions resulted in at least partial audibility of frequency components falling within the dead region; the highest audible frequency ranged from about 1.5 to 3.8 kHz. For a given increase in AI, produced by increasing the upper cutoff frequency, the improvement in intelligibility was less for subjects with dead regions than for those without dead regions.

[Supported by the MRC (UK), with additional support from Starkey (USA), Defeating Deafness (UK) and RNID (UK).]
“Dead regions”: Test results and predictions from cochlear models
Dianne VanTasell, William Woods, Martin Rickert, Melanie Gregan and Timothy Trine, Starkey Laboratories, Inc.

Moore and his colleagues have developed an audometric procedure for diagnosing “dead regions” of the cochlea [B.C.J. Moore, M. Huss, D.A. Vickers, B.R. Glasberg, J.I. Alcantara (2000) A test for diagnosis of dead regions in the cochlea, Brit J Audiology 34, 205-224]. Those authors defined “dead regions” as “places” (along the basilar membrane) “with non-functioning inner hair cells (IHCs) and/or neurons.” The diagnostic test is based on the assumption that detection of pure tones at frequencies that would maximally excite a cochlear “dead region” actually takes place when the excitation spreads to adjacent basilar membrane locations where the IHCs and neurons are functioning. In cases where pure-tone quiet thresholds are based on this off-frequency listening, a low-level broadband masking noise will shift behavioral thresholds more than would be expected if the tone were being detected via excitation at a cochlear place corresponding to the stimulus frequency.

Masked pure-tone thresholds for 27 ears of 16 subjects with primarily high-frequency sensorineural hearing losses were measured using the threshold equalizing noise (TEN) developed by Moore and colleagues for use in diagnosing cochlear “dead regions.” As was observed by Moore and colleagues, some (but not all) subjects showed abnormal susceptibility to masking by the TEN. The results were not, however, entirely consistent with the simple concept of off-frequency detection at the edge of a region with completely non-functioning IHCs.

Results will be described and compared to published results in the literature, with emphasis on: 1) alternative explanations for results relating to stimulus and testing artifacts; and 2) predicted results based on cochlear modeling. In particular, modeling of results based on different assumptions about cochlear function (e.g., partially-functioning but not “dead” regions of IHCs) will be discussed.
SESSION NINE

SIGNAL PROCESSING

Moderator: Harvey Dillon

5:15 PM DIGITAL SIGNAL PROCESSING FOR "REAL-LIFE" HEARING INSTRUMENTS

Stefan Launer, Phonak Corporation, Switzerland

Although fast-acting compression systems are integral components of modern hearing aids, research results have not consistently demonstrated their benefit over conventional linear amplification. This study intends to examine two factors that may have contributed to this inconsistency in results: audibility of speech cues and alteration of temporal information. We have therefore measured the effect of presentation level and compression characteristics on compression benefit using a simulated three-channel compressor. The peaks of the noise in different octave bands were matched between each compression condition and of the linear reference at three different sensation levels: 5, 15 and 25 dB. Nine different combinations of time constants were tried together with two different compression ratios. An acoustical analysis revealed that the speech-to-noise benefit through the compressor systems ranged from 5 dB for the fastest compressor towards 0 dB for the slowest compressor. Twenty hearing-impaired listeners and 5 normal-hearing listeners participated in the study. Speech recognition was measured with sentences in fluctuating speech shaped noise for all systems. Additionally, a steady state speech noise was measured for the linear system.

Results shows that only one-third of the hearing-impaired listeners had positive or no benefit from fast-acting compression across all presentation levels. The normal-hearing listeners had the largest compression benefit, around 3.5 dB. It was found that a negative compression benefit often was associated with poor speech recognition in noise using linear processing. On the other hand, the listeners with positive compression benefit had very close to normal speech recognition in noise using linear processing and therefore good suprathreshold hearing.

The effect of changing the compression characteristics i.e. time constants and compression ratio revealed different effects on the hearing-impaired listeners. The group of listeners with positive compression benefit showed successively decreasing benefit when the time constants were increased. The same tendency was seen for the normal-hearing listeners and the effect was predicted from acoustical analysis. For the group with worse negative compression benefit the reverse effect was observed.

The result questions the general clinical prescription of compression where the degree of compression uses to be proportional to the degree of the hearing threshold. We have indications that a good suprathreshold hearing is required to benefit from fast-acting compression. For the group of listeners with poor suprathreshold hearing a slow-acting compressor is needed in order to avoid further degradation of speech recognition.
A METHOD TO MEASURE THE EFFECTS OF NOISE REDUCTION ALGORITHMS USING SIMULTANEOUS SPEECH AND NOISE
Björn Hagerman and Åke Olofsson
Karolinska Institutet, Sweden

Most hearing impairments reduce the ability to understand speech in the presence of background noise. Therefore, there is a great need to enhance the signal-to-noise ratio (S/N) in the hearing aid. It is, however, difficult to estimate the improvement in technical terms. No standardised methods are offered. Our approach is to present speech and noise simultaneously and make two measurements, one of them with the noise phase reversed. Taking the sum of the corresponding two output signals, or the difference, the output speech or the output noise can be extracted. Thus the gain can be calculated for each of them, although they are present at the same time and influence the signal processing of the hearing aid in a normal way. Five different hearing aids were tested with various speech and noise signals and S/Ns. Noise reduction up to 4 dB was measured with noise reduction algorithms (NRAs) activated. The NRAs in the hearing aids measured were more efficient at positive S/N and almost not useful at negative S/N. Fast compression was also tested and degraded S/N for input S/Ns better than -5 dB, but improved S/N for very bad S/N at the input.

PERCEPTUAL CONSTRAINTS TO HEARING AID PROCESSING ALGORITHMS
Michael A. Stone and Brian C.J. Moore
University of Cambridge, United Kingdom

Digital signal processing offers great flexibility in the design of hearing aids. This flexibility comes at a cost that has not been apparent in analogue aids. Aside from the time delays introduced by conversion to and from the digital domain, signal processing that performs frequency response shaping and dynamic-range compression introduce further delay. Other algorithms, which also work in the modulation domain, can require even longer delays. Consideration of the possible disruption to lip-reading led McGrath and Summerfield (1983) to recommend an upper bound of about 40 msec.

Previously, we have shown that this bound has to be reduced still further when taking into account the subjective reaction by the aid wearer to their own voice quality. We have also measured objective parameters related to voice production and shown that there is a more rigid limit of around 30 msec imposed by the speech motor system. These data were derived from algorithms producing delay constant across frequency. More recently we have been looking at the effect of delays that vary across frequency. This paper reviews these experiments to produce design guidelines for future processing algorithms.
SESSION TEN

DIRECTIONAL MICROPHONES

Moderator: Sig Soli

8:00 AM  A NEWLY DEVELOPED IN-SITU MEASUREMENT SYSTEM: THE DIRECTIONAL HEARING AID ANALYZER

Arlene Neuman and King Chung, Michael Steele and Matthew Bakke
Rehabilitation Engineering Research Center on Hearing Enhancement, Lexington Center

Recent advances in digital technology have brought forth significant advances in directional microphone technologies. Many of the high performance hearing aids are equipped with directional microphones. However, there are no available standard clinical or electroacoustical test procedures to allow researchers or clinicians to measure the directional performance when they fit hearing aids to users or to test for microphone drift or degradation of directional performance.

We have developed a Directional Hearing Aid Analyzer (DHAA) that utilizes a probe microphone to measure the in-situ directional performance of a hearing aid in the user's ear canal. The measurements are made from 16 loudspeaker locations (evenly spaced over 360°). The measurement results reflect the directional performance of a hearing aid accounting for the user's body baffle and head shadow effects, individual outer ear transfer function, microphone directivity patterns, hearing aid frequency response and hearing aid signal processing strategies, e.g., the choice of compression characteristics. These results are reported as spectral plots at the 16 loudspeaker locations and as polar plots at different frequencies. In-situ directional performance of commercially available behind-the-ear and in-the-ear hearing aids with directional microphones will be reported. In addition, KEMAR's probe mic response while wearing a custom hearing aid with programmable directivity patterns, namely, omni-directional, cardioid, hypercardioi and supercardioid, will also be reported.

8:35 AM  FREQUENCY EFFECTS AND DIRECTIONAL HEARING AID BENEFIT

Todd Ricketts
Vanderbilt University

Killion et al (1998) have advocated an articulation index weighted directivity index (AI-DI) as an enhancement to the traditional directivity index (DI). It has been proposed that the AI-DI may provide a reasonable estimate of the improvement in speech recognition in noise afforded by directional hearing aids in a diffuse listening environment (referred to herein as directional benefit). The premise forwarded for the AI-DI is that improved directivity in the most important regions of speech (based on AI theory) should be weighted more heavily when calculating DI. While some evidence exists suggesting a
relationship between average DI and behaviorally measured word recognition (Ricketts and Dittrich, 2002), the appropriate frequency importance weightings for DI have yet to be systematically investigated.

In this experiment the impact of the frequency range over which high directivity was present on directional benefit (as measured by word recognition in noise) was examined for three groups of ten listeners (30 total) with varying degrees of sensorineural hearing loss. Word recognition, as measured by hearing aid processed recordings of the Connected Speech Test Version 3 (CSTv3) (Cox et al., 1989) presented at a +2 dB SNR in a semi-diffuse listening environment was evaluated. These speech stimuli were recorded for both directional (high directivity) and omnidirectional (low directivity) hearing aid modes through a KEMAR fit with a single commercial hearing aid. The recordings were then high and low-pass filtered. The filtered stimuli were then mixed back together using the following combinations: Omnidirectional below 750 Hz and directional above 750 Hz; Directional below 750 Hz and omnidirectional above 750 Hz; Omnidirectional below 1500 Hz and directional above 1500 Hz; Directional below 1500 Hz and omnidirectional above 1500 Hz; Omnidirectional below 3000 Hz and directional above 3000 Hz; and Directional below 3000 Hz and omnidirectional above 3000 Hz.

Individual listener’s speech recognition under these conditions (after applying NAL-NL1 prescriptive gain) was compared to articulation index directivity index (AI-DI) predictions assuming a variety of band importance function weightings.

Results to date reveal a significant effect of frequency range of high directivity and an interaction between hearing loss configuration and the appropriate frequency importance weighting function for the AI-DI. The results will be discussed relative to their potential impact on clinical fitting recommendations of existing hearing aids and on the design philosophy for future directional microphone hearing aids in terms of the frequencies for which greatest directivity is of maximum importance.

9:10 AM

THE EFFECTIVENESS OF ADAPTIVE DIRECTIONALITY BY DUAL MICROPHONES IN A DIGITAL HEARING AID

Wouter A. Dreschler and Monique Boymans
Academic Medical Centre, Amsterdam, THE NETHERLANDS

While signal-processing schemes based on spectral and temporal differences have only effects in terms of listening comfort, directional microphones have proven to be really effective in terms of improvement of the signal-to-noise ratio. The introduction of dual-microphone systems has renewed the interests in directionality and various studies point out that a significant benefit can be obtained in specific situations.

This study provides further experimental data on the effects of adaptive directionality in a clinical population and the selection of tests focuses especially on the following questions:
- What is the added value of adaptive directionality relative to fixed directionality measured in the same hearing aids and for the same subjects for single noise sources as a function of azimuth?
- What is the added value of adaptive directionality in conditions with more than one noise source?
- What are the effects of hearing aid type (bte versus itc)?
- Is there a negative effect of adaptive directionality on the accuracy of horizontal localization?

This study also touches two side aspects:

1. The technical specification of directionality is in terms of polar patterns and directivity indices. The polar pattern is usually measured with non-simultaneously presented signals at different azimuths. The directivity index is usually measured with a diffuse sound field. In spite of the different measurement techniques the directivity indices for microphones with a fixed directivity pattern can be calculated from the polar pattern. However, the introduction of microphones with adaptive directionality caused a dissociation between directivity assessed as polar patterns and directivity assessed as directivity indices. It is important to differentiate measures of directivity that have been measured simultaneously and non-simultaneously in technical specifications in the future.

2. In this study we applied both SRT-measurements using an up-down method and Just Follow Conversation (JFC) measurements using with a method of adjustment and we found a clear non-linear relationship between the results obtained with both techniques.

The latter side effects will be discussed in the presentation.
SESSION ELEVEN

ALTERNATIVE THERAPIES FOR SENSORINEURAL HEARING LOSS

Moderator: Jay Rubenstein

10:05 AM  COMBINING ELECTRIC AND ACOUSTIC HEARING FOR SEVERE HIGH-FREQUENCY HEARING LOSS

Christopher Turner and Bruce Gantz
University of Iowa

Many people suffer from severe high-frequency hearing loss, in which the lower frequencies of speech and other sounds are perceived adequately, but the regions of the cochlea responding to higher frequencies are damaged to the point where amplification provides little or no benefit. The possibility that electrical stimulation of the inner ear for high-frequency sounds can be used to supplement the residual acoustic hearing of the lower frequencies is tested in these studies. It is shown that residual hearing can be preserved after introduction of a short cochlear implant, and that such a device can provide a substantial benefit in speech understanding to the patient with severe high-frequency hearing loss. The position of the electrode within the cochlea is shown to be an important factor in the success of such a device.

10:40 AM  PLASTICITY IN THE DEVELOPING AUDITORY SYSTEM INDUCED BY ELECTRICAL STIMULATION WITH A COCHLEAR IMPLANT

Patricia Leake
University of California, San Francisco

Studies in an animal model of congenital deafness have shown significant neurotrophic effects of electrical stimulation, which partially prevents degenerative changes that otherwise occur following early deafness. Stimulation delivered by an implant over a period of several months promotes significantly increased survival of cochlear spiral ganglion neurons, although results vary with the specific parameters of stimulation. Administration of GMI ganglioside (which enhance the activity of neurotrophins) may further improve neural survival. In contrast, cochlear trauma from electrode insertion causes marked neural degeneration in the damaged region.

The functional consequences of electrical stimulation have been studied in acute electrophysiological experiments conducted in the auditory midbrain (inferior colliculus, IC). Results indicate that the fundamental tonotopic organization of the central auditory system develops normally despite severe auditory deprivation after early deafness. However, chronic stimulation delivered unilaterally by a single bipolar channel of a cochlear implant in young, deafened animals can cause marked expansion of the central representation of that channel, distorting the tonotopic gradient of the IC. In contrast,
normal (or even sharpened) central representations may be maintained by highly
controlled, alternating stimulation of two channels.

The marked effects of electrical stimulation in animals deafened early in life
suggest that synchronized neural activity elicited by a cochlear implant can exert a
powerful influence on the deafened auditory system during maturation. Central auditory
signal processing may be profoundly altered by specific experience with electrical
stimulation, especially with respect to the capacity of individual implant channels to elicit
highly restricted central activation patterns. Findings to date emphasize the importance of
initial fitting of the cochlear implant in the naive, developing auditory system. In pediatric
implant recipients, it may be highly advantageous to introduce stimulation in a manner that
emphasizes segregation of inputs from individual channels and encourages their
discrimination.

11:15 AM  TREATMENT OF AUDITORY NEUROPATHY: COCHLEAR IMPLANTS OR
HEARING AIDS
Fan Gang Zeng, Sheng Liu and Ying-Yee Kong, Arnold Starr and Henry Michalewski,
University of California, Irvine, and Jon Shallop, Mayo Clinic

Different from traditional sensorineural hearing loss, persons with auditory neuropathy
typically have otoacoustic emissions but absent neural potentials in their auditory brainstem
responses. In addition, they often complain about understanding speech in noise and have
suprathreshold processing deficits disproportional to the audiograms. The lesion sites in
auditory neuropathy have been narrowed down to the region between inner hair cells and
the auditory nerve, including the hair cell-nerve synapses. The absence of the auditory
brainstem responses is due to either dys-synchronous or insufficient nervous activity.
Because neuropathy patients typically cannot derive benefits from hearing aids, some have
opted for cochlear implantation. With universal infant hearing screening and the ever
increasing life span in today's society, more and more people will be likely diagnosed with
auditory neuropathy or a neuropathy component in their hearing loss. Here we address two
important issues related to the treatment strategies for auditory neuropathy: (1) To what
extent can cochlear implants help? (2) Have we really exhausted all options in hearing
aids?

In cochlear implants, electric stimulation of the auditory nerve induces highly synchronous
nervous discharge and may compensate for or hopefully overcome the dys-synchronous
problem in neuropathy patients. We will review extensive audiological, psychophysical,
electrophysiological, and speech recognition data collected in auditory neuropathy patients.
In particular, we will contrast pre- and post-surgical data in those neuropathy patients who
have received a cochlear implant. We find that persons with auditory neuropathy have often
low-frequency hearing loss, reduced pitch discriminability at low frequencies, severely
impaired temporal processing, and abnormal masking patterns. In contrast, they have
relatively normal intensity processing and pitch discrimination at high frequencies (>4
kHz). Our preliminary data show that the cochlear implant has improved the temporal
processing in neuropathy patients but not to the level of performance as typically observed
in non-neuropathy implant users. This improvement is reflected by better temporal
processing measures and also by the presence of electrically evoked auditory brainstem
responses particularly at slow stimulus rates. We recommend that patients who have an
adaptation problem with high-rate stimulation be fitted with a slow-rate stimulation strategy.
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incorporating the maximal number of electrodes. Functionally, the implant appears to help improve hearing in noise.

The available data also suggest that innovative signal processing in hearing aids may provide benefits to alleviate the problems associated with auditory neuropathy. In cases of hearing loss, audibility ought to be provided by means of amplification. Besides that, the needs in auditory neuropathy patients are drastically different from what are available in most of today's hearing aids. First of all, amplitude compression as employed in most hearing aids is not appropriate for these patients because they have already suffered a significantly reduced ability to follow temporal fluctuations. Alternatively, an envelope expansion scheme should be more beneficial than a linear or compressive scheme. Second, unlike a typical hearing aid user who may have mostly high frequency loss, many neuropathy patients have good sensitivity and discriminability at high frequencies, suggesting that frequency transposition to high frequency can be another beneficial option. Third, some neuropathy patients showed much better performance with live voice than headphone presentation, indicating the need for a greater bandwidth of hearing aids than is conventionally available. Finally, clear speech has shown to provide greater intelligibility in noise than conversational speech, suggesting that neuropathy patients may benefit from a signal-processing scheme enhancing the clear speech components. In short, our design goal should be to make speech clearer not necessarily louder for persons with auditory neuropathy.
Poster Abstracts

Posters for Session A should be up by 8 A.M. Thursday, August 22. Posters down after 10 P.M. Thursday, August 22 or before 7 A.M. Friday, August 22. Presenters should be at their posters from 9:45 - 11:00 A.M.; 4:30 - 5:00 P.M.

POSTER SESSION A
Thursday 8:00AM – 10:00PM

Compression

PA1
Effects of varying hearing aid input-output functions on speech recognition and overall sound quality preference ratings
King Chung, Lexington School/Center for the Deaf, Mead Kilhion and Laurel Christensen, Northwestern University and Etyonic Research

The input-output function of a hearing aid reflects the amount of gain provided at different input levels. It affects the audibility of the sounds and also the comfort of hearing aid use. An inspection of commercially available hearing aids revealed at least five common types of input-output functions: (1) linear peak-clipping (LIN-PC), (2) linear compression limiting (Lin-CL), (3) curvilinear WDRC (WDRC-Curvi), (4) high-level-linear WDRC (WDRC-Kamp), and (5) high-level-limiting WDRC (WDRC-FDRC). In addition, Goldberg proposed an input-output function (WDRC-Gold) that has low-level compression, a mid-level linear region and high-level compression limiting. Few researchers have systematically explored whether any input-output function is better for speech recognition and more preferred at different input levels or whether certain input-output functions are more suitable for listeners with different types of hearing loss. The purpose of this study is to explore the effects of varying the input-output functions on speech recognition and sound quality preference ratings for listeners with flat or mildly sloping sensorineural hearing loss.

The commercially available hearing aids have vastly different implementation of hearing aid characteristics, e.g., level detection systems, attack and release times. The interactions of these characteristics make it impossible to draw conclusion on the effects of varying a particular hearing aid characteristic. Therefore, a two-channel programmable digital hearing aid was built for this study. It allowed the varying of hearing aid input-output functions while keeping other characteristics constant. Seven hundred and twenty IEEE sentences mixed with a four-talker babble were processed by the six input-output functions at input levels of 45, 65 and 90 dB SPL. The sentences were then played back to: (1) subjects with normal hearing, (2) subjects with flat loss, and (3) subjects with sloping loss. Two signal-to-noise ratios were used for each group in order to bracket their signal-to-noise ratios for 50% speech recognition (SNR-50). The gains of the input-output functions were matched at 65 dB SPL. Sentences processed at 65 dB SPL hearing aid input were presented to the subjects' comfortable listening levels and sentences processed at 45 and 90 dB SPL input were presented at levels as if the volume control was fixed. The experimental conditions were counterbalanced across subjects, input-output functions and input levels. Results suggested that a good hearing aid, which yields low SNR-50s and high preference ratings, makes soft sounds audible, and loud sounds comfortable and free distortions. The exact input-output function implemented to achieve these goals is less important.

PA2
Changing speech perception and localization in new hearing aid users with wide-dynamic-range multichannel compression and linear fittings
E. William Yund, and Christina M. Roup, VA NCHCS, Helen J. Simon and Al Lotze, Smith-Kettlewell Eye Research Institute

A new hearing aid user gains access to auditory information that has become progressively unavailable due to hearing impairment. Previous work indicates that speech perception and localization adapt toward optimizing performance with the reduced auditory cues as hearing loss develops. This suggests that the auditory information provided by the new hearing aid may not be of immediate value without additional adaptation or learning. Other research also indicates that a period of acclimatization is required for the listener to obtain maximum benefit from new hearing aids. It has been suggested that there may be differences in the time course of acclimatization for different types of hearing aids and that such differences could interfere with comparative evaluations among hearing aids.

Unless performance measures were repeated over time, effects of duration of use and relative benefit would be confounded. The present study, which will continue over the next two years, was designed to clarify some aspects of these issues.

Adult participants with bilateral, mild to moderately-severe SNHL were fit with binaural ITC hearing aids. The hearing aids were programmed with either wide-
dynamic-range multichannel compression (WDRMCC) or linear amplification (NAL-R) programs and used for 32 weeks. The WDRMCC programs conform to the manufacturers' specifications (GN Resound and Sonic Innovations). Prior to hearing-aid use and during the use period, four experimental measures were monitored: (1) the Profile of Hearing Aid Benefit (PHAB), (2) sound localization, (3) consonant discrimination and confusion matrices, and (4) fMRI activations for difficult consonant discriminations. At the end of the first 32 weeks, the programs in the hearing aids will be changed (from WDRMCC to NAL-R, or from NAL-R to WDRMCC) and the same set of measures will be obtained with the second program. While the first 32 weeks measure performance for listeners with no previous hearing aid experience, the second 32 weeks measure performance for experienced hearing aid users whose previous hearing aid experience is well defined and whose performance with those hearing aids is well characterized on the same experimental measures. Preliminary results from the first 32-week period will be presented, with the primary focus on the similarities and differences between WDRMCC and NAL-R fittings, in hearing aid benefit, sound localization, consonant discrimination, and consonant confusions. [This work is supported by VA RR&D. Hearing aids provided by the manufacturers.]

**PA3**

**The effect of varying compression time-constants using experimental programs in the Phonak Claro hearing aid**

Thomas Stainsby, Brian C.J. Moore and José Alcántar, University of Cambridge England, and Volker Kuechel, Phonak AG, Switzerland

The relative efficacy of five hearing-aid programs that differed in their choice of compression attack and release time constants was assessed. The study used the Phonak Claro hearing aid programmed with custom software. Four different programs in the aid differed in the compression time constants used across twenty frequency channels, in the following configurations: (1) 8 ms attack and 32 ms release time in all bands; (2) a progressive decrease in attack time from >500 ms at the lowest frequency to 72 ms at the highest frequency and in release time from >500 ms to 108 ms; (3) a progressive increase in attack time from 32 ms at the lowest frequency to >500 ms at the highest frequency and in release time from 80 ms to >500 ms; and (4) 24 ms attack and 110 ms release in all frequency bands. Stimuli were presented via a circumaural headphone mounted over the hearing aid. Presentation was monaural. In a fifth condition that did not use the Claro aid, participants were presented with a headphone signal that had been filtered to implement the Camfit linear-gain hearing-aid prescription (Moore and Glasberg, 1998). Five subjects with moderate sensorineural hearing loss were tested in a counterbalanced order across conditions. Gains and compression ratios in conditions (1) to (4) were determined for each subject from the audiometric thresholds, using the DSL prescription method (Cornelisse et al., 1995) as implemented in the Phonak fitting software. Speech perception was assessed using VCV nonsense-syllable lists presented in three different types of noise (multitalker babble, cafeteria, or single female speaker), or in quiet. Percent correct was measured as a function of speech-to-background ratio. An ANOVA revealed significant effects of the type of background noise and of condition. Scores obtained with the linear-gain Camfit formula implemented via headphones were significantly higher than for any of the hearing-aid conditions using compression. Longer time constants, whether at low or high frequencies (conditions 2 and 3) tended to produce poorer scores than conditions with short time constants (conditions 1 and 4), but this effect just failed to reach statistical significance.

**PA4**

**Quantifying the effect of release time from compression on temporal cues of speech**

Loriene Jenstad and Pamela Souza

University of Washington

There is considerable interest in determining the optimal release time for wide-dynamic range compression hearing aids (e.g., Bentler & Nelson, 1997; Neuman et al., 1995). To date, there is no consensus on which release times will optimize both speech intelligibility and speech quality. At present, we lack a clear understanding of what is happening to speech when it is compressed.

The gross time-intensity variation, or temporal envelope, is the speech dimension most likely to be affected by changing release time. Thus, the research presented here addressed two questions: first, how does release time affect the temporal envelope of speech; and second, how do the alterations of the temporal envelope relate to speech intelligibility? Two measures were used to quantify the effects of release time ranging from 12 ms to 800 ms on vowel-consonant nonsense syllable tokens: the envelope difference index (EDI; Fortune et al., 1994) and the consonant-vowel ratio. Shorter release times had a greater effect on the envelope difference index and the consonant-vowel ratio than longer release times. This effect was greater for higher input levels than low and
moderate input levels (p<.001). A hierarchical cluster exploratory analysis was used to identify clusters of consonants that were affected in similar ways by release time. The groups were interpreted to correspond to manner and voicing distinctions, features that are carried by the temporal envelope. This quantification of the acoustic effects of release time will be related to measures of speech intelligibility. The implications for hearing aid fittings will be discussed.

[Research supported by the ASH Foundation and the Canadian Institutes of Health Research.]

## Noise Reduction

### PA5

**Kalman filtering and Independent components analysis for the extraction of speech from background**

Karl Wiklund, McMaster University, Adaptive Systems Laboratory

A frequent problem encountered by those with hearing loss is the detection and understanding of a desired sound in a noisy environment. An example of such is the problem of understanding speech in a noisy room. For the purposes of a hearing aid, it is desirable to eliminate as much of this interference as possible before further processing is carried out to compensate for the specific deficiencies of the wearer’s hearing. This is similar to the problem of blind-source separation, where multiple interferers are present, and the goal is to estimate all of them.

However, the environment in which a hearing aid must operate presents severe problems for existing blind source separation algorithms. The presence of noise, and a potentially non-stationary mixing system are two prominent difficulties that must be overcome if such methods are to be effective. In addition, it is necessary for rapid de-mixing to take place, something which most blind source separation algorithms are incapable of.

By describing blind source separation in terms of known Bussgang approaches to the problem, a relationship to Kalman filtering can be described. This is beneficial since the Kalman filter explicitly takes noise into account, as well as the possibility of dynamically changing states. The non-linear nature of the problem however, necessitates the use of a non-linear form of the algorithm, such as the Unscented Kalman Filter.

Using the UKF, two new BSS algorithms have been written and tested on a variety of signals, including speech. The methods have a number of attractive qualities including robust performance in noise, system tracking, and rapid convergence. In addition, preliminary results suggest that these methods possess a degree of universality in that the same contrast function is capable of separating both sub- and super-gaussian sources. Here we present the new methods, demonstrate their performance using speech data, and discuss deficiencies and directions for future work.

### PA6

**Single-microphone noise reduction for hearing aids: Low-power and low-delay DSP implementation using the Ephraim & Malah algorithm**

Alain Dufaux, Tam King, Hamid Sheikhzadeh, Robert Brennan and Todd Schneider
Depfactory Ltd., Canada

**Single Microphone Noise Reduction** is a difficult problem, especially in hearing aid applications, which require critical low-delay and low-power constraints. Traditional techniques like spectral subtraction are generally considered poor in performance, and subjective tests often show that the original noisy signal is perceptually preferred to the processed speech. This result is due to artifacts that are introduced in the signal, taking the form of a disturbing musical noise.

Today, the Ephraim & Malah [Capp94] algorithm is getting widely accepted [Marz01] as one of the best solution for processing signals in stationary and even slightly statistically varying background environments. It’s main advantage is to decrease the residual musical artifact to an imperceptible level. The particularly good result of the Ephraim & Malah technique resides in a dual frequency-selective estimation of the Signal-to-Noise Ratio (SNR) of the input speech, and particularly in the mutual behavior of those two SNR values:

- The first estimate, an instantaneous estimation of the \( SNR \) (the “a-posteriori” SNR, \( R_{post} \)) is performed in each successive frame.
- The second estimate, a smoothed \( SNR \) value (the “a-priori” \( R_{pre} \)) is recursively determined for each frame, the amount of smoothing being made adaptive to the a-posteriori SNR according to a nonlinear expression. Smoothing is more important in case of low a-posteriori SNRs.

In this paper, an optimized implementation of the Ephraim & Malah noise reduction is considered on an ultra low-power miniature DSP system. The DSP system consumes about an order of magnitude less power (<1mWatt) than systems performing similar
tasks. This efficient implementation is accomplished through the use of two processing units running concurrently. In addition to the 16-bit fixed-point Harvard DSP core, an efficient WOLA ("Weighted Overlap-Add", [Bren98]) filter-bank unit performs the windowing, FFT and vector multiplications, required to transform the signal into frequency sub-bands, apply the Ephraim & Mahal algorithm in each band (gain application), and re-synthesize the output signal. The gains depend on particular values of \( R_{post} \) and \( R_{prio} \). These gains are efficiently determined, using a 2-D table lookup, in which entries are the quantized SNRs, \( R_{post} \) and \( R_{prio} \). The result is a system with a group delay of only 10 ms, providing an impressive noise reduction quality in preliminary subjective listening tests, compared to traditional methods. Further objective and subjective evaluation is in progress.

**PA7**

Reduction of the cocktail party effect for hearing impaired patients by time-frequency signal processing.

*Michael Orr* and *Brian Lithgow*, Monash University, Robert Mahoney, Australian National University

The inability to conduct conversation or distinguish sounds in noisy environments is a common complaint of hearing aid patients. Extensive research into reducing this "cocktail effect" carried out over recent years has met with limited success. Research conducted into the enhancement of speech for normal hearing subjects has encountered similar problems. However, an algorithm using techniques from both signal processing and hearing aid research utilising both time and frequency information could improve both research fields.

The proposed system uses a filter bank of Morlet wavelets to provide time frequency information in the 62 hertz to 8 kilohertz range. Three parallel algorithms, characterising the long-term, recent, and present speaker features process the time-frequency information to produce separate sub-space representations of the sound. Information from all three algorithms is combined using both statistical measurements and linear algebraic techniques.

Whilst only very preliminary results are available, the authors believe a hearing aid using this algorithm in combination with conventional compensation techniques can produce a marked reduction in the cocktail party effect.

**PA8**

Objective assessment of noise reduction algorithms in digital hearing aids

*Vijay Parsa* and *Donald Jamieson* National Centre for Audiology, University of Western Ontario, Canada

Individuals with sensorineural hearing loss report particular difficulty in understanding speech in noisy backgrounds. In fact, poor speech intelligibility in noise is one of the most important reasons for obtaining a hearing aid, and is a significant influence on the level of satisfaction reported by hearing aid users. Recent technological innovations in digital hearing aids have led to the development of several "advanced" noise reduction methods. However, the hearing aid industry does not use any validated or standardized measure to assess the Signal-to-Noise Ratio (SNR) improvement provided by these algorithms. This absence of a standardized measure prevents clinical audiologists from assessing the relative benefits of various devices that offer alternative noise reduction algorithms. In this paper, we describe a technique for objectively measuring the noise reduction performance of a hearing aid.

Our technique uses the system identification method. This approach employs a time-varying adaptive filter to model the dynamic behaviour of the hearing aid with speech plus noise stimuli. The noise reduction performance is assessed by comparing the model residue before and after application of the noise reduction algorithm. Significant features of this approach include: 1) flexibility -- speech and music stimuli in the presence of a variety of background noise sources (narrowband, broadband, multi-talker, impulsive etc) can be used to assess the performance of noise reduction algorithms under a variety of operating conditions; and 2) versatility -- single microphone, dual-microphone, and array processing noise reduction algorithms, and FM systems can all be assessed for their performance.

We will present results from computer simulations, and from experimental data collected from commercially available hearing aids to show the effectiveness of the system identification approach in predicting the performance of alternative noise reduction techniques.
Phase-opponency based noise reduction
Michael Anzalone and Laurel Carney
Syracuse University

A signal processing technique for noise reduction has been developed, based on a physiological model for masking in the healthy auditory system. The technique, based on the phase-opponency (PO) model of Carney, et al. (Acustica, 2002, in press) doesn’t require an estimate of the additive noise and is less computationally intensive than previous noise reduction techniques. While the PO model, which is based on neural timing, is physiologically limited to low frequencies, it can be extended to high frequencies for the purpose of signal processing. The input is first filtered using a bank of overlapping linear gamma-tone filters. Pairs of overlapping filters can be chosen such that the filter phases differ by 180° (thus they are “phase-opponent”) at a particular frequency. The presence of a narrowband signal near this frequency results in a drop in the value of the running cross-correlation between the two filter responses. Pairwise cross-frequency correlations for the entire filter bank can be used to detect narrowband signals as well as to specify the gains of another linear gamma-tone filter bank. Frequency ranges that contain narrowband signals are amplified and other frequencies are attenuated, resulting in noise reduction.

Preliminary testing of the PO technique has been done on both pure and FM tones in additive wideband noise. The results show that the technique functions over a wide range of signal to noise ratios. Use of the technique can give substantial improvements in the signal to noise ratio for preliminary tests with narrowband signals and broadband noise. These results suggest that the PO technique is a feasible approach to noise reduction. Further studies need to be performed to test the technique with more complex signals such as speech and with more realistic noise. The versatility and simplicity of the PO technique suggest potential applications in hearing aids, the front ends of automated speech recognition (ASR) devices or other communication devices.

Probabilistic de-noising model for auditory spectrum and robust speech representation
Xugang Lu, McMaster University

Traditional speech enhancement methods for robust speech representation usually improve the SNR based on MMSE, such as spectral subtraction and Kalman filtering. Also, maximum likelihood based speech estimation from noisy speech signal is used on full band of the speech signal, also the noise is estimated on full frequency band. Apparently, noise energy distribution changes differently in different frequency bands and different time windows. In this paper, a probabilistic de-noising model used in each auditory frequency channel is proposed, maximum likelihood estimation for speech signal is done in each auditory frequency channel. In this estimation, not only the nonlinear frequency resolution perception mechanism is combined, but also some hearing perception mechanisms can be incorporated into the calculation, such as temporal masking mechanism, lateral inhibition mechanism, etc. Some mechanisms are used for de-noising to reduce noise level, some mechanisms are used for enhancing those dominant perception features which are not easily masked in noisy condition.

The proposed probabilistic de-noising model is as following. 1. First speech signal is decomposed into 20 auditory frequency channels by basilar membrane which function is the same as constant Q band pass filtering, nonlinear compression function of half wave rectifying function and low pass filtering are used for temporal profile extraction in each frequency channel. 2. The probabilistic de-noising model is used in each sub-band, we suppose that the sub-band output is made up of additive components of useful speech signal and noise signal, the probabilistic de-noising model uses maximum likelihood estimation method to estimate the valid speech signal from the noisy observation. The valid speech signal is estimated by reducing the noise threshold in each sub-band. The noise threshold is estimated adaptively based on local time window observations, then the valid speech signal is calculated by reducing the noise estimation adaptively. 3. After getting the de-noised output of each auditory frequency channel, a temporal integration is used for local signal energy calculation to get the de-noised auditory spectrum. 4. Enhancing the dominant spectral peaks, a local lateral inhibition mechanism is used for the nonlinear frequency interaction, thus the enhancement auditory spectrum can be gotten.

So far, because our experiments are based on speech recognition performance, robust speech recognition is used to test the processed signal, the enhanced auditory spectrum is transformed into cepstral domain with a de-correlation transform by DCT, HMM based training and recognition methods are used for the experiments. Results show that the enhancement processing features are much more robust than traditional features, it can get about 10-30% improvement in recognition rate in different SNR conditions.
situations (experiments are done based on isolated key words recognition).

In conclusion, the proposed enhancement method have the promising advantages, not only the theoretical estimation for clean speech from noisy speech based on probabilistic model is used on sub-band, but also much more hearing perception mechanisms are combined: nonlinear frequency resolution, dynamic compression, adaptive noise threshold detection, lateral inhibition (frequency suppression). The speech representation based on this model is much more robust than traditional model based representation.

Speech Perception and Psychophysics

**PA11**

Measuring loudness of long and short tone using magnitude estimation

Michael Epstein, Mary Florentine and Soren Buus, Northeastern University, Boston

Loudness is one of the most important perceptual qualities of sound. Much effort has been expended in developing reliable methods of measuring loudness, but most of the methods have used long-duration sounds. Because most natural sounds are characterized by relatively brief periods of high intensity, it may be important to use short sounds for loudness measurements. Whereas magnitude estimation has been shown to be useful for measuring loudness of long sounds, McFadden (1975) questioned the accuracy and reliability of this method for measuring the loudness of short sounds. He suggested that it produced unreliable results and should not be used to assess how loudness depends on stimulus duration. To examine this issue further, the present study obtained loudness functions for 5- and 200-ms tones in nine normal listeners using magnitude estimation. Loudness matches between the short and long tones were also obtained using an adaptive 2IFC procedure. The average amounts of temporal integration (defined as the level difference between equally loud short and long tones) obtained with the two procedures show good agreement. However, this may not apply to individual listeners. Some listeners show poor agreement, whereas others show good agreement. These results indicate that magnitude estimation provides a rapid and accurate means for assessing group-average loudness functions for tones of different durations. Nonetheless, it appears that magnitude estimation often is too susceptible to judgment biases and variability to reveal detailed features of individual listeners' loudness functions for tones of various durations, even if it does reveal their general form.

[This research was supported by NIH/NIDCD grant R01DC02241]

**PA12**

Development of a self-administered hearing test for the elderly: effect of variability of equipment

John Culling and Fei Zhao, Cardiff University, UK, Dafydd Stephens, University Hospital of Whales, UK

There is an increasing need for a cheap and simple screening test for presbyacusis. We are developing such a test, based on speech in noise.

The test is intended for self-administration with the user's own domestic audio equipment. An important pre-requisite to this approach is to establish that variations in the quality of domestic audio equipment will not blur the differences in performance between those with and without a hearing impairment. A series of physical measurements and psychoacoustic experiments indicate that equipment of widely varying price, including some of the cheapest available, produces very consistent recognition of speech in noise from normally-hearing listeners. Measures of frequency response for different headphones show large variations at high and low frequencies, but a flat response from every pair in the mid frequencies important for speech. Total harmonic distortion (THD) from these pairs of headphones varied between 0.1 and 4 %, but measurements of speech reception thresholds (SRTs) using deliberately introduced distortion (from an instantaneous compressive non-linearity) found little increase in SRT (about 1 dB at 8% THD). Measurements of frequency response and distortion from audio sources, particularly CD players, but also audio tape players, showed substantially less variability in frequency response and THD than headphones. Pilot version of the test, using bisyllabic words in speech-shaped noise, have been created and calibrated by measuring psychometric functions among normally-hearing listeners. Some words displayed markedly steeper psychometric functions than others. Consistent with the previous findings, these pilot versions of the test produced indistinguishable results from all headphones tested for normally-hearing listeners. Pilot versions of the test have also been presented to normally-hearing listeners over loudspeakers in different rooms. Preliminary results indicate that there is also little influence of the presentation.
room or equipment. Results from hearing impaired listeners have been more variable. Performance is worse on average, but seems to be more affected by the quality of the equipment than for the normally-hearing.

PA13
Tone decay for hearing-impaired listeners with and without dead regions in the cochlea
Martina Huss and Brian C.J. Moore, University of Cambridge

A characteristic of normal hearing is that a sustained tone with a frequency within the standard audiometric frequency range remains audible when presented at a level well above absolute threshold. However, for pure tones with frequencies well above 8 kHz the loudness may decrease within seconds and the tones may decay to inaudibility even when presented at a level between 20 and 40 dB SL. Scharf (1983) suggested that loudness adaptation depends on whether the excitation pattern evoked by a tone in the auditory system is broad or spatially limited. Pure tones of low sensation level and/or with a frequency close to the upper limit of hearing have a restricted spread of excitation, which may lead to adaptation. The present study investigates the perceived decay of pure tones for nine normal-hearing subjects (absolute thresholds better than or equal to 20 dB HL) and twelve subjects with moderate to severe sensorineural hearing loss, using a wide range of frequencies (0.125, 0.5, 0.75, 1, 2, 3, 4, 6, 8 and 12 kHz) and a method similar to that proposed by Carhart (1957). A complete loss of function of inner hair cells over a certain region of the cochlea, a “dead region”, had previously been diagnosed for seven of the twelve subjects (Moore, 2001). No consistent association was found between the degree of tone decay and the presence of a dead region. Subjects with extensive dead regions did not experience significantly more tone decay (within one minute of presentation) than subjects without a dead region even when the frequency of the tone fell within or close to the edge of a dead region. These results do support the hypothesis that tone decay depends on the spread of audible excitation. The results are consistent with the idea that tone decay may depend on the physiological condition of the inner hair cells and/or associated synapses and the number of inner hair cells involved in the transduction of a sustained pure tone.

PA14
Prevalence of dead regions in patients with high-frequency sensorineural hearing loss
Sarosh A. Kapadi, University of Southampton, Institute of Sound and Vibration, UK, Samantha Blakemore, Audiology Department, Royal Sussex County Hospital, Debra Graumann, Audiology Department, Royal South Hants Hospital, Andrew Phillips, Audiology Department, Royal Berkshire Hospital

The term “dead region” refers to a section of the basilar membrane at which there are no functioning inner hair cells (IHCs) and/or afferent neurons. Although no amount of amplification will evoke neural responses from a dead region, acoustic signals at its characteristic frequency may nevertheless be detected by spread of excitation to functioning adjacent IHCs (off-frequency listening). The presence of a dead region may have important implications for hearing aid fitting. A number of studies have shown that amplifying frequencies that fall within a dead region is not advantageous (and may be disadvantageous) for speech recognition, presumably because the off-frequency signals are not interpreted correctly, or degrade the “on-frequency” signals within the same channels.

Although previous studies have indicated that dead regions may be common in hearing-impaired individuals, no formal estimates of their prevalence in representative clinic populations have been reported. The present study sought to estimate the prevalence of dead regions in patients with high-frequency sloping sensorineural hearing loss attending a hearing aid clinic. The relationship between the occurrence of dead regions and the audiogram pattern in these patients was also examined.

Sixty-six ears from 40 patients attending the main hearing aid clinic in the city of Southampton were tested. All ears had high-frequency hearing losses with audiogram slopes of at least 20 dB/octave over at least one octave. Dead regions were identified using the threshold equalising noise (TEN) test, using TEN (masking) levels of 70 and 80 dB/ERB.

Overall, 82% of ears tested exhibited a dead region at least at one test frequency (95% CI 71%-89%). A similar estimate of prevalence was obtained in left ears only and right ears only. Prevalence was greatest at frequencies of 4000 and 5000 Hz. However, it is likely that the true prevalence at frequencies above 5000 Hz was underestimated by practical limitations of the test. Dead region prevalence increased with increasing hearing threshold level, up to approximately 70 dB. Within the sample tested, no
relationship was observed between prevalence and the slope of the audiogram. These results suggest that the prevalence of dead regions in hearing aid clinic populations may be greater than previously suggested. However, the study also indicated that the TEN test and diagnostic criteria currently proposed need further validation as an accurate instrument for the diagnosis of dead regions.

**PA15**

Psychophysical tuning curves measured with hearing aids: An alternative method
Patrick Murphy, Michael Nilsson, Victor Bray Jr., Sonic Innovations

Psychophysical Tuning Curves (PTCs) have been used to determine the frequency selectivity/resolution of the auditory system. The methodology for measuring PTCs is time consuming and difficult for the average researcher, let alone sites with limited equipment. A new method involving the measurement of adaptive thresholds in the presence of narrowband maskers under headphones has been developed to allow the measurement of PTCs with hearing aids. The goal is to evaluate the impact of amplification on frequency resolution in normal-hearing and hearing-impaired listeners.

This presentation will focus on describing the procedures used to measure PTCs in hearing impaired subjects as well as in normal listeners. Briefly, 2000 Hz pure-tone pips with a duration of 150 msec are continually presented every 500 msec to one ear via THD-40 headphones. The 300 Hz narrow-band masker was designed to be slightly wider than the calculated critical band for a 2 KHz probe signal to ensure maximum masker efficiency. The masker frequencies were presented in 200 Hz intervals, with the lowest masker being 300 Hz, and the highest 3000 Hz. This masker interval produces well-defined PTCs with observable LF slopes, HF slopes, and LF tails. The probe tone level is based on the subject’s threshold to the tone pips in quiet + 5 dB. The Masked thresholds are averaged across three consecutive ascending and descending runs.

Using this procedure, 5 hearing impaired subjects were tested unaided, with linear amplification, and with WDRC to evaluate their effect on the slopes of the PTCs. It is proposed that multichannel compression may be effective at reducing the upward spread of masking, thus improving the Signal-to-Noise ratio when listening to speech in noise. Given this, we should expect an increase in the LF slope of the PTC relative to the linear amplification and unaided conditions. Preliminary results will be presented and discussed.

**PA16**

Information theoretic measures for discrimination of stop consonants
Sharba Bandyopadhyay, Murray Sachs, Eric Young, Center for Hearing Sciences, Johns Hopkins University

Recent studies on discrimination of speech in the auditory nerve (AN) assume some form of neural code for the representation. That is, an assumption that the central nervous system (CNS) applies to the responses of auditory nerve fibers (ANFs). In the present study discrimination of voiced stop consonants has been examined electrophysiologically and discriminability has been quantified using information-theoretic distance measures based on classification theory, independent of an a priori knowledge of the code (Johnson et al, 2000). These measures quantify how much each portion of the response contributes to information coding and the fidelity of the encoding. Individual AN fiber responses to five synthesized CVC_CV speech segments (e.g. /dæk_dæk/) at 70 and 50 dB SPL were recorded using standard electrophysiological techniques. The consonants fall within a continuum from /b/ to /d/. The vocalic portions of the stimuli were identical. The F2 and F3 starting frequencies vary from 0.9 kHz to 1.7 kHz and 2.0 kHz to 2.8 kHz respectively, in steps of 0.2 kHz. Discriminability between each stimulus and the first stimulus was quantified using the de-biased information measure. Distances were large at BF's where the formants differ and the greatest distances were observed for fibers in the region of the high frequency bursts. F2 and F3 JNDs were computed from the data assuming that fibers in F2 region code for only F2 and that fibers in the F3 region code for only F3. These results show that F2 is coded with greater fidelity than F3 and that JNDs are larger in the syllable final consonant than at both the releases, which are almost equal. Low spontaneous rate fibers code for the formants with greater fidelity than high spontaneous rate fibers at 70 dB SPL, but the difference decreases with level. Additional modeling experiments were done with larger sets of stimuli to separate the effects of F2 and F3 and to study differences in coding of upward and downward transitions.

[Supported by NIDCD grant DC00109]
A physiologically based predictor of speech intelligibility and listening environment adversity for normal-hearing and hearing-impaired listeners

Ian Bruce, Jeff Bondy, Simon Haykin, and Suzanna Becker, McMaster University

The low power, low MIPS available in general-purpose DSP hearing aids force algorithms to be extremely lean. The difficult environment in which an aid is required to operate often requires a very robust implementation, reducing the number of MIPS available for performance enhancement algorithms.

The Discrete Wavelet Transforms (DWTs) have been shown to be less computationally expensive than the Discrete Fourier Transform (DFT) in some cases. Also, beamforming wideband signal approximations done by applying phase weights produce errors as the bandwidth versus the center frequency grows (Buckley 1987). The family of power minimization algorithms can only produce phase weights for a narrow band without producing error. The error from phase weight implementation increases as the Q of the signal of interest shrinks. Bifurcating DWT trees produce constant Q frequency bands. Therefore, there is a natural fit to using the DWT decomposition for broadband beamforming, in that consistent error bounds are formed in each subband.

To make the representation optimal, one must always be aware of the source, environment and receptor. By utilizing a constant Q decomposition we match the assumptions of the algorithm and the environmental characteristics. We are studying several subbanding analysis structures that attempt to match the receptor and source. 1) A bifurcating tree, with frequencies resolved more tightly in proportion to the Articulation Index (AI). This is an attempt to match the frequency decomposition characteristics of the human auditory system. 2) A temporally efficient tree whose subbranches are split to produce good time resolution to capture fast transients from the source (i.e. most consonants) but still shows good frequency resolution keyed to the critical band structure of the AI. In this way we are trying to match the frequency decomposition characteristics to both the source (speech) and the receptor (the auditory system).

These two algorithms are compared, using performance and computational efficiency as metrics, with the standard DFT implementation. It is shown that the subband strategies produce good AI-weighted directivity gain at a reduced computational load.

The impact of hearing aid signal processing on the vowel formant structures of children with moderate-to-severe hearing loss

Sheila Pratt, Judith Grayhack, Catherine Palmer, Lisa Taubman, University of Pittsburgh, Diane Sabo and Julie Hauser, Children's Hospital of Pittsburgh and University of Pittsburgh

The loss and restoration of auditory feedback in people with little and no hearing produces notable reversals in speech precision, particularly in adults who acquired their hearing loss after speech and language maturation (Lane et al, 1995; Walstein, 1990). The impact of less severe hearing loss on auditory feedback for speech is less well documented, although the indication is that adults with mild to severe adventitious hearing loss exhibit no loss of speech clarity, and restoration of audibility in the frequency regions corresponding to their hearing loss produces little, if any, perceptual differences in their speech (Goel & Kaufman, 1984). In contrast, children with moderate hearing losses exhibit both segmental and suprasegmental speech differences even while wearing hearing aids implying developmental differences in the use of auditory feedback for speech (Eifenein et al, 1994). Moreover, the pattern of speech differences produced by children with hearing loss suggest that their auditory feedback mechanisms may be more sensitive to acoustic perturbations than those of adults. Moreover, case reports have suggested that acoustic perturbations associated with signal processing can impact the speech of some young children although no research has systematically examined the potential effects. To that end, the following preliminary study was implemented to look at the effects of three common hearing aid signal-processing configurations on the vowel formant characteristics of children with moderate-to-severe hearing loss.

Six children (aged 5–7 years, two girls, four boys) with moderate-to-severe sensory hearing loss served as subjects. The children were previous users of linear hearing aids and had intelligible speech, age-appropriate phonological inventories, intact oral mechanisms, and normal word recognition skills. Speech samples were collected from the children when fitted with laboratory hearing aids that were programmed to three different types of signal processing configurations: single-channel linear with volume control, single-channel WDRC, and two
channel with WDRC below and adaptive compression above 1500 Hz. The study was double-blinded with regard to processing configuration and the order was counter-balanced across the children. A speech sample also was obtained from the children without benefit of hearing aids. The speech samples consisted of /b/v/u/ single word productions elicited from pictures in random order. The samples were recorded shortly after each processing configuration was first fitted, one week later, and approximately two months after the fitting. In addition, the children’s perception of F2 loci was assessed with an identification task during each test session. First and second formant center frequencies and onset transitions of the words were measured from spectrographic display with formant tracking. The results from the final recording session for each signal-processing configuration indicated no difference in the F1 transition ratios across processing configurations, however, differences were observed with the F2 transition ratios, particularly for the high and mid front vowels. The F1 and F2 center frequencies also were different across the configurations resulting in differences in vowel area. Most notably, the F2 was higher on the high and mid front vowels when produced in the two-channel condition than in the other conditions. [This study was funded by grants from Oticon, Inc., the National Organization of Auditory Research, and the University of Pittsburgh.]

Directional Microphones

PA19

The validation of a sound field simulator for measurement of the real world performance of directional microphones for hearing aid

Cynthia Compton, Gallaudet University, Arlene Neuman, and Harry Levitt, City University of New York

The amount of noise reduction directional microphones can provide is related to their directivity as well as to the settings where they are used. The latter has not been easy to replicate in the laboratory, indicating a need for a research tool for assessing the expected real-world benefit of such microphones. This investigation compared the effectiveness of four test environments in assessing the real-world directional microphone performance: Two traditional sound booth environments, a real restaurant, and a newly developed sound-field simulation system consisting of an octagonal array of eight loudspeakers. Tested with binaural KEMAR recordings of three hearing aid microphones having radically different directional characteristics, normal hearing subjects required identical signal-to-noise ratios (SNRs) for 50% correct performance (HINT) in both the real and simulated environments. Further, scores improved as directivity increased. This did not occur in the traditional settings. Clinical and research ramifications of these results will be discussed. [This investigation is the doctoral dissertation of the first author and was supported by the Graduate Center of the City University of New York and by a student grant from Etymotic Research.]

PA20

The effect of directional hearing aid microphones on the localization ability of normal hearing and hearing-impaired adults

Arlene Neuman, City University of New York, and King Chung, Rehabilitation Engineering Research Center on Hearing Enhancement, the Lexington Center

Several current hearing aids now have programmable directivity, i.e., the dispenser may program the hearing aid to have cardioid, supercardioid, hypercardioid, or omnidirectional directivity patterns. While the cardioid pattern will attenuate signals originating from the rear, hypercardioid and supercardioid patterns of directivity attenuate signals originating from the sides. It is unknown that whether hearing aid users can identify the direction of the speech source and subsequently turn to the speaker in order to obtain the full benefits of directional microphones. The purpose of this study is to examine listeners' ability to identify the source of sounds originating from 16 loudspeaker locations evenly spaced in 360° azimuth while they are using hearing aids with directional microphones.

Subjects listen to stereo recordings of speech and noise stimuli made through a pair of hearing aids with programmable directivity mounted on a Knowles Electronic Manikin for Acoustic Research (KEMAR). The KEMAR recordings are played through insert phones while the subject is seated at the position where the KEMAR recordings were made. The subject will identify the speech source by naming the speaker number. Subjects’ localization ability will be tested in both binaural and monaural conditions (i.e., binaural, right only, left only). Localization data will be reported for normal hearing and hearing-impaired subjects. Results of this study will give insights to the ability of hearing aid users to localize the sound source when using hearing aids with directional microphones of varying directivity. [Research supported by NIDRR]
Distance and reverberation effects on directional benefit

Benjamin Hornsby and Todd Ricketts, Vanderbilt University

Previous research suggests that unaided and aided speech recognition performance in noisy, reverberant environments generally decreases with increasing listener to source distance, at least until the source reaches the "critical distance" from the listener (e.g., Peutz, 1971). This decrement occurs even when the speech source level is held constant at the listener's ear. Systematic research investigating the impact of source-to-listener distance on directional benefit, however, has not been completed.

The current project examines the interactive effect of changes in reverberation time and source-to-listener distance (both within and beyond the critical distance), on the benefit provided to persons with hearing loss by current digital directional hearing aids. Sentence recognition, in a relatively diffuse noise (44 dB SNR), was assessed at multiple distances from the source speaker (4, 8 and 16 feet) using a hearing aid capable of both omnidirectional and directional modes. A lecture hall (~ 550 sq. ft) was used as the test environment. Testing was completed under conditions of moderate (860 ms) and low (340 ms) reverberation. The low reverberation condition was obtained by adding acoustic blankets to the walls.

Results to date suggest that performance in both directional and omnidirectional hearing aid modes are reduced as the distance from the source-to-listener increases. In addition, the effect of source-to-listener distance appears to vary based on microphone mode, with a more negative effect of increasing distance observed in the directional microphone mode. Contrary to expectations, directional benefit provided by current microphone technology was still apparent (although greatly reduced) even for distances beyond the critical distance in each environment. Results will be discussed in terms of the interactive effects of speaker-to-listener distance, critical distance, and reverberation time on directional benefit.

Auditory localization with directional and omnidirectional microphone hearing aids

Paula Henry and Todd Ricketts, Vanderbilt University

Studies examining auditory localization in individuals with hearing impairment wearing hearing aids have shown mixed results. Some have demonstrated that the use of hearing aids has resulted in poorer localization accuracy than without hearing aids (Dermody & Byrne, 1975; Noble, Byrne & LePage, 1994). On the other hand, some studies have found no significant effect of hearing aids on localization accuracy (Haufler et al., 1983; Koelhne & Besing, 1997; Noble & Byrne, 1990; Noble, Ter-Horst & Byrne, 1995).

There were two primary aims of the current study. The first aim was to examine auditory localization in the lateral horizontal plane by listeners with impaired hearing with and without amplification. The second aim of this study was to examine the effects of different hearing aid microphone directional sensitivity patterns on the accuracy of auditory localization in the lateral horizontal plane. To date, little is known about the effects of different microphone directional sensitivity patterns of hearing aids on auditory localization ability. Recent data from our laboratory (Ricketts, Henry, & Gneuwikow, in preparation) indicated that some listeners reported differences in localization abilities on a self-report inventory between directional and omnidirectional microphone modes.

Participants were evaluated in three listening conditions: unaided, aided with a pair of behind-the-ear (BTE) hearing aids utilizing omnidirectional microphones, and aided with a pair of behind-the-ear (BTE) hearing aids utilizing directional microphones. The stimulus used was a 200 ms burst of 4000 Hz narrow band noise. This stimulus was randomly presented from a subset of an array of 45 loudspeakers located in the lateral horizontal plane. Participants responded to the perceived location of the sound by calling out a number corresponding to one of the loudspeakers.

Results across the three listening conditions will be discussed with a focus on the differences in localization accuracy between the omnidirectional and directional microphone conditions.

Speech recognition performance and speech quality ratings in asymmetric listening conditions with adaptive directional microphone hearing instruments

Emma Payne and Mark Lutman, University of Southampton, UK

While the benefits of directional microphone technology in improving speech intelligibility in noise are well documented, potential benefits of adaptive directional systems are less well understood, especially for asymmetric listening conditions.
Adaptive directional instruments may be configured to vary their properties according to an estimate of signal-to-noise ratio, utilizing an omnidirectional setting during low-noise periods. For binaural users of two adaptive directional instruments that operate independently of one another, there is the potential for conflicting settings in asymmetric listening environments. This may affect speech recognition and/or sound quality. The aim of the present study was to assess speech recognition performance and speech quality in a variety of asymmetrical noise backgrounds for binaural users of hearing instruments set to combinations of adaptive directional, fixed directional and omnidirectional modes.

Sixteen experienced hearing aid users with moderate symmetrical sensorineural hearing impairment were fitted with bilateral Phonak Claro BTE instruments. After a period of fine-tuning and familiarization, they performed speech recognition testing and speech quality ratings for a range of noise and hearing aid conditions. BKB sentences were presented at a level of 65 dB(A) in an anechoic laboratory always from a single loudspeaker at 0° azimuth. Speech weighted noise was presented from azimuth angles: 0°, 180°, (90°±270°), (120°±190°), (170°±240°) and was varied adaptively to approach the asymptotic noise level that gives a score of 71% correct. For each of the five noise conditions the hearing instruments were set to each of five settings: both omnidirectional, both fixed directional, both adaptive directional, omnidirectional-L/adaptive-R and adaptive-L/omnidirectional-R. For each of the 25 conditions, subjects also rated the quality of running speech in noise in terms of overall comfort, loudness of noise, loudness of speech and clarity of speech.

Results showed improved speech recognition performance with adaptive and fixed directional microphones relative to omnidirectional in all conditions where there was an opportunity to exploit directional characteristics. In these conditions, the adaptive setting produced results that were no different from normally hearing subjects. Performance for the fixed directional system was poorer than the adaptive system in some conditions. For all speaker conditions, the asymmetric hearing aid settings did not confer any particular disadvantage for speech recognition, always giving scores significantly better than the omnidirectional setting and in some instances better than the fixed condition. The results from the quality rating were similar to the speech recognition data. For all speaker conditions where directional characteristics can be utilized, the adaptive and fixed hearing aid settings were rated as significantly clearer and more comfortable than the omnidirectional and asymmetric conditions. The asymmetric settings did not appear to confer any particular disadvantage for listening comfort and quality. It is concluded that binaural adaptive directional instruments provide advantages for speech recognition in noise without disadvantages for speech quality, even when the adaptive systems run independently on the two instruments.

[This work was supported by Phonak Hearing Systems]

Finding the correct methodological procedure in estimating the directivity index for current directional microphone technology in hearing aids

Andrew Ditteberner and Ruth Bentler, University of Iowa

Unwanted noise is a major problem for most users of hearing aids. A number of strategies, particularly the use of directional microphone technology, have increased in popularity in recent years. However, when describing and comparing the directional characteristics between different types of directional microphone systems, there have been an abundance of confusing and outright incorrect claims regarding this technology. These claims stem from the misuse of the Directivity Index (DI). In this study, we examined and compared current methods used to estimate DI as a function of frequency. We also evaluated the perceptual relevance of DI as an estimator of signal-to-noise ratio (SNR) benefit for normal hearing and hearing impaired population. It was found that conventional methods of calculating DI from polar data were incorrect when the technology was evaluated with a manikin. Also, higher spatial sampling was required to obtaining a reliable summation approximation of the off-axis response of a microphone, an important component of the DI measure. Finally, preliminary findings have found the DI measure to be a good predictor of SNR benefit for normal hearing and hearing impaired individuals in free field environments with isotropic background noise. Problems were found when using the DI to predict performance in environments with reverberant characteristics and anisotropic background noise.
Plasticity and Acclimatization

Commercial acclimatization to amplified speech in adults
Kevin Munro and Mark Lutman, University of Southampton, UK

With the provision of amplification, the listener receives newly available speech cues that were previously inaudible. In addition, cues that were previously audible are now received at a higher intensity. These changes may immediately confer greater intelligibility. However, there may be consequential neural reorganization resulting in an improvement in performance over time (perceptual learning or auditory acclimatization). Some studies measuring speech recognition, intensity discrimination, and loudness perception have confirmed the existence of auditory acclimatization while others have failed to demonstrate an acclimatization effect. This suggests that the acclimatization effects may be small or non-existent. Previous studies in our laboratory (Munro and Lutman, IHCON 2000) did not show any clear evidence of acclimatization despite subjects making regular use of hearing instruments that substantially improved audibility. The conflicting findings suggest that acclimatization may only be apparent under specific test conditions. It was hypothesized that the amplified level of speech used in our previous work was not sufficiently different from that experienced in everyday life, prior to aiding, to induce perceptual learning.

The present study addressed this hypothesis by using speech presented at 55, 62 and 69 dB SPL. Sixteen subjects with moderate bilateral sensorineural hearing impairments were fitted monaurally with identical linear hearing instruments that provided approximately 20 dB real ear insertion gain at 2-4 kHz. The self-reported use of the hearing instrument was typically 8-12 hours per day. The main outcome measure was the Four Alternative Auditory Feature word recognition test. Subjects were tested at six-week intervals over a post-fitting period of 12 weeks. Acclimatization was defined as an improvement over time in recognition score in the aided ear relative to the unaided ear. The results revealed a significant improvement in performance over time, predominantly at the highest presentation level (mean increase in benefit at 69 dB SPL was 6%). It is concluded that, after provision of a hearing instrument, the auditory cortex reallocates resources in the intensity domain. This results in an increased cortical representation of behaviorally important speech sounds that are heard at a higher intensity than previously. Studies must use an amplified speech signal that is more intense than commonly experienced in everyday life prior to aiding, in order to show acclimatization. This finding improves our understanding of the conditions required to measure acclimatization and should result in a more robust methodological framework for future studies.

[This work was supported by Department of Health]

Plasticity in auditory Cortex after sensory experience with spatiotemporal input patterns
Pritesh Pandya, Raluca Moucha, Navzer Engineer, Daneil Rathbun, and Michael Kilgarr, University of Texas at Dallas

Compelling evidence exists that the adult brain can be modified by sensory experiences. This experience-dependent plasticity has implications for clinical remediation strategies in a vast majority of auditory disorders. Neural plasticity mechanisms are involved in hearing loss, tinnitus, cochlear implants, the development of speech and language, and many other auditory system impairments. It is now well-established that features of the sensory input provide much of the information to direct plasticity in the primary auditory cortex (A1). Understanding how auditory input can guide cortical plasticity will help in developing techniques that maximize recovery from many debilitating neurological conditions, including hearing loss.

One of the ongoing objectives of our lab is to characterize the spectrototemporal discharge pattern of spatially distributed neuronal populations in A1 with complex stimuli in naive rats. Complex sounds such as speech and animal vocalizations contain spectral and temporal modulations that are perceptually important. In this study, the temporal and spectral characteristics of neural representations of vocalization sounds were studied in neuronal populations of A1 in adult rats. Population PSTHs, neurograms, & direction selectivity were computed. Consistent with findings in other species, the spectrototemporal discharge patterns of A1 neurons appears to be related to the spatiotemporal acoustic input pattern.

Our lab has also been conducting a series of studies to investigate how the distributed cortical responses to complex stimuli change after long-term sensory experience. Specifically, we are evaluating how sensory experience alters the distributed cortical response by engaging neural plasticity mechanisms...
triggered by activation of nucleus basalis neurons located in the basal forebrain. Vocalization sounds were repeatedly paired with electrical activation of the basal forebrain – 300 times a day for one month in 13 rats. After extensive experience with vocalizations, population analysis suggests a temporal and spatial sharpening of the distributed cortical responses. We also observed a high frequency cortical map expansion when human vocalizations were paired. Interestingly, this map reorganization was blocked by unpaired background sounds suggesting that acoustic context contributes to cortical plasticity. Our preliminary results support a learning hypothesis in which intensive and focused exposure to complex acoustic signals can increase the fidelity of neuronal representations and generate a more temporally coordinated distributed neuronal response.

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PA27

An animal model of cortical reorganization in deafness and its remediation
Marcia W. Raggio, San Francisco State University

Primary auditory cortex (AI) of cats, monkeys and, most likely, humans shows a continuous gradient of frequencies along one cortical axis and a discontinuous, modular organization along the orthogonal axis. The modular organization is related to spectral, intensity, and binaural processing. This organization is also revealed using peripheral electrical stimulation. It has been found that after a short duration of profound deafness, the organization is maintained, however, longer duration of unremediated deafness results in a degradation of the cochleotopic gradient as well as the functional modularity of cat AI. Using conventional amplification in an animal model is problematic. Therefore, this deafness/electrical stimulation model may be useful in revealing physiological consequences of remediated hearing loss in terms of the central representation of sound. Known studies of monaural vs binaural hearing aid fitting and word recognition suggests significant plasticity in the adult auditory system with amplification. Reorganization findings of the auditory cortex using electrical stimulation will shed light on the nature and time course of improvements demonstrated in hearing aid patients.

Chronic electrical stimulation in deafened animals can further change the cortical organization resulting in additional loss or in restoration of some aspects of normal cortical organization. The different outcomes depend on several aspects, including stimulation properties and behavioral relevance of the input. Implications for the prevention or reduction of central auditory consequences of hearing loss are discussed.

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Similar organizational changes would be expected with hearing loss, as well as with the application of amplification.
We compared the effectiveness of three procedures for fitting multi-channel compression hearing aids: (1) the Cambridge procedure for loudness equalisation (CAMEQ), which aims to amplify speech so as to give equal loudness per critical band over the frequency range 500 to 5000 Hz, and to give similar overall loudness to "normal"; (2) the Cambridge procedure for loudness restoration (CAMREST), which aims to amplify speech so as to restore "normal" specific loudness patterns, over a wide range of speech levels; (3) the Desired Scensation Level Input/Output rule (DSL I/O), which aims to map the dynamic range of normally hearing people into the reduced dynamic range of hearing-impaired people, with "full" restoration of audibility. Forty subjects were fitted with Danalogic multi-channel compression digital hearing aids, using each of the three fitting procedures in turn; 20 were fitted bilaterally and 20 were fitted unilaterally. In each group of 20 subjects, 10 were experienced hearing aid wearers and 10 were inexperienced wearers. Prescribed insertion gains were verified using real-ear measurements. Immediately after fitting with a given procedure, and one week after fitting, the gains were adjusted, when required, by the minimum amount necessary to achieve acceptable fittings. On average, the adjustments were smallest for the CAMEQ procedure, slightly larger for the CAMREST procedure and were largest of all for DSL I/O. For the latter, the gain changes were mostly negative, especially for high frequencies and the higher input level. Gain changes were largest for the inexperienced group, and always in a negative direction. After these gain adjustments, users wore the aids for at least three weeks before speech reception thresholds (SRTs) for sentences in quiet and in steady and fluctuating background noise were measured, and the Abbreviated Profile of Hearing Aid Benefit (APHAB) questionnaire was administered. The scores on the APHAB test and the SRTs did not differ significantly for the three procedures. SRTs were slightly better for the inexperienced group than for the experienced group. However, this was most likely due to the smaller hearing loss for the former group. We conclude that the CAMEQ and CAMREST procedures provide more appropriate initial fittings than DSL I/O for both experienced and inexperienced hearing aid wearers, fitted bilaterally and unilaterally.
The spectral characteristics of children's speech are known to differ from those of adults in a number of ways. In general, children's speech is lower in overall amplitude and higher in fundamental frequency. In addition to adult-child differences, spectral differences have been reported for different microphone locations around the talker. Specifically, the frequency response of a talker's own voice at 0° azimuth and at the ear (simulating the position of a hearing-aid microphone) has been found to differ substantially. Because the development of normal voice and speech production in children is, in part, dependent on monitoring one's own voice, the spectral characteristics of speech at the ear are an important consideration when fitting hearing aids to prelingually hearing-impaired infants and children. In part I of this study, the spectral characteristics of speech recorded at the ear and at 0° azimuth (reference position) were compared for adult-child differences as well as microphone location differences. Simultaneous recordings of short sentences produced by 20 adults and 25 children (ages 2-4) were made at 0° azimuth and at the ear. Preliminary results revealed lower overall amplitude for the children's speech and a loss of high-frequency energy at the ear relative to the reference position for both children and adults. In part II, the influence of various signal-processing strategies was examined to determine which scheme optimized audibility of low-level, high-frequency speech sounds. Differences in the spectral characteristics of running speech and selected phonemes will be discussed.

[Supported by NIDCD]

PB4

The impact of hearing aid signal processing on novel-word learning in children with moderate-to-severe hearing loss
Sheila Pratt Catherine Palmer, Judith Grayhack, Lisa Taubman, University of Pittsburgh, Diane Sabo and Julie Hauser, Children's Hospital of Pittsburgh

Little is known about how children function with different types of hearing aids. Of particular concern is that the signal processing schemes in hearing aids have been developed without consideration for the auditory, linguistic and cognitive limitations of infants and children. Hearing aids have great potential to impact children's abilities to develop auditory-based speech and language. A consistent finding is that children with hearing loss have reduced vocabularies and lexical skill development which may directly link to their abilities to parse ongoing speech into words, map them to their lexicons, and produce them as speech (Davis, Elfenbein, Schum & Benliner, 1986; White & White, 1987). To assess the notion that hearing aid processing schemes may impact children's ability to learn and access novel lexical-items, the following study was implemented.

Six children between the ages of five and seven years with moderate-to-severe hearing losses acted as subjects in this study. All of the children had language, speech, auditory memory, speech recognition, and visual processing skills within normal limits. All previously wore linear binaural amplification successfully.

A double blind single-subject treatment-design was employed using multiple baselines and replication across the six children. The children wore open-platform laboratory hearing aids that were programmed to conform to three different configurations: a single-channel linear configuration with compression limiting, a single-channel WDRC configuration, and a two-channel configuration consistent with WDRC below and adaptive compression above 1500 Hz. The order that the children wore the different configurations was counterbalanced across the children and randomly assigned.

The children were exposed to novel words in a computer-controlled memory game with sessions occurring twice a week. The novel words had a CVC syllable structure and were arranged in three lists matched for phonetic composition and degree of novelty. The words were presented in sound field at a distance of 18 inches from the children's head at presentation levels that varied randomly from 45 to 85 dB SPL. A variable presentation level was used in order to simulate real-world presentation levels and to stress the distinguishing characteristics of the three hearing aid configurations. During the memory game each novel word was associated with an image of a novel object or an unusual color (also balanced for novelty) presented on a computer monitor. Initially the words were presented as an incidental learning task but it proved too difficult for all of the children, so the task was modified to include knowledge of results. Delayed recall of the words, which was tested at each subsequent session, was used as the primary measure of learning with the criterion set at five out of eight words correctly recalled for two consecutive sessions. All of the children were limited to 11 treatment sessions per treatment phase. Immediate recall, and vocal reaction time also were used as measures of learning and lexical access. In addition, speech-error data were recorded as part of the
treatment sessions. As secondary tasks, the perception of voice-onset-time and F2 loci in synthesized words was assessed at the beginning and ending of each treatment phase with a word identification task.

All of the children exhibited a treatment effect and maintenance with at least one hearing aid configuration. The patterns of delayed recall per configuration were not consistent across all children, but based on the slopes of the delayed recall learning functions, a comparison of ranks using an Exact Friedman statistic indicated a significant difference across the configurations (FR(x) = 6.87, p = .03, df = 2). The slopes of the single-channel configurations were ranked as better than the two-channel configuration. The single-channel configurations were not different although the WDRC configuration had more first place ranks than the linear configuration. The children with better performance on the speech perception tasks had greater learning effects as measured by the slope and extent of the learning functions. The reaction-time data corresponded well with the recall data. In addition, the speech-error data recorded as part of the training sessions suggested hearing aid configuration differences in that more fricative and voicing distortions, as well as intrusive stopping, were noted for the hearing aid configurations with which the individual children had had the least success.

PB6
Cortical evoked auditory potentials to evaluate hearing aid fittings in young infants
Suzanne Purdy, Harvey Dillon, Richard Katsch, Lydia Storey, Mridula Sharma, National Acoustics Laboratory, Australia

With the advent of universal newborn hearing screening there is increasing need for objective techniques to evaluate hearing aid performance in young infants. Electrophysiological techniques that have been used for objective hearing aid evaluation include the auditory brainstem response (ABR), steady state evoked potentials (SSEP) and obligatory cortical auditory evoked potentials (CAEP). Cortical evoked potentials have some advantages over other electrophysiological procedures for hearing aid evaluation as they can be elicited using speech stimuli, the stimulus duration is longer than that typically used for ABR testing so compression circuitry is activated, and cortical responses correlate well with auditory perception. Cortical potentials can be reliably recorded in awake infants. Our results for infants aged 3-7 months who have normal hearing show that the CAEP recorded using a conventional vertex-mastoid electrode montage consists primarily of a large P1 peak. Within each subject, the evoked response differs reliably in shape across the five speech and non-speech stimuli used. Aided cortical evoked potentials have been recorded in infants and children with moderate to profound hearing loss whose hearing aids were fitted based on the NAL-RP (linear) and NAL-NL1 (non-linear) prescriptive methods. CAEP results in these children are consistent with the degree of hearing loss and hearing

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aid characteristics. Changes in CAEP characteristics occur with changes in hearing levels and/or hearing aid adjustments. The experimental aim (data are still being collected) is to examine whether hearing-impaired children wearing individually prescribed hearing aids also have evoked cortical potentials that differ reliably across the set of stimuli. If so, the technique may provide an objective test of whether amplification is adequate to enable selected stimuli to be detected and differentiated.

**Outcomes**

**PB7**

A new method for evaluating hearing aid outcomes in the real world.

Arlene Neuman, The Graduate Center, City University of New York

A user-convenient field evaluation unit has been developed in order to investigate methods of improving the reliability and validity of outcome measures obtained under real-world listening conditions. This unit will facilitate the simultaneous collection of digital stereo recordings of the sound environment and a record of the subject’s rating of the sound quality through the hearing aid. A Xybernaut MA IV Basic “wearable” computer is the heart of the system. We are using a Measurement Computing DAT16/16AO PCMCI A data acquisition card and Knowles FG-3329 hearing aid microphones with a flat frequency response as the input to the computer. The computer has a 200 MHz Pentium MMX processor, 64 MB of RAM and a 12 GB hard drive and runs on a Windows 98 operating system. The wearable computer runs on a rechargeable battery that lasts for approximately six hours per charge. The computer is small (4.6” x 7.5” x 2.5”) and light (2.90 lbs). The subject wears the computer in a pack/belt combination. The 6” diagonal Xybernaut touch-sensitive flat-panel display may be worn on the belt or on the user’s wrist.

Custom-designed data collection software (written in National Instruments’ LabVIEW programming language) consists of programs for digitizing and saving the output of the binaural head-worn microphones and for the collection of subjective rating data. The rating data are obtained using the touch screen.

The field evaluation unit should allow control over analysis of subjective data by providing an objective record of the acoustic environment in which the subjective data were collected. This type of procedure should be useful for single memory hearing aids with conventional processing, multi-memory hearing aids, directional microphone hearing aids, and signal processing hearing aids. The field evaluation unit and its potential uses will be described.

**PB8**

Utility approach to measuring hearing aid outcomes

Harvey Abrams, Theresa Hnath-Chisolm and Maura Kenworthy, VA Medical Center and University of South Florida

Considerable attention has been paid in recent years to documenting the benefits of audiological intervention, particularly those outcomes associated with amplification. While research and clinical audiologists have at their disposal an array of disease-specific measures (e.g. HHIE, APHAB, COSI) that can be used to measure hearing aid benefit, the focus has shifted recently to demonstrating benefit as measured by generic, health related quality of life measures. Unfortunately, such standardized generic instruments as the widely used MOS-SF36 have been resistant to revealing the improved quality of life that can be achieved with properly selected and fit hearing instruments. Our laboratory has been experimenting with a measure traditionally used by healthcare economists to determine a population’s health state preference for the purposes of healthcare policy and planning - a concept known as utility - to measure quality of life benefits achieved with amplification. In this presentation, we will define and describe utility as used by healthcare economists, examine how utilities are measured in healthcare research in general and audiological research in particular and present the results of a recently completed study that examined if hearing aid use would result in improved quality of life as measured by utilities.

In our study, utilities were obtained using the U-Tier II, an interactive software program designed to measure an individual’s health state preference or utility. This study also examined the issue of numeracy, which is described as an understanding of basic probability, and its effect on an individual’s ability to accurately complete utilities. Data from 54 individuals fit with hearing aids in this randomized, controlled, pre-test/post-test experimental design study were analyzed. The participants completed the U-Tier II, a test of numeracy and the International Outcome Inventory for Hearing Aids (IOI-HA). Three utility approaches were used: standard gamble (SG), time trade-off (TTO) and rating scale (RS).
The results of the investigation were as follows:

1. Statistically significant differences between pre- and post-intervention utility scores were measured using the TTO and RS approaches, suggesting that the effects of hearing aid intervention on health-related quality of life can be measured using a utility approach.

2. Our subjects' understanding of basic probability (numeracy) was a factor in their ability to successfully complete utility measures.

3. IOI-HA total scores were significantly correlated with utility outcomes as measured by the TTO and RS approaches.

PB9

Connected discourse tracking as a hearing aid evaluation measure
Jill Preminger, University of Louisville School of Medicine, Department of Surgery

Researchers and clinicians continue to search for sensitive and reliable field study evaluation techniques which measure hearing aid benefit. Unfortunately it is extremely difficult to measure hearing aid benefit in the real world. The changing listening environment makes it difficult to determine whether differences in judgments are due to changes in hearing aid processing, changes in attitude, or simply changes in the environment. Objective laboratory tests are useful because performance can be measured under carefully controlled conditions. This allows for reliable measures. Unfortunately, laboratory measures are not always good predictors of field study results. New laboratory tests are needed which can simulate real world listening in the more reliable laboratory environment.

Connected discourse tracking is a method which allows for direct observation of the hearing aid user, in a controlled laboratory environment. With this procedure, a speaker reads from a prepared text, while a receiver must repeat everything the speaker says with 100% accuracy. The tracking procedure allows for the evaluation of successful and unsuccessful communication strategies, as well as overall communication abilities. Connected discourse tracking has been used extensively in the evaluation of cochlear implant performance, but the use of this procedure has not been reported with hearing aid users. The purpose of this project was to design a tracking procedure which could be used reliably in the laboratory in order to evaluate hearing aid performance.

In an initial experiment, eight listeners with hearing loss wore an experimental hearing aid which was programmed with two different frequency gain characteristics (FGCs). Listeners performed significantly better on an objective speech perception test with the 'Good' FGC as compared to a 'Poor' FGC. Tracking in words per minute was measured for each FGC using magazine articles as stimuli in an auditory only condition. As a group, subjects were able to track significantly more words per minute for the 'Good' FGC as compared to the 'Poor' FGC. But, the individual results only showed significant differences between the two FGCs for three of the eight subjects.

A second study was completed with normal hearing listeners in order to improve the sensitivity and the reliability of the tracking procedure. The text from the Speech Intelligibility Rating Test was used; these materials have been shown to be equivalent for auditory only presentation. (In a pre-recorded format.) Both individual and group performance was analyzed in consideration of FGC condition, text passage, and text order.

[This work was supported by the Mary and Mason Rudd Surgical Research Fund, Jewish Hospital Foundation, Louisville, KY]

PB10

Associations between personality and self-report hearing aid outcome for subjects who receive free hearing aids.
Robyn Cox, Genevieve Alexander, University of Memphis and Memphis VAMC, Ginger Gray, University of Memphis

Self-report outcome data provide a measure of the daily life impact of a hearing aid fitting from the client's point of view. For the most part, we tend to assume that self-report outcome data primarily reflect the real world effectiveness of the hearing aid and the efficacy of the fitting strategy as applied to the individual patient. There is relatively little research that assesses the validity of this assumption, or the extent to which other variables might influence subjective outcomes of hearing aid fitting. However, a few studies suggest that personality attributes, such as extraversion and anxiety, can account for at least 10-20% of the variance in self-assessments of hearing aid outcome. To put this modest relationship in perspective, it is helpful to note that aspects of personality have been found to be generally more effective than audiogram-based hearing impairment or any laboratory speech understanding test in accounting for variance in subjective benefit or satisfaction. In previous research on this topic, it has not been possible to statistically account for differences in the amplification used by subjects. In this paper, we will present results of ongoing research that accounts for hearing aid fitting and other variables in a design that will provide a more accurate estimate.
of the extent to which personality is predictive of hearing aid fitting outcomes.

The subjects are 140 hearing-impaired veterans (all men) aged 60 or more who have been fitted bilaterally with the same model of hearing aid (programmable, single channel, single memory, analog, WDRC). Before the fitting, all subjects complete a comprehensive personality inventory (NEO-FFI) and estimates of locus of control and coping strategies. Self-report outcomes are measured at 3 weeks, 3 months, and 6 months post-fitting. Outcome domains include daily use, benefit, residual disablement, and satisfaction. Potential covariates include pre-fit expectations, health status, unaided disablement, and response bias. We will address the following questions:

1. How accurately can we predict the subjective outcomes of a hearing aid fitting (benefit, satisfaction, etc) based on personality and related variables?
2. How much is the prediction improved when we include objective data about hearing impairment and the fitting itself?
3. At what post-fitting interval do subjective hearing aid outcomes stabilize?

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**PB11**

Cross-language assessment of functional hearing

Sigfrid D. Soli, House Ear Institute, Chantal Laroche, University of Ottawa Canada, Lena Wong, Hong Kong University, China, Masae Shiroma, International University of Health and Welfare, Japan, Andrew J. Vermiglio, House Ear Institute, USA

The need for functional hearing assessment protocols that are standardized across languages has grown significantly with the advent of international multicenter device trials, and with an increasingly multilingual population worldwide. We are addressing this problem through the development of standardized, norm-referenced, adaptive measures of speech intelligibility in Japanese, Cantonese, Mandarin, Canadian French, and Spanish. The design of these measures is patterned after the English Hearing In Noise Test (HINT). We will present results from an ongoing international research collaboration among scientists in the United States, Japan, China, Canada, and in several Latin American countries that has as its goal the development of these measures in new languages. The methods of selecting speech materials, recording the materials, synthesizing appropriate masking noise, equating the difficulty of the recorded sentences, forming sentence lists, normalizing the lists, and determining reliability and sensitivity will be presented. Samples of the materials from each language (English, Japanese, Mandarin, Cantonese, Canadian French, and Spanish) will be used to demonstrate the procedures by which standard dialects of each language were selected. Cross-language comparisons of the spectral and temporal characteristics of the speech materials will be presented. Norms, reliability coefficients, and standard errors of measurement for each language will also be reported. Methods for comparing and/or equating functional hearing ability across languages will be described.

**PB12**

Objective sound quality assessment in hearing aids

Birger Kollmeier, Medizinische Physik, Germany

A review is given of work currently performed in our laboratory to objectively predict the subjectively assessed sound quality of different hearing aid algorithms using auditory models. The basic idea and layout of the method originates from telecommunications and audio coding: The original and the "distorted" (coded or nonlinearly processed) acoustic signal is compared on a perceptual level (i.e. at the output of an appropriate auditory model) rather than on the acoustical level for the assessment of speech coders (Hansen & Kollmeier, 2000) and audio coders (Huber & Kollmeier, 2001). In hearing aid applications, the methods are applied to model the subjective preference data across different monaural and binaural noise reduction algorithms from six hearing-impaired subjects (Marzinkz, 2001). The preference data was converted into a rank scale using the Bradley-Terry-Luce (BTL)-model. The results show a high correlation between subjective and objective quality judgments. The Log-Area-Ratio (LAR) objective measure shows the highest correlation with overall subjective preference, while the PMF measure (Hansen & Kollmeier, 2000) corresponds best with the subjectively perceived amount of noise suppression. These results indicate that such objective methods can be used as an efficient tool in hearing aid development and assessment.
Directional Microphones

PB13 Improving speech intelligibility in noise with a fixed beam array
Heleen Luts, Lab. Exp. ORL, Belgium, Wim Soede, ARDEA, The Netherlands and Jan Wouters, Lab. Exp. ORL, Belgium

Besides amplification offered by a hearing aid, hearing-impaired listeners need a better signal-to-noise ratio than normal-hearing listeners do, because of the poor speech recognition in noisy environments. A real-time fixed endfire array, which consists of bidirectional microphones, has been developed in order to ameliorate the signal-to-noise ratio. The array is meant to be used in combination with a hearing aid and depending on the acoustical situation a choice can be made between one, three or even five active microphones.

The aim of this study was to evaluate this microphone array. We carried out perceptual measurements of the speech intelligibility in a realistic environment, because this results in the most relevant information towards clinical practice and everyday life. A group of ten normal-hearing and ten hearing-impaired subjects participated in the listening tests. Sentences were presented in a reverberant room, with two types of noise and six different noise scenarios with single and multiple noise sources. Our main interest was the improvement of speech intelligibility that can be obtained with the microphone array in comparison to an omnidirectional hearing aid microphone. Therefore we measured the speech reception threshold in each condition for an external omnidirectional hearing aid microphone, one and three active array microphones.

With three microphones improvements up to 12 dB for normal-hearing and 9 dB for hearing-impaired listeners were achieved relative to an omnidirectional microphone for a single noise source at 90°. Even with a noise source at 45° benefits of 4 dB were obtained. To evaluate the advantage of two additional microphones, listening tests were carried out with a five microphone array in two noise scenarios: one noise source at 45° and three noise sources at 90°, 180° and 270°. Ten normal-hearing subjects participated. In both conditions an extra improvement of about 1 dB was obtained. These improvements in signal-to-noise ratio are very promising in view of the compensation hearing-impaired listeners may need. The array with three bidirectional microphones, named Linkit, is now commercially available.

PB14 A subband beamformer on an ultra low-power miniature DSP platform for hearing aids
Edward Chau, Hamid Sheikhzadeh, Robert Brennan, and Todd Schneider, Dspfactory Ltd.

Directional signal processing algorithms employing microphone arrays have been found to improve the intelligibility of speech in noise for many hearing aid users. While many different algorithms have been proposed for hearing aids, the constraints in size and processing power due to the arrays and hearing aids prevent many algorithms to proceed beyond the simulation stage. A practical algorithm designer for hearing aids must, therefore, weigh the constraints against the complexity of the algorithm in order to implement a successful design.

With practical implementation in mind, we present the design and implementation of a subband cardioid beamformer on an ultra low-power miniature DSP platform for hearing aids, using a 2-microphone endfire array. The focus is on implementing a working beamformer on very low-resource hearing aids, instead of on designing a beamformer that may be theoretically superior but impractical to implement. Taking advantage of several innovations in the design of the Toccata DSP platform and the subband beamforming algorithm, the beamformer is implemented in real-time, running on a 5 MIPS DSP core at 1.25 V.

The subband beamformer extends the classical time-domain, narrow-band algorithm to a frequency-domain, broadband implementation, so it is suitable for general speech signals. An oversampled, weighted overlap-add filterbank is used to allow wide gain and phase adjustments in the frequency-domain. A subband IIR filter is proposed to overcome the non-zero bandwidth of the frequency bands, and to introduce a nearly linear phase adjustment across the bands. The subband implementation also allows the beamformer to be integrated with additional algorithms at different frequency ranges.

A crucial benefit of the subband approach is that the beamformer can be easily enhanced to introduce variable beampatterns at different frequency bands, enabling the beamformer to simultaneously cancel multiple narrow-band signals that occupy different frequency bands. We have implemented a simple real-time demonstration on the Toccata hardware that
showcases this important benefit of the subband approach, which greatly enhances the classical 2-microphone algorithm with only minimal increase in algorithm complexity.

Finally, the subband beamformer has been evaluated with acoustic measurement made in a recording studio. It is found that, in the fixed fullband implementation, the subband beamformer performs favorably to the theoretical cardioid beamformer. With the enhancement for canceling multiple narrow-band sources, the subband beamformer outperforms the classical beamformer when multiple sources are present. In both cases, a maximum of approximately 20 dB in front-to-back gain can be observed.

PB15

Zooming in on speech: multimicrophone noise reduction strategies in hearing aids

Ian Wouters, Lab. Exp. ORL, Belgium, Jean-Baptiste Maj, Lab. Exp. ORL and ESAT-SISTA, Belgium, Marc Moonen, ESAT-SISTA, Belgium, Ann Spruyl, Lab. Exp. ORL and ESAT-SISTA, Belgium

The performance for speech understanding in noisy environments of an adaptive beamformer in a single two-microphone hearing aid is studied. This technique has been implemented in behind-the-ear hearing aids and evaluated physically and perceptually with normal hearing, hearing impaired listeners and cochlear implantees. The tests are carried out for 2 omnidirectional and 1 omni plus 1 directional microphone configurations, and with different jammer noise source scenarios (single and multiple noise sources, diffuse), different spectro-temporal jammer noise sounds (stationary and speech-modulated speech-weighted noise, multitalker babble), and tested using different speech materials. Considerable improvements are obtained relative to directional microphones.

The comparison with respect to performance and robustness of this adaptive noise reduction scheme with the adaptive zooms as introduced in some modern hearing aids, with fixed beamformers and with other more complex signal processing strategies, will be discussed. Recent hardware evolutions in hearing aids allow the application of these signal processing approaches.

PB16

On the sensitivity of two-microphone directionality to head-size changes

Tao Zhang, William Woods, Dianne van Tasell and Thomas Burns, Starkey Labs, Inc.

Multi-microphone directional processing is known to be quite sensitive to variability across design- and use-conditions. One well understood source of such variability is the match among the microphones, where the processing is typically designed assuming very close matching and efforts are made to ensure this when in use. Another source of variability is head size. Arrays are typically designed based on one head (e.g., KEMAR) or free-field conditions, but actually used on many different heads (from children to large adults). In order to quantify the effects of differences in design- and use-conditions due to head-size changes, we measured and modeled, using a rigid sphere, the responses of two microphones in aids (ETE and ITE styles) on a KEMAR mannequin 1.0m from a source in anechoic conditions. We then varied the sphere radius to simulate changes in head size. Initial results with the sphere for an array designed to maximize Directivity Index (DI) on KEMAR indicate that the frequency-averaged DI (estimated from azimuthal-plane responses) increases by 0.1-0.9 dB (depending on aid style) with a radius reduction of 40%, and decreases by 0.1-0.8 dB with a radius increase of 40%. The main reason for the larger changes is a gain maximum to the side of the sphere (ipsilateral to the aid) that tends to reduce DI when present (i.e., it contributes significant non-look-direction gain). When the sphere is reduced, the non-look direction gain is reduced, and DI can increase. Changes in DI due to suboptimal design of the array as a function of head size always tend to reduce DI, but these changes are small relative to the changes due to the non-look direction gain influence. Thus, for sphere-size reductions the DI increases because non-look direction gain due to the sphere decreases faster than directional gain decreases due to non-optimality. Due to the limited accuracy of the sphere as a model for KEMAR, we are currently working on empirical methods to extend the frequency range of and verify the model predictions.

PB17

A microphone array for hearing enhancement

Bernard Widrow, Stanford University and HearingPoint Systems, Inc.

A directional acoustic receiving system is constructed in the form of a necklace including an array of six microphones mounted on a housing supported on the
chest of a user by a conducting loop encircling the user's neck. Signal processing electronics contained in the same housing receive and combine the microphone signals in such a manner as to provide an amplified output signal which emphasizes sounds of interest arriving in a direction forward of the user. The amplified output signal drives the supporting conducting loop to produce a representative magnetic field. Hearing aids with telecoils can receive the magnetic signal and reproduce the sounds of interest. The microphone output signals are weighted(scaled) and combined to achieve directivity responses that are 20-40 dB below that of the forward direction. The weighting coefficients are determined by an optimization process. By bandpass filtering the weighted microphone signals with a set of 12 filters covering the audio range and summing the filtered signals, the receiving microphone array having a small aperture size achieves a 60 degree directivity pattern that is essentially uniform over frequency in three dimensions. This method enables the design of highly directive hearing instruments which are comfortable and convenient to use. The array provides most users with dramatic improvements in speech perception, particularly in the presence of background noise, reverberation, and feedback. Using a modified form of Dr. Sig Sohl's HINT test, patients hearing in approximately isotropic noise with their hearing aids at a recognition rate of 20% have risen to a rate of 80% when using their hearing aids with the array.

When using the array, a wearer can hear his or her own voice so clearly that those who have been deaf since birth often show easily noticed improvements in their speech patterns within an hour of wearing it. The array may prove to have applications in speech therapy.

The array has also been used as an input device to a cochlear implant processor, resulting in improvements in speech perception with the implant. This is a subject of continuing exploration.

PB18

Evaluating the use of objective metrics for assessing a binaural signal processing algorithm

Nandini Iyer and Jeffery Larsen, University of Illinois at Urbana-Champaign, Michael Kramer, Phonak AG, Charissa Lansing, Robert Bilger, Douglas Jones, William O'Brien, Bruce Wheeler, and Albert Feng University of Illinois at Urbana-Champaign

Hearing aid users often report reduced intelligibility while listening to speech in noisy environments. A number of signal processing algorithms have been developed for hearing aids to facilitate listening in noisy environments. Comparison of these algorithms can be achieved via human subject testing or the use of engineering metrics. Whereas human subject testing is the ultimate measure of effectiveness of an algorithm, it is time-consuming and costly. In contrast, engineering metrics are objective and are easily obtained. However, it is not clear which of these metrics are best correlated to human subject data. The objective of the current experiment was to determine whether a subset of commonly used engineering metrics could predict human subject performance in terms of speech intelligibility and speech quality. The investigation was done to assist parameter selection for a real-time, binaural, frequency domain adaptive beamformer algorithm known as the frequency-band minimum variance (FMV) algorithm, which is under development in our laboratory. Key algorithm parameters were initially chosen by informally listening to the processor's output. Subsequent listening tests by normal and hearing-impaired listeners showed that the algorithm improved speech intelligibility of a target in the presence of multiple interferers (Larsen et al., 2002). In this study, we adjusted the parameters of the FMV algorithm so that it was optimized for "best performance" with respect to several objective metrics, including signal-to-noise ratio, speech intelligibility and sound quality metrics. Speech intelligibility and speech quality measures were then obtained from 10 normal and 10 hearing-impaired listeners for each of the parameters. Speech intelligibility was measured using HINT sentences, CST and MRT words, all presented from four spatially separated sources of interferers at different signal-to-noise ratios (SNR). The signals were presented to listeners seated in a sound-treated room via headphones after they were processed through the FMV algorithm. Speech quality measures were obtained using a paired comparison method. Results from the experiment indicated that some objective metrics are better predictors of human subject performance than others and as such they are useful for rapid optimization of the signal processing algorithm.

PB19

Signal Processing:
Compression

PB19

Alternative Compressive Hearing Aid Algorithms Derived From Loudness Psychophysics and Cochlear Models

Julius Goldstein, Hearing Emulations, LLC, and BECS Technology, Inc. St. Louis, VA Medical
Center, Melin Oz and Peter Gilchrist, Hearing Emulations LLC, and Michael Valente, Washington University School of Medicine

Compressive hearing aid amplifiers automatically reduce their gain with increasing sound level to accommodate the reduced dynamic range resulting from sensorineural hearing loss. Known studies of the maximum sound levels tolerated by hearing impaired subjects indicate that no similar compression range exists for all listeners with similar hearing loss. Thus, Hood and Poole (1966: J. Acoust. Soc. Am. 40, 47-53) estimate a mean loudness discomfort level of ~110 dB SPL for all degrees of SNHL, while Pascoe (1989: 13th Danavox Symp., 129-151) finds a systematic mean increase in uncomfortable loudness beyond 100 dB hearing level for hearing losses exceeding 40 dB. Simplified “tip-tail” models (Goldstein, 1995: Hear. Res. 89, 52-68) of these data were derived by representing hearing loss up to 40 dB in each audio frequency band as a reduction in “compressive tip” gain, while greater hearing loss is represented by replacing the “linear tail” with an expansive function. More elaborate models of SNHL were also derived from Fletcher and Munson’s (1933: J. Acoust. Soc. Am. 3, 82-108) classical equal loudness contours to include differences in normal sound transduction for low frequencies (< 500 Hz). Clinical psychophysical study is being initiated of user preference for alternative fitting algorithms based on the SNHL models. We have developed a PC-based master hearing aid with multichannel adaptive BPNL compressive amplifiers (Goldstein, 2001: U.S. pat. appl. 09/935,510) that is programmed with alternative fitting algorithms. Subjects listen to speech processed by the master hearing aid at input sound levels of 40, 60, and 80 dB SPL, and are instructed to indicate their preference for the fitting that best represents an experience of soft, comfortable, and loud sounds. Our strategy and preliminary results will be reported.

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**PB21**

**Compressing critical bands for digital hearing aids**

Masato Hisitani, Kei Kobayashi, Kohshi Shinohara, Keiichi Yasu and Takayuki Arai, Sophia University

We discussed an algorithm for hearing aids to compensate for the frequency selectivity of hearing impairments. We focused on the wider critical band of hearing-impaired people and proposed a method of signal processing in which a speech signal is processed to reduce interference between adjacent frequency bands. Two approaches were tested; in each case, a signal was compressed toward the center of each critical band along the frequency axis.

The first approach was based on a filter bank with a set of bandpass filters. Step 1: An input signal was passed through an FIR filter at each of 21 critical bands, and transformed into the Hilbert envelope, which contains the most essential acoustic information. Step 2: The same input signal was also passed through another FIR filter at each of 21 critical bands whose bandwidth ranged from 50 up to 90%...
compared to the original bandwidth. Then the carrier component, the bandpass signal divided by its envelope, was computed. Step 3: The product of the Hilbert envelope obtained from Step 1 and the carrier component obtained from Step 2 was taken for each band. Step 4: Finally, a summation of all the outputs from the 21 bands was calculated.

The second approach was based on the fast Fourier transform (FFT). Step 1: An input signal was transformed from the time domain to the frequency domain by the FFT. Step 2: The amplitude and the phase spectra of the FFT were calculated. Step 3: The amplitude spectrum compressed toward the center of each critical band along the frequency axis was computed for each band. The compression rate ranged from 50 up to 90%. Step 4: The partially compressed amplitude spectrum was multiplied by the original phase spectrum to resynthesize a band-limited signal. Step 5: Finally, the overlap add technique was applied to the IFFT of the product from Step 4.

To evaluate the processed signals, a perceptual experiment was conducted with five severely hearing-impaired subjects. The subjects listened to three types of stimuli: the original and the processed speech by the two approaches. Then the subjects evaluated the intelligibility with a five-point scale. The result was that subjects rated best those tokens for which the compression rates were 70% for the filterbank approach and 60% for the FFT approach. We also discussed the applicability of our algorithm for a real-time processing system with a DSP.

**PB22**

Multichannel compression based on auditory filterbank
Volker Holmann and Tobias Peters,
Medizinische Physik, Universität Oldenburg,
Germany

This study investigates fast acting multichannel compression based on a nonlinear auditory filterbank. The filterbank consists of a linear, low delay analysis-synthesis system based on Gammatone filters /1/ and an additional nonlinear and invertible compensation filter /2/ that accounts for the level dependence of the auditory filter shape. In each frequency band, fast acting, broad range amplitude compression is performed in order to compensate for the recruitment phenomenon. The compression stage directly uses the Hilbert envelope for level estimation in each band. Interaction between frequency bands is introduced by the filter overlap only.

The system is evaluated by using current loudness models for hearing impaired subjects and by informal listening tests (quality comparison). The influence of the filter nonlinearity is particularly evaluated by comparing the performance of the system with and without the nonlinear compensation filter.

[This work was supported by BMBF].

**Electroacoustic Issues and Room Acoustics**

**PB23**

Comparison of two methods to generate SII values from clinical measures of frequency
Susan Scollie, and Richard Seewald, University of Western Ontario, Canada, William Cole, Stymonic Deign, Inc., Margaret Cheesman, University of Western Ontario

The Speech Intelligibility Index (SII) is a standardized method for computing a single value which represents the amount of speech that is audible and above the levels of masking and/or reverberation spectra (ANSI, 1997). An important variable in the computation of the SII is the estimation of the peaks of the speech signal. The standard SII procedure estimates these peaks as 15 dB above the long term average speech spectrum. However, Pavlovic and Studflecker (1984) demonstrated that Articulation Index values were more accurate in predicting intelligibility when the actual levels of speech peaks were used rather than the standard 15 dB estimate. In this study, we will compare the standard SII with a modified procedure which predicts the peaks of speech from measured levels of clinical test signals. Results will quantify the effects of modifying the standard procedure, as changes in SII values.

**PB24**

Objectively measured and subjectively perceived distortion in nonlinear systems
Martin Hansen, Widex Audiology Research Lab, and Åke Olofsson, Karolinska Institutet

The use of digital signal processing in hearing aids has made it possible to introduce increasingly sophisticated signal processing schemes, which change the properties of the aid depending upon the input signal. For example, amplitude compression in a hearing aid is designed to apply a time-varying and frequency dependent gain so that the level of the resulting output signal matches a desired target level. Another example is speech enhancement.
If the properties of the aid are varied too rapidly, a distortion can be heard. At present there is no standardized way to measure this distortion with realistic, broadband input signals. Olofsson has proposed a method (Olofsson, Second Biennial Hearing Aid Research and Development Conference, Bethesda, Md., 1997) for defining nonlinear distortion. This method exploited the fact, that the real and imaginary parts of the analytical signal corresponding to the input signal loose their property of being a Hilbert pair after being passed through a nonlinear system. The amount of mismatch between the real and imaginary part of the output signal quantified the degree of nonlinearity of the system and could be expressed as a signal-to-noise ratio.

We determined the subjectively and objectively measurable amount of nonlinear distortion introduced by 12 different compression systems. These systems differed with respect to the number of independent compression channels (1, 3 or 15), compression time constants (slow, fast or instantaneous), compression threshold in each channel (low or high) and the existence of a post-filtering unit for the gain signal to minimize spectral smearing. Recordings of male and female speech and of piano and guitar solo music served as input test signals to each compressor system. The recordings were made with a very low-noise microphone in order to avoid that amplified background noise would add to the nonlinear signal distortion under investigation. The subjectively perceivable amount of distortion was assessed on a rating scale by a group of 10 otologically normal subjects.

A reasonably monotonic relation between the subjective and objective measures of distortion was observed. Also, the results were in line with expectations; namely that the amount of distortion was negatively correlated with the duration of the compression time constants and with the compression threshold level, while it was positively correlated with the number of independent fast-acting compression channels.

PB25

Analysis of speech and non-speech hearing aid test signals
Rebecca Warner and Ruth Bentler, University of Iowa

When fitting hearing aids to children, non-speech test signals are often used to verify hearing aid performance. The results of the hearing aid verification are then used to predict the audibility of real speech for that child. A problem with this procedure is that some hearing aids respond quite differently to speech than they do to non-speech test signals. Several investigators (Steinachowicz et al., 1996, Scollie & Seewald, in press) have shown large differences in measured hearing aid gain between speech and non-speech input signals. In order to minimize this error, it is important to verify hearing aid performance with test signals that closely resemble speech. The purpose of this study was to compare and contrast the spectral characteristics of commercially available speech and non-speech test signals, as well as two different samples of continuous speech. All of the signals were also compared to published information regarding the long-term average speech spectrum (LTASS). Unprocessed test signals and speech were analyzed to better understand the temporal and spectral differences between the different signals. The current stage of this investigation involves characterizing how compression knee points and ratio, time constants, and channel distribution impact the spectral and temporal characteristics of these signals.

The non-speech signals analyzed included the Fonix speech-weighted composite noise, Audioscan RM500 swept and dynamic signals, the ICRA noise for a single male talker, and the ICRA noise for a single female talker. The speech signals analyzed included the four available on the Audioscan VF-1 system: DSL-calibrated speech, male speech, female speech, and child speech, as well as a passage from the speech intelligibility rating (SIR) test, and the Rainbow Passage. All speech and non-speech signals were analyzed to obtain information regarding the long- and short-term spectral characteristics. One-third octave band (1/3 OB) rms levels were obtained for each signal to provide information regarding the long-term spectrum. 1/3 OB levels were obtained every 125 ms and averaged for the duration of each signal. Maximum and minimum levels in each 1/3 OB were also obtained, in order to provide information regarding the short-term spectral characteristics, or the dynamic range of each 1/3 OB of each signal. Preliminary results indicate differences in long- and short-term spectra across the different signals, as well as differences between the analyzed signals and published information regarding the LTASS. These differences must be considered when using these signals to predict hearing aid response to real speech.
Room acoustics and hearing rehabilitation
Hans Verschuure and L. Nijs, Audiological Center Erasmus MC Rotterdam, the Netherlands
Dept. Architecture, Technical University Delft, the Netherlands

Normal use of hearing aids is in a reverberant and noisy situation. Hearing aids tend to amplify all sounds wanted and unwanted. Because most patients have poor signal-to-noise ratios for understanding speech this puts restrictions on the conditions under which users of hearing aids can understand speech, particularly if reverberation is involved. Older people and people with handicaps often live in conditions of large rooms with little acoustic damping for hygienic reasons. We seem to prescribe them hearing aids without realizing the limitations under which they can benefit from a hearing aid. As a result we often get unsatisfied users blaming the hearing aid for poor performance while in reality the acoustical environment makes it impossible for them to understand speech.

Results of clinical speech-in-noise testing can be translated into critical speech transmission scores. They can be directly translated into minimum signal-to-noise ratios and to maximum reverberation times. The presentation will show the limitations of reverberation in sound demonstrations and present the way the limitations in background noise and reverberation can be calculated. It is necessary to distinguish between the near field and the far field.
The described method is valid in the far field, which under normal circumstances and room dimensions is valid at a distance between speaker and listener of over about 2 meters.

We show examples of assessing the acoustical quality of living conditions, as they exist in homes and working environments for people with a learning disability. In this group the number of hearing-impaired persons is quite high. It shows that many of them can benefit only little from conventional hearing aids; the use of effective directional microphones is rather limited in this group because of the handling problems. The conclusion is that the acoustics of their living and working quarters should be improved before hearing aids are provided.
Fittings and Features

**PCI**

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 15.2%. 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 21 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.

**PC2**

What future does prescriptive fitting have in a climate of technological advance?

Graham Naylor, Oticon Research Centre ‘Eriksholm’, Denmark

The current explosive growth in technological complexity and flexibility of hearing aid signal processing is accompanied by a striking lack of guidance regarding the appropriate application of diverse features for the benefit of individual hearing aid clients. There is a tacit implication that because a given feature has demonstrably beneficial effects for some clients, it has no disbenefits for anybody. However, it is becoming clear that what is most ‘sophisticated’ is not always most beneficial for a given client.

This presentation addresses this problem first by putting forward a structured view of prescriptive fitting. An integrated fitting scheme to arrive at optimum final signal processing settings consists of three components: (1) the Rationale (what we want to achieve, e.g., re-establish normal loudness for conversational speech), (ii) the Prescription (how to do that, e.g., gain rules relating to audiometric data), (iii) Finetuning procedures (ways to migrate the setting towards a de facto acceptable endpoint). Present-day fitting schemes specify at most only the first and second elements.

General-purpose fitting schemes do not distinguish between categories of user as far as the Rationale is concerned, although the Prescription may be varied for e.g., steep or profound losses. Whether or not it was the original intention, the popular general-purpose fitting schemes have come to be regarded as suitable for a very wide range of clients, and the question of whether the Rationale is relevant to a given client scarcely arises. Targeted fitting schemes on the other hand distinguish between clients on the basis of their personal characteristics and supposed or assessed consequent needs. They include statements regarding candidacy, thus defining the target group.

Examples will be given of both types of fitting scheme. It will be seen that general-purpose schemes are really targeted schemes with a broad target group, in which the Rationale defines broad aims which would be clearly inappropriate for very few clients.

A fitting scheme whose Rationale and Prescription address a limited part of the client population has the possibility to achieve significantly greater benefits for
its targeted clients than a broadly-aimed fitting scheme can for those same clients. Likewise, for the wrong group of clients, a targeted scheme may lead to markedly greater dissatisfaction than a general-purpose scheme. These principles will be illustrated with data from the recent study by Gatehouse et al. (IHCON 2000).

Considering this analysis in the context of practical fitting procedures, the following points will be elucidated:
1. Why the individual cases of greatest benefit will result from use of targeted schemes
2. Why general-purpose schemes nevertheless will continue to be successful
3. How new targeted schemes can be generated
4. How intelligent choice of targeted fitting scheme for individual clients is made possible
5. Why different fitting schemes may be categorically dissimilar, and the implications of this for the optimal fitting process, including Finetuning

**PC3**

Prescription of gain for previous users of linear and non-linear amplification
Steen Ø Olsen and Ole Dyrlund Jensen, 
GN Resound Clinical Research

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.

**PC4**

User preference fitting: challenges to common belief in the industry
Thomas Lunner, Joachim Neumann, Thomas Svertsen, Oticon Research Centre ‘Eriksholm’, Denmark

Finding the gain adjustments that perfectly suit the individual hearing aid user is not trivial. Typical fitting rationales and rules on how to carry out fine-tuning are often based on average data and common belief. In recent studies, hearing impaired subjects ($n=160$) were provided with tools that allowed them to adjust the gain of a hearing instrument to their own preference in several listening situations. These adjustments are quite reproducible (< 5 dB test-retest variance). However, the individual preferred gain settings differ considerably from the gain of the subjects’ own hearing aids, and spread greatly around known fitting rationales. Furthermore, some remarkable aspects that differ from traditional understanding of hearing instrument fitting were observed:

The subjects' adjustments simultaneously improve speech intelligibility and listening comfort (in comparison to their own well-fitted modern aids). In more than half of the cases, the preferred gain diverges by more than 10 dB from the gain in the subjects own fine-tuned aids. These results will be discussed.

**PC5**

A new view on fitting asymmetrical hearing losses
Joachim Neumann, Thomas Svertsen, Thomas Lunner, Claus Nielsen, Oticon Research Centre 'Eriksholm', Denmark, Marie Öberg and Gunilla Wästöm, University Hospital, Sweden
The rehabilitation of individuals with an asymmetrical hearing loss is not trivial. One often-employed approach is based on balancing loudness between the two ears. This study investigates the ability of subjects with an asymmetrical hearing loss to adjust the gain of their two instruments themselves in a range of listening situations. The results show: (a) the hearing impaired were able to fit both hearing instruments with the tools provided, (b) their gain adjustments are only partly determined by the goal of a loudness balance and (c) they show a benefit in comparison to their own instruments from wearing an experimental hearing instrument providing the self-adjusted gain. Their subjective benefit is observable in a questionnaire and their objective benefit is observable in a speech-in-noise test. Especially subjects performing poorly with their own aids show considerable benefit from the self-adjusted gain.

PC6
Description, rationale, and efficacy of a digital, non-custom, instant-fit CIC hearing aid.
Robert Ghent, Victor Bray and Patrick Murphy, Sonic Innovations, Inc.

A non-custom, instant-fit hearing aid has been developed with the goal of delivering improved speech recognition in quiet and in noise using advanced digital signal processing while providing economic advantages for the manufacturer, the dispensing channel, and hearing-impaired individuals. The product concept is an off-the-shelf, field-programmable, hearing aid that fits in the ear canal. Product design features will be discussed along with the recommended instant-fit protocol. Data from two clinical sites will be presented that show benefit on objective measures (speech recognition performance on the HINT in quiet and in noise, probe microphone measures, and coupler target matching) and subjective measures (APHAB). Also included are data from an ear canal dimensional study that evaluated goodness of physical fit using non-custom and custom ear tips for the product.

Outcomes

PC7
The effect of acceptance of background noise on hearing aid outcome
Anna Nabelek, Samuel Burchfield, Joanna Webster, Mindy Freyaldenhoven Department of Audiology and Speech Pathology, University of Tennessee, Knoxville

Adult patients with similar histories, audiograms, speech perception scores, and receptive communication experience dramatically different outcome with hearing aids. Some become highly successful hearing aid users while others experience only limited success or reject their hearing aids altogether. In clinical populations, patients with unsuccessful outcome report problems posed by background noise. However, speech understanding in background noise is a poor predictor of hearing aid outcome. Because of this, a new procedure for assessing the effect of background noise on hearing aid outcome is being investigated. In this procedure the amount of background noise subjects are willing to accept when listening to speech is assessed. Acceptance of background noise will be analyzed as a predictor of eventual hearing aid outcome in a large group of subjects.

To determine acceptance of background noise listeners first select their most comfortable listening level for a story and then select the maximum level of background noise that they are willing to accept while listening to and following the words of the story. The difference between the most comfortable level and the acceptable level of background noise in dB is referred to as the acceptable noise level (ANL). ANL scores, with and without hearing aids, are being determined in adult subject when they first receive binaural hearing aids and at two and three months post fitting. ANL scores are being compared with subjective impressions of benefit (APHAB), report of wearing aid use (locally constructed questionnaire), and speech perception in background noise (SPIN) test.

Preliminary results indicate that ANL scores range from 5 to 20 dB and that individuals are capable of making reliable judgments about their acceptance of background noise. It appears that individuals who accept higher levels of background noise (small ANL scores) are more successful hearing aid users than the individuals who accept little noise (larger ANL scores). Judgments of acceptable background noise do not appear to be related to the type of background noise, the subjective preference of background noise, gender, age, hearing status, or to listening with or without hearing aids. Results for about 30 subjects will be reported.
[This work is being supported by NIDCD].
Sensitivity of self-assessment questionnaires to differences in hearing aid technology
Pamela Souza University of Washington, Seattle, Bevan Yuen VA Medical Center, Seattle, Jennifer McDowell, Margaret Collins, and Carl Loovis, VA Medical Center, Seattle

Subjective assessment instruments have become a popular tool for measuring hearing aid benefit. These questionnaires are designed to measure overall improvements in communication or patient satisfaction and are most often used to compare unaided versus aided responses. With the availability of new amplification technologies, an important issue is whether such questionnaires can distinguish relative differences in amplification benefit.

In this study, several established instruments, including the Abbreviated Profile of Hearing Aid Benefit (APHAB), the Hearing Attitudes in Rehabilitation Questionnaire (HARQ) and the Hearing Handicap Inventory for the Elderly (HHIE) were compared to a newly developed questionnaire designed to assess patient responses to specific hearing aid technologies. This questionnaire, the Effectiveness of Auditory Rehabilitation (EAR) consists of two modules. Questions in the first module cover intrinsic issues surrounding hearing loss, including the functional, emotional, and social effects of hearing impairment as well as quality of life. Questions in the second module relate to extrinsic factors such as comfort, convenience, and cosmetic appearance of the hearing aid.

Subjects were 75 listeners aged 35-94 years with hearing losses ranging from mild to severe and were fit with binaural hearing aids in a standard clinical setting. For each subject, data included a set of baseline questionnaires obtained approximately one month prior to receipt of the hearing aid, a set of questionnaires completed three months after the hearing aid fitting, and measures of overall satisfaction and adherence. Test-retest reliability was also measured for each questionnaire. Specific amplification features under study included the type of hearing aid (analog, programmable, digital); aid style (BTE, ITE, ITC, CIC); circuit type (linear, WDRC); and a number of specific features (directional microphone, multiple memories).

The type of hearing aid had little effect on overall communication ability; however, programmable and digital aids received higher ratings for convenience and technology than analog aids. With regard to circuit type, overall ratings showed no difference between linear and nonlinear aids regardless of the test instrument. Surprisingly, directional microphone hearing aids received similar ratings to aids with omnidirectional microphones, even for assessment scales specific to communication in background noise. These results demonstrate the complex relationship between perception of benefit by the hearing aid wearer and standard clinical effectiveness measurements.

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The performance-perceptual test for comparison of perceived versus measured performance
Gabrielle Saunders, Anna Forline, and Jennifer Sowards, National Center for Rehabilitative Auditory Research

The Performance-Perceptual Test (PPT) enables a direct comparison of measured and perceived ability to understand speech-in-noise, using the same test materials, adaptive test procedure and scoring method. A Performance speech reception threshold in noise (SRTN) and a Perceptual SRTN are measured using the HINT procedures. For the Performance SRTN subjects repeat back each sentence, for the Perceptual SRTN subjects indicate whether they can ‘just understand everything that was said’. If a subject indicates they can understand everything, the signal-to-noise ratio (S/N) is made more adverse; if not, the S/N is made less adverse. The difference between the Performance SRTN and Perceptual SRTN is called the Performance-Perceptual Discrepancy (PPDIS). The magnitude of the PPDIS indicates how accurately individuals estimate their hearing ability. A positive value indicates ‘overestimation’; a negative value indicates ‘underestimation’. The relationship between measured performance (Performance SRTN), discrepancy between perceived and measured performance (PPDIS) and reported handicap, hearing aid satisfaction and attitudes towards hearing loss was examined. The tools used were the Hearing Handicap Inventory for the Elderly/Adults, the Satisfaction with Amplification in Daily Life/Expected Consequences of Hearing Aid Ownership (SADL, ECHO) and the Attitudes Towards Loss of Hearing Questionnaire, respectively. The PPT was run unaided and aided at 50, 65 and 80 dB SPL. Subjects with a range of hearing abilities and ages participated. All completed audiological tests, the questionnaires and the PPT in a single session. Subjects returned for retesting 1-2 weeks later. Mean Performance SRTNs are within 1dB of Perceptual SRTNs at all test levels. Test-retest correlations for all...
conditions and levels are strong (r-values vary between 0.88 and 0.99). Mean PPDIS values are almost zero, suggesting that the instructions used to obtain the Perceptual SRTN yield equivalent thresholds to those obtained with the HINT adaptive procedure. PPDIS values range from -4.4 to +4.2 dB and provide insight into hearing aid satisfaction. Data from the SADL and ECHO shows that individuals who are more satisfied with their hearing aids or who have more positive expectations about hearing aids tend to have higher PPDIS values than individuals with less positive SADL/ECHO scores. That is, negative impressions of hearing aids are accompanied by 'underestimation' of hearing ability, while positive impressions are associated with an overestimation of hearing loss. These and other relationships between the PPDIS and the test measures will be discussed. [This work has been funded by the VA RR&D service].

PC10

Multi-channel sound-field system for assessing the real-world benefit of hearing aids
Lawrence J. Revit, Revitronix, Robert B. Schulein and Mead C. Killion, Etymotic Research

Researchers, engineers, and clinicians have expressed a need for an efficient and repeatable means of testing the real-world benefits of advanced hearing aids, particularly in noise. Toward that purpose, this presentation describes the development of such a means: multi-channel sound recordings and a multi-channel listening system for testing the real-world performance of both hearing and hearing aids. The system creates, in the laboratory or clinic, controlled acoustic conditions that accurately represent those encountered by hearing-impaired listeners in the real world. Researchers, engineers, and clinicians, therefore, have an efficient and repeatable means of quantifying real-life improvements of hearing in noise. The design goals of the system required that: 1) the simulated environments should sound real; and 2) the simulated environments should allow hearing aids and the hearing mechanism to perform as they do in the real world. This presentation traces the development of the system, describes its characteristics, and gives examples of how it has been used in research and development settings, to date. A separate presentation (C. Compton, et al.) will describe the formal clinical validation of the system.*

*This last sentence, of course, is contingent on the acceptance of the Compton et al. submission.

PC11

Test paradigm manipulation during the evaluation of speech recognition in noise
Michael Nilsson, Robert Ghent, Victor Bray, Jr., Sonic Innovations

Benefit from amplification can be expressed in many domains, one of which is speech recognition, either in quiet or noise. One method in wide-spread use, is the Hearing In Noise Test (HINT; Nilsson et al, 1996) which measures a Reception Threshold for Sentences (RTS) using an adaptive technique. Originally designed with a single, moveable source and a fixed presentation time, several modifications to the materials have been investigated to understand the utility and generalization of HINT performance to other, non-clinical conditions, as well as its sensitivity to testing advanced signal processing features available in new hearing aids. Noise onset time, single versus multiple noise sources, correlated versus uncorrelated maskers, and steady-state versus modulated maskers (multi-talker speech) were all manipulated in a multi-site evaluation using normal-hearing and hearing-impaired listeners. Data will be presented and indicates that performance is highly dependent upon the level of these manipulations, necessitating careful selection and control of test conditions when evaluating benefit from signal processing.

Signal Processing

PC12

Detection thresholds for frequency-dependent group delay: Implications for digital hearing-aid design
Kathryn Arehart, University of Colorado at Boulder, James M. Kates, Cirrus Logic / Audiologic Design Center, John Hansen, Jessica Rossi-Katz and Ajay Natarajan, University of Colorado at Boulder

In many digital hearing aids, the signal processing is implemented using the fast Fourier transform (FFT). The transform size is generally chosen so as to provide frequency resolution comparable to that of the human auditory system at low frequencies. The FFT size normally used, for example a 128-point transform at a sampling rate of 16 kHz, also introduces unwanted processed signal delay. A newly introduced dynamic range compression system based on digital frequency warping (Kates, 2002) allows frequency resolution that follows the auditory critical
band frequency scale while reducing the system delay. However, the delay in the standard FFT processing does not depend on frequency, while the delay in the new warped compression system is frequency-dependent, with the greatest delay at low frequencies.

The digital frequency warping is implemented as a cascade of all-pass filter sections. An important design objective of the frequency-warping system is to determine its optimal filter length. In this paper, we address two issues related to this objective. First, we are interested in determining if the delay effects are audible for the shortest filter that gives adequate frequency resolution. Second, we are interested in determining the longest filter length that gives undetectable delay effects. An increase in the warped FIR filter length, and thus, in the frequency resolution of the system, is feasible as signal processing power increases. However, the increase in frequency resolution occurs at the expense of increased group delay. The perceptual effects of the group delay will be discussed in the context of the optimal filter length for the new frequency-warping compression system.

In this paper, we report detection thresholds for frequency dependent group delay for conditions simulating individuals listening to another talker (as opposed to listening to their own voices). Detection thresholds were obtained in a group of normal-hearing listeners and in a group of hearing-impaired listeners using a three-interval three-down one-up adaptive procedure.

Test stimuli included clicks and CVC syllables processed to duplicate the delay effects of the frequency-warped system. The stimuli for the hearing-impaired subjects were amplified and shaped using the NAL-R response appropriate for each individual hearing loss. The delays were provided by a cascade of all-pass filter sections, with the threshold established by determining the number of sections for which the delayed stimulus could be detected in comparison with the undelayed input. The first-order all-pass filters sections were designed so that the warped frequency scale corresponded to the auditory Bark scale at the 16-kHz sampling rate used in the experiment.

### PC13

**Characterizing the dynamic behavior of advanced hearing instruments in real-life environments**

Peter Nordqvist and Arne Leijon.

KTH - Dept of Speech, Music and Hearing

Advanced hearing instruments adapt their behavior depending on the input signal, for example to reduce the audibility of background noise, separate speech from noise, sharpen speech formants, and/or emphasize weak consonants.

Some types of so-called noise suppression only reduce the long-term gain response in noisy environments. The effect can be illustrated by an average gain response for each environment, using realistic test signals. The only question is: Which averaging method gives the most relevant information?

It is more difficult to describe the rapid (“syllabic”) dynamic behavior. We need a reasonably simple method to characterize statistically the time-frequency modulation patterns of hearing-aid gain as a function of the time-frequency modulation patterns in the input signal.

1. To describe slow adaptation, we compare two methods to obtain a single environment-specific gain response curve: (A) The log ratio between output and input long-term average power spectra, and (B) the time average of log ratios between output and input short-time power spectra.

2. To describe fast adaptation we compare three methods:
A: Level distributions and modulation spectra of input and output signals, measured in separate frequency bands.
B: A parametric ARMA system model, describing the hearing aid gain at each time and frequency as a linear function of current and previous input and output signal levels in all frequency bands.
C: A finite-state system model, using a Hidden Markov Model (HMM) to represent the modulation patterns of the hearing-aid gain frequency response as a function of the input signal modulations.

We will demonstrate how these methods can illustrate clinically interesting features of a few modern hearing instruments. We also apply the methods to simulated noise-suppression and compression algorithms.

The conventional separate-band approach (method 2A) can clearly show the effective modulation transfer function within each band, but it can not illustrate the correlated gain adaptations when compression...
channels are wider than the analysis bands. The ARMA approach (2B) can reveal across-frequency correlations, but the linear model structure may be too restrictive for some adaptive hearing-instrument algorithms. The results of the HMM approach (2C) are easy to understand and can clearly illustrate the function of noise suppression and fast compression algorithms. A first-order HMM cannot completely represent the modulation spectrum of speech, but it is probably sufficient to represent the temporal characteristics of gain adaptation.

**PC14**

Influence of delay and nonlinearities on the performance of adaptive filters with correlated disturbance

Marion Schabert and Walter Kellermann, University of Erlangen-Nuremberg, Germany

Adaptive filters are a powerful vehicle in signal processing. They are used to identify and track unknown systems in various application areas such as echo cancellation and feedback control for digital hearing aids.

In hearing aids, the receiver signal feeds back to the microphone via the unknown feedback path. This feedback signal is recorded together with the external source signal. Filtering the receiver signal with a feedback path model and subtracting it from the microphone signal performs feedback suppression. Adjustment of the filter weights is based on the estimation error signal which is only accessible with an additive disturbance, i.e., feedback minus estimated feedback plus external source signal.

As this disturbance is not independent of the driving signal (receiver signal) of the adaptive algorithm, the optimum solution cannot be achieved by standard adaptive filters based on Wiener filter theory, and the feedback path cannot be identified.

The procedure models the unknown system plus the relation between driving signal and disturbance (external source). Assuming linear signal processing for compensation of hearing loss, it tends to cancel the external source signal.

To examine such systems we investigated system identification with correlated disturbances for open loop systems. Decorrelation by passing the driving signal through both delay and nonlinear devices is examined in this work. Influence on the achievable system identification and convergence is derived with respect to decorrelation means. We investigate both white and colored noise as driving signals. The used algorithms are time domain algorithm NLMS (normalized least mean square) and frequency domain algorithm FLMS (frequency domain LMS).

Results we obtained so far indicate improvements in system identification, especially for the application of nonlinearities together with FLMS algorithm. Both necessary delay and parameters for various nonlinearities are quantified for a desired system identification performance from a signal processing point of view.

**PC15**

Novel numerical representations for low-power audio signal processing

Roger Chamberlain, Yen Hsiang Chew, Varuna DeAlwis, Eric Hemmeter, John Lockwood, Robert Morley, Ed Richter, Jason White, and Huakai Zhang, School of Engineering and Applied Science, Washington University, St. Louis

One of the major technical issues facing the designers of modern digital hearing aids is the need to minimize the power consumption of the system to prolong battery life. As new signal processing techniques are proposed, the computational requirements invariably grow, putting additional pressure on power consumption. In this work, we investigate the use of non-standard numerical representations for the audio signals being processed, showing how the power consumption can be lowered for audio signal processing while maintaining (and even improving) overall signal quality.

Standard numerical representations include fixed-point representations (typically 16 bits) and floating-point representations (either 32- or 64-bit IEEE standard). In this study, we compare the power consumption of a 16-bit linear representation with several different floating-point representations (4- to 6-bit exponent and 4- to 6-bit mantissa) and a 9-bit logarithmic notation. Each representation is tailored to provide a dynamic range of approximately 100 dB and a signal-to-quantization-noise ratio of approximately 30-35 dB (i.e., optimized for understanding of speech signals). The power consumption is investigated while computing a series of multiply-accumulate operations. The multiply-accumulate (MAC) is the most common computation in audio signal processing.

For each representation, we design a hardware MAC unit in the VHDL language and perform a standard-cell synthesis, layout, and place-and-route targeting the AMI Semiconductor 0.5 micron VLSI integrated
circuit process. The resulting design is simulated using the Mentor Graphics MACH-PA power analysis tool, with input vectors modeling a 21-tap finite impulse response band-pass filter. The simulation output both verifies correct operation of the circuit and provides information on power consumption.

The results show a significant power savings (greater than 5x) using both the floating-point representations and the logarithmic representation. This is primarily due to the ability to either eliminate (in the case of the logarithmic representation) or significantly reduce the size of the hardware multiplier required as part of the MAC unit. We will present both novel techniques for implementing the accumulation function with a logarithmic representation as well as the power consumption associated with each numeric representation.

[This work was supported by NIDCD].

PC16

Efficient subband representations for spatial filtering for hearing aids
Jeff Bondy, Simon Haykin, Ian Bruce and Susanna Becker, McMaster University

The low power, low MIPS available in general-purpose DSP hearing aids force algorithms to be extremely lean. The difficult environment in which an aid is required to operate often requires a very robust implementation, reducing the number of MIPS available for performance enhancement algorithms.

The Discrete Wavelet Transforms (DWTs) have been shown to be less computationally expensive than the Discrete Fourier Transform (DFT) in some cases. Also, beamforming wideband signal approximations done by applying phase weights produce errors as the bandwidth versus the center frequency grows (Buckley 1987). The family of power minimization algorithms can only produce phase weights for a narrow band without producing error. The error from phase weight implementation increases as the Q of the signal of interest shrinks. Bifurcating DWT trees produce constant Q frequency bands. Therefore, there is a natural fit to using the DWT decomposition for broadband beamforming, in that consistent error bounds are formed in each subband.

To make the representation optimal, one must always be aware of the source, environment and receptor. By utilizing a constant Q decomposition we match the assumptions of the algorithm and the environmental characteristics. We are studying several subbanding analysis structures that attempt to match the receptor and source: 1) A bifurcating tree, with frequencies resolved more tightly in proportion to the Articulation Index (AI). This is an attempt to match the frequency decomposition characteristics of the human auditory system. 2) A temporally efficient tree whose subbranches are split to produce good time resolution to capture fast transients from the source (i.e. most consonants) but still shows good frequency resolution keyed to the critical band structure of the AI. In this way we are trying to match the frequency decomposition characteristics to both the source (speech) and the receptor (the auditory system).

These two algorithms are compared, using performance and computational efficiency as metrics, with the standard DFT implementation. It is shown that the subband strategies produces good AI-weighted directivity gain at a reduced computational load.

PC17

Subband adaptive feedback cancellation on an ultra low-power miniature DSP platform for hearing aids
Dequn Sun, Robert Brennan, Todd Schneider and King Tam, Dspfactory Ltd., Canada

Acoustic feedback in hearing aid systems is a common problem due to the high gains typically required in hearing aids, as well as the close proximity of the microphone and receiver. Previous research in this area proposed methods to detect feedback and reduce the hearing aid gain, consequently reducing the effectiveness of the hearing aid.

Other research has dealt with adaptive cancellation of the feedback in a digital system using a fullband implementation of the well-known Least Mean-square (LMS) algorithm. This method uses one adaptive finite impulse response filter to model the feedback path in the full frequency band of interest. The model of the feedback path is then used to estimate the feedback signal, which is then subtracted from the hearing aid input signal. While this method theoretically allows high gain of the hearing aid, the fullband implementation of the LMS algorithm has known performance limitations, particularly with highly-colored signals and high-order filters. High-order filters require increased computational power and result in slower convergence of the filter coefficients. Convergence is further slowed when the input signals are colored.

We present the design and implementation of an LMS-based feedback cancellation system that efficiently divides the input signal into near-orthogonal subbands using an efficient WOLA
(Weighted Overlap-Add) filterbank coprocessor. Each subband contains an adaptive filter which models the feedback path only within its limited bandwidth. The adaptive filters are lower-order than the equivalent fullband filter and can therefore be updated using the LMS algorithm with less computational cost. Furthermore, performance is improved because the lower-order adaptive filters converge to their desired values more quickly. The near-orthogonal nature of the subband signals allows the filters to be adapted independently.

Another benefit of the subband approach is the reduced spectral dynamics, or whitening, of the LMS input signals. A colored input signal is decomposed into subband signals which have reduced spectral dynamics, resulting in behavior that is closer to the ideal LMS performance. This results in improved convergence performance of the LMS algorithm, thereby increasing the effectiveness of the feedback cancellation system.

The real-time implementation of the system exhibits overall system performance that is improved over other known feedback management systems for hearing aids. The subband LMS algorithm runs on the Toccata DSP platform, operating at hearing aid power levels.

Methods to further improve the performance of the LMS algorithm in subbands are being investigated.

**Polymer cantilever array for hearing aid and cochlea implant application**

Tao Xu, Fan-gang Zeng, Mark Bachman, Guann-Pying Li, University of California, Irvine

In the world, 10 percent of the general population suffers from hearing loss and that number rises to about 35 percent after age 65. Therefore, it is very significant to develop some new devices that may improve hearing loss, such as digital hearing aids and cochlea implants. In these devices, digital signal processing (DSP) technique is usually used to process the sound signal due to its frequency analysis ability. However, the high power consumption and long processing time are two big problems that the DSP will face when the more channels are required.

The basilar membrane and the tectorial membrane are two kind of organic membranes in mammalian cochlea. The membranes are just like the mechanical filters. Resonance will happen at some parts of the membrane when the sound arrives. The vibrations will lead to the stimulation of the inner hair cells, which will produce the hearing. The normal mechanical filters are always made of the high Youngi's modulus material, like silicon, in order to obtain the high quality factor. However, the research results show that the organic membranes in mammalian cochlea just have about 1-10 of the quality factor (Q10) when the resonant frequency is in sound range. They are much lower than that of the normal mechanical filter. As an organic material, the polymer has much lower Youngi's modulus than the silicon. Therefore, the low quality factor can be obtained with polymer. In this paper, we present a polymer cantilever array, which was made using the polymer with 4.4 GPa of Youngi's modulus. The quality factors (Q10) of the cantilever in the cantilever array were 9.38, 10.11, 11.56, 14.01 when their resonant frequency is 286 Hz, 720 Hz, 2.868 kHz, 6.948 kHz, respectively.

With the MEMS technology, we can make the cantilever array small enough to assemble it in a small size, so it can provide many enough cantilever when the more channels are required. In addition, because the optical method was used to detect the vibration of the cantilever, the device is free from the electromagnetic interference, has high resolution, and needs low power consumption.

**Alternative Therapies for Sensorineural Hearing Loss**

**PC19**

Reverse Transfer Function (RTF) measurement: An objective tool to determine the performance of the active middle ear implant transducer

Mark Winter, Benno Paul Weber, and Thomas Lenarz, Department of Otolaryngology, Medical University of Hanover

The Symphonix Vibratin® soundbridge (VSB) has proven to be an alternative treatment modality for patients suffering from moderate to severe sensorineural hearing loss. The results from the majority of our 39 patients implanted over the last 5 years are very positive and encouraging. Clinically, variations in benefits obtained from the VSB are observed across the patient population with differences in functional gain and speech understanding results. Currently there is no available means to objectively measure the transducer performance (principally influenced by the coupling of the FMT to the long process of the incus) neither intra nor postoperatively.

When stimulated, the active middle-ear implant elicits a sound pressure level in the external auditory ear canal transferred via the tympanic membrane acting as
a loudspeaker. The generation of the sound pressure level in the external ear canal is known as the Reverse Transfer Function (RTF). Measurement of the RTF shows a high correlation to the displacement of the stapes footplate and to individual functional gain measurements in the sound-field. Therefore the RTF can be used to describe the stimulation of the inner ear.

The measurement procedure involves presentation of signals in the form of sinusoidal sweeps over the audible frequency spectrum and a probe microphone placed in the ear canal. This enables and provides the possibility to analyse the RTF precisely to disclose coupling effects that would otherwise remain undetected by functional gain measurements. Thus it offers new opportunities to evaluate the biomechanical effects of middle ear implants. Intra-operatively it can be used to optimise the coupling and post-operatively to tailor the fitting of the AP.

To date we have measured the RTF for 31 patients postoperatively and collected preliminary intra-operative data. The data will be presented and viewed in relation to possible correlations to the individual's performance on standard audiometric measures.

[This work was supported by Symphonix Devices, Inc.]

**PC20**

**Speech-to-text technologies for communication enhancement: a discourse analysis**

Anita Haravon and Loraine Obler, City University of New York, Graduate Center, Judith Harkins, Gallaudet University and Harry Levitt, City University of New York, Graduate Center

Automatic speech recognition (ASR) has the potential to improve communication for people with hearing loss by supplementing the spoken word with text. This project evaluates the communicative effectiveness of ASR technology as compared to its more costly, yet more accurate counterpart, computer-aided real-time transcription (CART). Six deaf-hearing dyads participated in the study and communicated under five conditions: face-to-face-only, CART-face-to-face, ASR-face-to-face, CART-only, and ASR-only.

Results are reported for four measures of communicative effectiveness. Task outcome was measured using the Map Task, a collaborative problem-solving task, scored for route reproduction accuracy. Mean scores across communication conditions were not significantly different. However, irrespective of communication condition, participants with high speech-reading scores tended to reproduce the route more accurately than did participants with low speech-reading scores. Completion time was significantly longer in ASR conditions than in face-to-face-only and CART conditions. In terms of transcription accuracy, ASR was found to be significantly less accurate than CART. The 10-percentage-point discrepancy between ASR (87%) and CART (97%) transcription accuracy was expected since CART is known to be a more reliable transcription method. The discrepancy between accuracy in the ASR-only (87%) and ASR-face-to-face (77%) conditions may reflect that the hearing participants were faced with a dilemma—whether to maintain communicative norms (eye contact, visual feedback) with their deaf partners or focus on the ASR output. The conflicting tasks may have resulted in sloppier speech and thus less accurate transcription. Participants' subjective rankings of conditions were generally in agreement across participants with CART-face-to-face ranking first, face-to-face-only and CART-only tying for second, ASR-face-to-face ranking third, and ASR-only coming in last.

These results suggest that increased communicative effort was needed in the ASR conditions than in the CART conditions to achieve the same level of performance. In terms of participant evaluation, CART was preferred over the no technology condition and the no-technology condition was preferred over the ASR condition. In terms of task outcome, speech-reading ability appears to be a better predictor of task outcome than communication condition. The multiple task demands of current ASR technology for face-to-face communication can prove detrimental to communicative success.

**PC21**

**Intervention for restricted dynamic range and reduced sound tolerance**

Craig Formby and LaGunn Sherlock, University of Maryland School of Medicine. Susan Gold and Ellen Frederick, University of Maryland, Tinnitus and Hyperacusis Center and Sharon Palmer, University of Maryland, College Park

The purpose of this presentation is two-fold. First, we will overview clinical and experimental evidence that provides the rationale and motivation for an NIH-sponsored clinical trial of an innovative intervention for reduced sound tolerance. The treatment approach, which assumes adaptive plasticity of loudness, extends desensitization principles from Tinnitus Retraining Therapy (TRT) to patients in the general hearing-impaired population whom experience
debilitating sound tolerance problems. Following this overview, we will describe the design of the randomized, double-blind placebo-controlled clinical trial, which has as its primary aim to assess the efficacy of the TRT-based treatment and its individual component parts (i.e., directive counseling and sound therapy). The target patient population for this small-scale clinical trial is a randomized sample of hearing impaired persons who previously have been unsuccessful users of hearing aids. Their reported failure to benefit from amplification was due either to (1) abnormally reduced sound tolerance (i.e., hyperacousis) or (2) a limited dynamic range for sound intensity (resulting from their sensorineural deficit and audibility loss). Ultimately, we expect to show that the TRT-based treatment is efficacious. Those patients randomized to the full-treatment arm of the study (or to an arm with one of the salient components of the treatment), who previously could not use amplification because of their sound tolerance problems, will, after treatment, be converted to successful hearing aid users. Furthermore, these treatment effects will be sustained with amplified sound.

[Research supported by NIH awards 1R01 DC04678 and R21 DC04514.]

**PC22**

**Evaluating the benefit of speech-recording hearing aids for children**

Denise Miller-Hansen, Children's Mercy Hospital, Peggy Nelson, University of Minnesota and Judith Widen, University of Kansas Medical Center

At Children's Mercy Hospitals and Clinics, audiologists have fit speech-recording (previously called transposition) hearing aids on children since 1995. Outcomes data were analyzed to determine the amount of gain received from these aids and the appropriateness of the fit. Specifically, aided pure tone and speech test results were evaluated for 78 children with bilateral sensorineural hearing loss fit who were with speech-recording hearing aids (AVR Sonovation ImpaCt DSR). Children ranged in age from 1.3 years to 21.6 years with a mean age of 10.6 years. Of these 78 subjects, 44 presented with profound loss, 19 with severe loss, 8 with moderately severe loss, 3 with moderate loss and 4 with mild loss. Subjects showed better pure tone averages (PTA) with the speech-recording aid than in the unaided condition. The mean improvement in PTA (5, 1, and 2 kHz) was 49 dB with a range of 44 to 53 dB. For high frequency PTA (1 and 2 kHz), the mean improvement was 56 dB with a range of 52 to 59 dB.

Aided results from 19 children who had previously used conventional hearing aids were also compared and contrasted with their performance using speech-recoding aids. For the sixteen children who were able to participate in word recognition testing there was a mean improvement of 12.5% (95% CI: 4% to 21%) with speech-recoding aids providing significantly better performance than conventional hearing aids. Aided pure tone averages obtained using the speech-recoding aids were compared with those of conventional hearing aids. Pure tone averages were better with the speech-recoding aids with a mean improvement of 11 dB (95% CI: 6 dB to 16 dB) over the conventional aided PTA. Finally, rate of hearing aid repair was analyzed for the same nineteen users. Speech-recording aids were repaired more frequently than the conventional hearing aids in this small sample with 1.2 versus 4 repairs per patient year (95% CI: 1.7 to 4.6). Illustrative case examples and further description of benefit will be provided.

**The Ear Canal and Earmolds**

**PC23**

**Characterization of critical features from digital imaging of static and dynamic ear canal impressions**

David Fabry, Roland Lehmann, Stirnemann Alfred, Hans Hessel, Phonak Ltd and Phonak AG

Recent innovations in hearing aid manufacturing have permitted the development of computer aided shell manufacturing for in-the-ear hearing aids. Based on laser scanning of 330 earmold impressions, a database has been developed that enables computer modeling of "critical features" present in the pinna, concha cartilagines, and bony portion of the external auditory system. To date, little attention has been given to predictive modeling of ear canal dimensions toward the development of improved shell-making techniques, and the eventual elimination of silicone earmold impressions. Although the recent emphasis in hearing aid commercial markets has focused on acoustic digital signal processing, digital mechanics serve as a primary mechanism for improved comfort, reduced occlusion and feedback, and higher overall satisfaction with hearing aid performance.

The focus of the present study was to use rapid prototyping technology for 3D analysis to describe and evaluate over thirty physical dimensions for each of 330 earmold impressions. For each subject, earmold impressions were made with three jaw positions (closed, intermediate, open) that permit
static and dynamic properties of the cartilaginous portion of the ear canal to be evaluated. Detailed results will be provided; preliminary analysis suggests that a subset of five or six dimensions may accurately account for the majority of variance across ears, and may be used to compensate or predict ear canal dimensions with limited data.

Additional implications of digital mechanics will be to discuss laser-imaging techniques of the ear canal, alternative service and delivery models for hearing aid delivery, and improved quality control for in-the-ear devices.

**PC24**

Sound attenuation characteristics of materials to couple hearing instruments to human ear canals

Vasant Kolpe, Robert Oliveira and Martin Babcock, Hearing Components

The progress in delivering amplified sound via digital and analog signal processing to the impaired human ear has been breathtaking especially during the last decades. The techniques to control quality and delivery of amplified sound to the ear are numerous. However, the sound attenuation characterization has been limited to some well-known classes of polymers and polymer foams. This hampers progress because of the lack of attenuation data and its relationship to basic viscoelastic polymeric properties.

We, at Hearing Components Inc., have constructed an acoustic fixture that enables us to test slow and fast viscoelastic foams. The characterization technique couples the fixture to the well-known Fonix® 6500 – CX hearing aid system. We will report new sound attenuation data that has been collected over a range of thickness under diametric compression for slow and fast porous foams and rubber film. Reproducible attenuation data over a range of frequencies provided by the test system allows optimized selection of thickness, shape etc. of the electronic module to the coupling material for the human ear canal.

[This work was supported by NIDCD].

**PC25**

The dynamic human ear canal: quantitative changes in canal volume with jaw articulation and its relevance to hearing aid use.

Robert Oliveira, Vasant Kolpe, and Greg Hoeker, Hearing Components

People involved in hearing aid R&D would agree that phenomenal improvements have occurred to the technology inside hearing aids over the past few years.

Hearing aid technological advances to enhance communication can be likened to automobile technological advances to enhance transportation. Certainly the technology of the engines of our cars has made similar dramatic improvements to those of the DSP inside our aids. In addition to the advances of engines, a better understanding of the interplay of tires and the road traveled has done much to provide improved transportation. Unlike the automobile and road in the analogy, the physical interaction of the hearing aid with the ear canal has been largely ignored.

Over the past 5 years we have been refining our studies on the human ear canal and its physical properties. Laser-digitizing systems were used to measure ear impressions. Physical changes in ear canals were captured at different jaw positions. Proprietary software was used to quantify these changes. New data will be presented on the surprisingly large volume changes that occur in the cartilaginous region of ear canals on jaw articulation. Further, we have investigated these changes as a function of age and see notable and clinically relevant age-related correlations. The basic mechanism for the age related change will be described using our magnetic resonance imaging of jaw position induced changes in the ear canal. Relevance to the successful use of hearing aid technology will be discussed.

[This work was supported by NIDCD].

**Spectral Enhancement**

**PC26**

Spectral sharpening based on empirical parameters

Peggy Nelson, Jeffery DiGiovanni, University of Minnesota, Department of Communication Disorders, Juan Carlos Tejero-Calado University of Malaga, Spain, Janet Rutledge, University of Maryland, Baltimore County

Digital hearing aids provide the opportunity for introducing new customized processing for individual listeners with hearing loss. One appealing customized algorithm might be spectral sharpening of noisy speech signals to improve signal-to-noise ratio and partially compensate for listeners' reduced spectral resolution. Recent attempts to implement spectral sharpening algorithms have not been satisfactory nor proven to improve speech understanding. However, recent data from DiGiovanni et al. (2002) indicated that spectral sharpening is a theoretically viable goal. In that study, increased peak-to-valley ratios of 6 dB with a valley width of at least 400 Hz provided
significant improvement for five hearing-impaired listeners' detection and discrimination of spectral peaks. Presented here is a spectral sharpening processing algorithm based on those findings that uses linear predictive coding (LPC) to identify the important spectral peaks in overlapping time windows. The 15 first LPC parameters are computed using a correlation method. Then, the LPC spectrum is computed and the 4 principal peaks are identified. The gain for the principal peaks is computed based on the dynamic range of the individual listener. The peak-to-valley ratios are sharpened by establishing a minimum peak spacing and a 6-dB minimum peak-to-valley ratio. The application uses 7.5 ms analysis frames and 30 ms Hamming windows, leading to a 4 to 1 time overlap. A 512-pt FFT is used to provide sufficient resolution for the speech sampled at 9.6 kHz. Synthesis is done by using an inverse FFT. Quiet and noisy nonsense syllables processed using these parameters were presented to listeners with hearing loss. Results will be described and future implications discussed. [Work supported by NIDCD R03-DC04125 and by the University of Minnesota.]

PC27
Perceptual evaluation of spectrally enhanced speech
Ayashkanta Rout, Ohio University, Brandon J. Laflen, Purdue University and Amy Neel, University of New Mexico

The effect of spectral enhancement on speech recognition and sound quality was investigated in subjects with normal hearing and sensorineural hearing impairment. A custom designed algorithm developed in our laboratory was used to preprocess speech stimuli at four levels of spectral enhancement. Connected speech, nonsense syllables, and vowels in CVC context were mixed with noise at +5 dB signal to noise ratio and spectrally enhanced. Eight normal hearing and 14 hearing impaired subjects participated in 2-3 sessions of two-hours each. The results indicate no significant improvement in speech recognition performance with spectral enhancement. Overall quality of the spectral enhanced speech was judged by the subjects on a paired comparison task. The hearing impaired subjects preferred the sound quality of the speech processed with a small amount of spectral enhancement. Normal hearing subjects preferred the quality of unprocessed speech (no spectral enhancement). Results will be presented in detail and implications of these findings for hearing aid design will be discussed. [This work was supported by Purdue Research Foundation grant 690/1353-3204]

PC28
A flexible, analytical framework for applying and testing alternative spectral enhancement algorithms
Brandon J. Laflen, Purdue University, Ayashkanta Rout, Ohio University, Thomas M. Talavage and Pranesh Thirukkonda, Purdue University

Previous research has demonstrated that as a result of cochlear hearing loss, auditory filters are broadened and the internal representation of the 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 pre-process 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 flexible framework has been developed for applying various spectral enhancement algorithms using Matlab. Supported algorithms differ by the mechanism for detecting peaks/valleys in the short-term speech spectrum, and include a bin-region collection algorithm, a filtered second-derivative based algorithm, and a variant of Cepstral analysis. The identified peak (valley) regions are amplified (attenuated) according to a variety of scales, including constant, linear, and exponential functions, with variable amplitude. Spectral shaping is accomplished through a finite impulse response (FIR) filter upon a 100% overlap moving window. The output produces WAV files from any desired combination of supported algorithms and scaling factors applied to any set of input WAV files.

Additionally, the framework permits empirical comparisons of the above output combinations through physiological models of the peripheral auditory system that generate neural activation patterns (e.g. the Auditory Image Model, AIM, Patterson et al.). Given a characterization of the hearing loss and a suitable metric for perceptual evaluation, each output combination is evaluated against the others, and against the theoretical "normal-hearing" activation pattern. These comparisons result in measurements of "perceptual difference," which can be useful in predicting the relative success of each processing strategy.
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