Clinical Relevance for the Veteran

SUMMARY OF SCIENTIFIC/TECHNICAL PAPERS IN THIS ISSUE

by Harry Levitt, PhD, Guest Editor

Achieving Prescribed Gain/Frequency Responses with Advances in Hearing Aid Technology Sammeth, et al. *(p. 1)*

Purpose of the Work. Technological limitations have, until recently, restricted the capability of in-the-ear hearing aids to match prescribed frequency-gain characteristics. New technology has recently become available which offers substantial improvements in this capability. The purpose of this study was to evaluate the extent to which this new technology is capable of matching prescribed frequency-gain characteristics for an in-the-ear hearing aid. **Subjects.** Sixty hearing aid users participated. Half of the subjects had been fitted with older generation in-the-ear hearing aids, the other half were fitted with modern in-the-ear instruments using the new technology. **Procedures.** Standard techniques were used in fitting all of the in-the-ear hearing aids. The quality of each fit was then evaluated by measuring the sound pressure levels generated in the ear canal by the hearing aids. **Results.** The older technology hearing aids were found to provide too much gain through the mid-frequencies, and too little gain in the high frequencies. In contrast, the newer technology hearing aids provided a much closer approximation to the prescribed gain across the frequency range. **Relevance to Veteran Population.** Large numbers of veterans are fitted with in-the-ear hearing aids each year. The prescriptive fitting of in-the-ear hearing aids can be improved substantially using the newer technology.

Properties of an Adaptive Feedback Equalization Algorithm Engebretson, et al. *(p. 8)*

Purpose of the Work. A common problem with hearing aids is that not all of the amplified audio signal reaches the eardrum. Some of it escapes and reaches the hearing-aid microphone and is amplified once again. This process, known as acoustic feedback, can cause whistling and other unstable behavior in the hearing aid. Digital signal processing techniques provide a means for reducing the effects of acoustic feedback in hearing aids. The purpose of this study was to develop and perform laboratory evaluations of one such method of feedback reduction. An evaluation of the technique with hearing-impaired subjects is reported in the companion paper in this issue, by French-St George, et al. (see p. 17). **Procedures.** The characteristics of the acoustic feedback signal are estimated using a known signal at the input to the hearing aid. An electrical signal with identical characteristics is then generated and subtracted from the signals being amplified, thereby canceling the feedback signal. Since the sound transmission characteristics of the feedback signal changes over time, the cancellation process also adapts over time providing continuous, adaptive feedback reduction. **Results.** Laboratory evaluations of the feedback reduction technique showed that the magnitude of the feedback signal could be reduced substantially. As a consequence, the output power of a hearing aid can be increased by a factor of 10 or more without causing the hearing aid to whistle. **Relevance to Veteran Population.** Many veterans using powerful hearing aids, or with earmolds that allow for significant acoustic feedback, are not receiving the amplification they require because of whistling and related problems. Hearing aids incorporating advanced signal-processing techniques for feedback reduction will alleviate this problem considerably.

Behavioral Assessment of Adaptive Feedback Equalization in a Digital Hearing Aid French-St. George, et al. *(p. 17)*

Purpose of the Work. In the companion paper by Engebretson, et al. in this issue (see p. 8), a method for acoustic feedback reduction was developed using digital signal processing techniques. The purpose of this study was to evaluate the technique with hearing-impaired subjects. **Subjects.** Nine hearing-impaired subjects (5 male and 4 female) having an average age of 63.4 years (range 39 to 76 years) participated in the study. **Procedures.** A wearable master digital hearing aid was used. The instrument was programmed to simulate the subject's own hearing aid both with and without acoustic feedback reduction. The subjects used the hearing aid for a range of conditions including speech at a low signal level both in quiet and in noise. **Results.** Hearing aid users are often limited by acoustic feedback when listening to speech in quiet at a low signal level. The feedback reduction technique allowed for significantly more gain to be used under these conditions without unstable acoustic feedback. The use of increased gain resulted in improved speech intelligibility. **Relevance to Veteran Population.** Many veterans who use hearing aids are not getting the gain they require because higher gain settings lead to unstable acoustic feedback (whistling). The signal pro-

cessing technique evaluated in this study would reduce this problem significantly.

Intelligibility of Frequency-Lowered Speech Produced by a Channel Vocoder Posen, et al. *(p. 26)*

Purpose of the Work. Frequency lowering is a form of signal processing designed to match the speech signal to the residual hearing capacity of a listener with a high-frequency hearing loss. The purpose of this study was to investigate new forms of frequency lowering for persons with no high-frequency hearing. **Subjects.** Two normal-hearing subjects participated. The test stimuli were processed to simulate a hearing loss in which no high-frequency speech cues are available. **Procedures.** The speech signal was subdivided into eight contiguous frequency bands covering the range from 1,000 Hz to 5,000 Hz. The variations in signal level in each band were reproduced on one of eight low frequency bands of noise. The eight low frequency noise bands covered the frequency range from 400 to 800 Hz, and had the same variations (modulations) in signal level as the eight high frequency speech bands. Speech recognition ability was then measured when only those frequencies of the speech below 800 Hz were present, and when the modulated noise bands were added to the low-frequency speech signal. In a modified version of the system, the noise bands were omitted for speech sounds with substantial energy in the low frequencies. **Results.** The addition of the modulated low-frequency bands of noise improved the recognition of speech sounds having significant information in the high-frequencies, but degraded the perception of speech sounds with mostly low-frequency energy. The modified system maintained the advantage for the high-frequency speech sounds without degrading the perception of the low-frequency speech sounds. **Relevance to Veteran Population.** The results of this research will facilitate the development of special-purpose signal processing hearing aids that will improve speech intelligibility for veterans with severe high-frequency hearing loss.

Toward a Theory of Optimal Hearing Aid Processing James M. Kates *(p. 39)*

Purpose of the Work. In most cases, a hearing loss is the result of damage to the inner ear. An ideal hearing aid for such losses would restore the functioning of the impaired ear to match that of a normal ear. As a first step towards developing such a hearing aid, a computer program was used to simulate the behavior of normal and hearing-impaired ears. The purpose of this study was to develop a theoretical model of the impaired auditory system in order to design hearing aids that would work more effectively with the various forms of hearing impairment. **Procedures.** A simplified theoretical model of the ear for a specific hearing impairment was developed in order to enable the development of a practical amplification system. The hearing-aid processing was designed to minimize the differences between normal and impaired ears to the extent possible. Processing examples consisting of several individual speech sounds presented for a flat hearing loss. **Results.** The results indicated that a 3-channel compression system, having frequency bands that change in response to the frequency content of the signal, and gains that are controlled by the peak signal level within each frequency region, will be close to the optimal solution. **Relevance to Veteran Population.** Hearing aids designed according to the procedures developed in this paper will be of greater benefit to hearing-impaired veterans.

Spectral Contrast Enhancement of Speech in Noise for Listeners with Sensorineural Hearing Impairment: Effects on Intelligibility, Quality and Response Times Baer, et al. *(p. 49)*

Purpose of the Work. Hearing aid users typically have great difficulty understanding speech when background noise is present. The purpose of this study was to evaluate a new method of processing speech signals in noise to improve both intelligibility and sound quality. **Subjects.** Five to eleven subjects participated in a series of four experiments. All subjects had a sensory (cochlear) hearing loss. **Procedures.** Major peaks in the frequency spectrum of the speech signal were enhanced. Large amounts of enhancement were found to reduce intelligibility, whereas moderate amounts of enhancement produced no significant change in intelligibility. Subjective judgments of intelligibility and sound quality showed a preference for moderate amounts of enhancement. Combining moderate amounts of enhancement with a reduction in the range of intensity variation in the speech signal produced a significant improvement in the intelligibility of speech-in-noise after some practice with the processed signals. The method of processing was also evaluated using response time measurements. This method of measurement was found to be a more sensitive indicator of relative improvement. **Results.** Both the intelligibility and sound quality of speech-in-noise can be improved by enhancing peaks in the frequency spectrum combined with a reduction in the range of intensity variations of the speech signal. **Relevance to Veteran Population.** Signal processing of the type described in this paper could be incorporated in the future development of advanced signal processing hearing aids. These hearing aids would provide improved intelligibility and overall sound quality for hearing-impaired veterans.

Development and Assessment of Two Fixed-Array Microphones for Use with Hearing Aids Bilsen, et al. *(p. 73)*

Purpose of the Work. Hearing-impaired listeners often have great difficulty in situations involving several competing sources of sound, such as at a party. Speech understanding under these conditions can be improved using a microphone that focuses on the speech signal coming from a given direction. This paper investigated the use of an array of microphones to provide the directionality needed to improve the intelligibility of a desired speaker against a background of other competing sound sources. **Subjects.** Thirty normal-hearing and 45 hearing-impaired subjects participated. **Procedures.** A microphone array was developed which could be mounted on the frame and legs of a pair of spectacles. Computer simulations showed that this arrangement could provide an improvement of 10 dB in speech-to-noise ratio at the higher frequencies. Laboratory measurements on an artificial head showed improvements of about 7dB. Clinical measurements on both normal-hearing and hearing-impaired listeners showed improvements in speech reception threshold equivalent to a 7 dB reduction in background noise. **Results.** Microphone arrays that can be mounted on a pair of spectacles can provide significant improvements in speech-to-noise ratio with corresponding improvements in speech intelligibility when the speech and noise come from different directions. **Relevance to Veteran Population.** Veterans who wear hearing aids and who have difficulty understanding speech when several other sources of sound are present would benefit from the use of microphone arrays of the type described in this paper.

Real-Time Multiband Dynamic Compression and Noise Reduction for Binaural Hearing Aids Kollmeier, et al. *(p. 82)*

Purpose of the Work. Binaural hearing aids have many potential advantages (over conventional monaural hearing aids) which have yet to be investigated. The purpose of this study was to develop and evaluate an advanced binaural signal-processing system in which a binaural noise reduction technique is combined with a technique for matching variations in signal level and the resulting changes in loudness to the auditory characteristics of the impaired ear. **Subjects.** Six adults with sensorineural impairments (i.e., impairments of the sense organ and/or its neural connections in the cochlea) were tested. **Procedures.** The signals reaching each ear were subdivided into 24 frequency bands corresponding to the critical bands of the ear. The range of variation in signal level in each band was matched to the available range of hearing in the impaired ear. In addition, an adjustment for the changes in loudness resulting from the impairment was implemented. The binaural noise reduction system amplified sounds coming from the front, and suppressed sounds, including reverberation, coming from other directions. **Results.** Several versions of the processing technique were tested and, if the method of processing was optimized for each subject, improvements in both intelligibility and sound quality were obtained. The binaural noise reduction system was found to work effectively for most subjects for a certain range of signal-to-noise ratios. **Relevance to Veteran Population.** Binaural signal processing techniques offer a means for improving both the quality and intelligibility of speech-in-noise for the large number of veterans who use hearing aids.

Digital Signal Processing (DSP) Applications for Multiband Loudness Correction Digital Hearing Aids and Cochlear Implants Dillier, et al. *(p. 95)*

Purpose of the Work. Recent advances in the development of electronic chips for digital signal processing allow for the flexible implementation of a large variety of speech processing techniques that could be of value in wearable devices such as hearing aids and cochlear implants. This paper evaluates several practical speech processing techniques of this type that could be implemented on a single chip. **Subjects.** Nine normal-hearing and six hearing-impaired subjects participated in an experiment on noise reduction, 13 users of conventional hearing aids evaluated a new method of loudness correction for speech in quiet and in noise, and five cochlear implant users evaluated new methods of coding speech signals for cochlear implants. **Procedures.** Noise reduction was achieved using two microphones feeding a signal processor which adaptively adjusted the inputs to improve the speech-to-noise ratio. The signal processor for loudness correction adjusted the sound level in three adjoining frequency regions to approximate normal loudness perception for hearing-aid users. The speech processor for cochlear implant users recoded the speech signal into sequences of pulses that were delivered to different regions of the cochlea by means of the implant. Unlike earlier methods of speech coding for cochlear implants, several pulse sequences were interleaved to stimulate several regions of the cochlea sequentially rather than simultaneously. **Results.** The two-microphone noise reduction technique was found to improve speech intelligibility in noise subject to certain design constraints which need to be incorporated into any practical system. The signal processor for loudness correction resulted in improved intelligibility for those subjects with poor speech discrimination using their own hearing aids. The new method of coding for cochlear implants was found to improve the discrimination of various consonants in speech, as well as providing useful information on the pitch of the voice. **Relevance to Veteran Population.** Improved methods of signal processing of the type

evaluated in this paper could be incorporated in wearable units, such as hearing aids or cochlear implants, for use by either hard-of-hearing or deaf veterans, respectively.

Design and Evaluation of a Continuous Interleaved Sampling (CIS) Processing Strategy for Multichannel Cochlear Implants Wilson, et al. *(p. 110)*

Purpose of the Work. The coding of speech information for cochlear implants can take different forms, including the use of analog or pulsatile stimulation. In analog coding, the speech waveform is delivered directly to the electrodes of the cochlear implant. In pulsatile coding, the speech signal is represented by trains of pulses which are delivered to the electrodes. The purpose of this study was to investigate an improved version of the pulsatile method of stimulating a cochlear implant. An Ineraid cochlear implant was used in these studies. This device allowed for multichannel stimulation (i.e., several different frequency regions in the cochlea could be stimulated with different electrodes). **Subjects.** Nine deaf subjects who had been fitted with an Ineraid cochlear prosthesis participated in the study. Seven of the subjects were selected for their high levels of speech recognition with the Ineraid cochlear implant. Two were selected for their relatively poor performance. **Procedures.** An improved method of digital stimulation was developed in which the pulse trains to different frequency channels were interleaved to avoid the inter-channel interference that occurs with simultaneous stimulation of two or more frequency regions in the cochlea, as occurs with the analog method. Standard speech recognition tests were performed for both the analog and pulsatile methods of stimulation. **Results.** The new pulsatile method of stimulation resulted in significant improvements in speech recognition relative to the traditional analog method of stimulation. Whereas all of the subjects showed some improvement in speech recognition, the two low performing subjects showed by far the largest improvements on a relative (ratio) scale. **Relevance to Veteran Population.** The new pulsatile method of stimulation may provide substantial improvements in speech recognition for those veterans who have been fitted with a multi-channel cochlear implant.

Effects of Noise and Noise Reduction Processing on the Operation of the Nucleus-22 Cochlear Implant Processor Mark R. Weiss *(p. 117)*

Purpose of the Work. The Nucleus-22 implant system transmits electrical impulses to an array of electrodes that have been implanted in a damaged cochlea. These impulses encode characteristics of the sounds that are received by the implant system microphone and measured by the system's signal processor. When the characteristics accurately represent information bearing speech cues, the ability of the user of an implant to understand speech can be increased. However, when speech is accompanied by noise, the accuracy of the extracted information will be degraded and speech perception will decrease. The purpose of this study was to develop a method of noise reduction for cochlear implants. **Procedures.** Measurement errors for signals with and without noise reduction provided by the experimental method of processing were compared. The data were then analyzed to determine the reliability of the observed reduction in measurement errors. **Results.** The data showed a significant improvement in the accuracy with which essential speech parameters could be measured. As a consequence, cochlear implants of this type (using this mode of processing), function more effectively in noisy environments. **Relevance to the Veteran Population.** Veterans who have been fitted with this type of cochlear implant, if modified to include this type of noise reduction processor, would be better able to understand speech in a noisy environment.

Measurements of Acoustic Impedance at the Input to the Occluded Ear Canal Larson, et al. *(p. 129)*

Purpose of the Work. Measurement of the acoustic impedance of the ear has several useful clinical applications. Test instruments currently used for the clinical measurement of acoustic impedance are limited to only one or two frequencies over a narrow frequency range. This paper reports on the development and evaluation of a new technique for measuring acoustic impedance for many frequency components over a wide frequency range. **Subjects.** Thirty-five normal-hearing subjects at two locations were tested. Ages ranged from 25 years to 35 years. **Procedures.** Estimates of the length and diameter of the ear canal were obtained on each subject. Acoustic impedance measurements using the new instrument were then obtained for a prescribed position in the ear canal and also for two standardized acoustic cavities used for calibration. The acoustic impedance of the eardrum was then derived from these measurements and the estimated dimensions of the ear canal. Computer-based techniques were used to automate the method of measurement and subsequent calculations. **Results.** The measurement technique was found to be both precise and practical using the computerized system. The measurements obtained in the study provided normative data on the acoustic impedance of the ear for multiple frequencies over a wide frequency range. These normative data cover both the expected impedance values and possible range of variation for normal ears. **Relevance to Veteran Population.** The measurement techniques developed and evaluated in this study will result in more effective clinical procedures for diagnosing hearing impairments in veterans and more effective methods of prescribing hearing aids.