GUIDE TO THE ISSUE *by Harry Levitt, Ph.D.*

The papers in this special issue have been divided into three groups. The first deals with various aspects of amplification using digital techniques that have shown promising results in either the clinic or the laboratory. The second group of papers deals with the specific problem of speech in noise and new methods of signal processing for noise reduction. The third group of papers provides examples of useful applications of digital technology in related areas, most notably in signal processing for cochlear implants, but also in developing improved methods of measurement.

SECTION I. DIGITAL TECHNIQUES IN ACOUSTIC AMPLIFICATION

The advantages of digital signal processing were clearly demonstrated in the experimental digital hearing aids that were first developed. A serious limitation of these early instruments, however, was that their size and power consumption were too large for a practical, wearable hearing aid. A useful compromise that reduces power consumption to a practical level for a wearable instrument while at the same time maintaining many of the advantages of digital techniques is that of combining analog and digital technology in order to minimize power consumption.

The most common form of hybrid analog/digital hearing aid is that in which the audio signal is processed by analog components (amplifiers, filters) under digital control. Virtually all of the programmable hearing aids that are currently available for clinical use are of this type. Another form of hybrid amplification is that in which the signal is represented by pulses of varying width (i.e., pulse- width modulation). Amplifiers of this type, known as Class D amplifiers, are very efficient with respect to power consumption and are being used increasingly for the power amplification stage in small in-the-ear (ITE) hearing aids where small size and low power consumption are of critical importance.

Given the limitations of hybrid analog/digital circuits of small size and very low power consumption, can a hybrid Class D amplifier provide the precision and flexibility needed for effective prescriptive fitting of ITE hearing aids? The paper by Sammeth, et al., addresses this issue by investigating the extent to which modern ITE hearing aids using hybrid Class D amplifiers are capable of achieving prescribed frequency-gain characteristics. The results show that modern ITE hearing aids using this hybrid form of amplification can achieve prescribed frequency-gain characteristics with significantly greater accuracy than conventional ITE hearing aids.

One of the many advantages offered by digital hearing aids is that of new methods of signal processing for improving acoustic amplification. One such possibility is the use of adaptive algorithms for reducing or canceling unwanted signals, such as noise or unstable feedback oscillations. The pair of papers by Engebretson and French-St. George describe the development and evaluation, respectively, of adaptive feedback equalization using an experimental wearable digital hearing aid. The results show that an additional 4 dB of gain could be achieved when adaptive feedback equalization was present, resulting in improved intelligibility.

Advanced signal processing techniques offer new opportunities for improving acoustic amplification, but it is important to know how to use this

new technology in order to achieve practical benefits. In many respects, the limitation on further progress is not so much that of not being able to process acoustic signals as needed, but rather that of not knowing what form this processing should take. The paper by Posen, et al., reports on new ways of processing speech for severely hearing-impaired individuals using frequency lowering. This form of signal processing has yielded mixed results in the past, improved perception being obtained for some speech sounds, and degraded perception for other sounds. In the study by Posen, et al., frequency lowering was used selectively depending on the acoustic-phonetic structure of the speech signal. For example, frequency lowering was not used whenever the input signal was dominated by low-frequency components, as typically occurs during semi-vowel and nasal consonants. The results of this investigation show improved performance for stops, fricatives, and affricates without degrading the perception of nasals and semi-vowels.

The last paper in this group, by Kates, addresses the issue of signal processing from a broad perspective, that of developing a general approach to the optimization of hearing aid processing by computer simulation of the impaired auditory system. From this vantage point, practical methods of signal processing for hearing aids can be optimized, as described in the paper. This approach represents a new way of thinking about the problem and provides a useful framework for developing improved methods of signal processing.

SECTION II. NEW METHODS OF NOISE REDUCTION

One of the most pressing problems to be addressed in the development of advanced signal processing hearing aids is that of reducing background noise. Methods of signal processing for noise reduction can be subdivided into two categories, single-microphone and multi-microphone techniques. Single-microphone techniques have received the greatest attention, presumably because conventional hearing aids have typically used only one microphone. The problem of single-microphone noise reduction is particularly difficult, however, and there is a long history of unsuccessful attempts at improving speech intelligibility in noise using a single microphone. The paper by Baer, et al., breaks this mold in that improvements in intelligibility using a single-microphone technique have been obtained. The basic approach is to enhance major peaks in the spectrum by increasing the differences in level between peaks and adjacent valleys. This method of spectral enhancement was combined with amplitude compression to achieve improvements in intelligibility equivalent to a gain of roughly 4 dB in speech-to-noise ratio.

Improvements in speech-to-noise ratio can be achieved much more easily using multi-microphone techniques. This is demonstrated in the paper by Bilsen, et al., who obtained improvements in speech-to-noise ratio of 7 dB using fixed- microphone arrays on the frame and legs of a pair of eyeglasses. Relatively simple methods of signal processing were used that could be incorporated in a practical hearing aid using existing technology.

The third paper in this group, by Kollmeier, et al., describes a binaural signal processing system combined with multiband amplitude compression. The binaural signal processor amplifies sounds emanating from the front and

suppresses sounds coming from the sides, including reverberation. The multiband compression algorithm matches the dynamic range of the amplified signal to that of the impaired auditory system. The two methods of signal processing were found to provide significant improvements in both speech intelligibility and sound quality.

SECTION III. DIGITAL TECHNIQUES APPLIED TO RELATED AREAS

Digital signal processing techniques have proven to be of value in a number of related applications. The paper by Dillier, et al., describes three separate applications of digital signal processing techniques that have been found to be useful for both hearing aids and cochlear implants. In the first application, a two-microphone noise reduction system has been implemented using an adaptive beam-forming algorithm in which a higher degree of directionality was achieved (i.e., the system behaves in much the same way as a highly directional microphone with concomitant improvements in the speech-to-noise ratio, provided the speech and noise come from different directions). In the second application, loudness distortions resulting from the hearing impairment are corrected using multiband amplitude compression. Although this form of compression is not new, its implementation in real-time using digital signal processing techniques based on relevant psychoacoustic data represents a useful step toward developing an improved multiband digital hearing aid. The third application describes an experimental evaluation of various signal stimulation strategies for use with multichannel cochlear implants. The results show that coding strategies involving the interleaving of pulsatile signals provide significant improvements and that coding strategies of this form for pitch information can be of use to a cochlear implant patient.

The next paper, by Wilson, et al., compared two approaches for representing speech information in a cochlear implant, a compressed analog representation and a digital representation using continuous interleaved sampling (CIS). The CIS procedure was found to yield substantial improvements over the compressed analog representation. The paper also reports on investigations for optimizing the parameters of the CIS procedure (e.g., pulse duration, pulse rate, interval between sequential pulses).

Methods of noise reduction developed for acoustic amplification systems can also be used with cochlear implants and other sensory aids. The paper by Weiss describes how the INTEL method of noise reduction has been implemented for use with the Nucleus-22 cochlear implant. The results of a computer simulation showed an effective improvement of about 7 dB in the speech-to-noise ratio. Subsequent experimental investigations with cochlear implant patients have obtained improvements in intelligibility corresponding to a 5 dB gain in intelligibility. These improvements are striking in that they were obtained with a single-microphone method of noise reduction. It is also interesting to note that previous evaluations of the INTEL method of noise reduction for acoustic amplification did not show any significant change in intelligibility, although improvements in overall sound quality were obtained.

The last paper in this special issue, by Larson, et al., deals with the application of digital technology to a difficult measurement problem, that of

wideband measurement of the acoustic impedance of the ear. Measurements of this type are useful not only for diagnostic testing but also in developing a quantitative description of the acoustic coupling between a hearing aid and the impaired auditory system. This information, in turn, can be used in developing more effective prescriptive procedures for hearing aids. The measurement of acoustic impedance over a wide frequency range is not a trivial problem, and this paper reports on an automated technique for obtaining acoustic impedance measurements and the reliability of such measurements obtained in two different laboratories by two different investigators.

This special issue has focused on issues of signal processing and its application to hearing aids and cochlear implants. A companion special issue, to appear shortly, will address more general aspects and clinical implications of this new technology.