TECHNOLOGY

Some Outstanding Issues in Hearing Aid Technology, Fitting and Research

By David A. Preves, Ph.D. and
Carol A. Sammeth, Ph.D.

New technology is constantly being introduced by hearing aid manufacturers, making this a very exciting time to be in the hearing health rehabilitation field. New hearing instruments incorporating state-of-the-art technology have produced many hearing aid fittings that are vastly superior to those with older technology. At present, however, new developments are sometimes placed on the market in a sort of "shotgun" approach; that is, without adequate guidance for how the technology should be applied. The question of candidacy for a particular device is often determined by trial and error because little is known about who among the hearing impaired might benefit from a new technological development.

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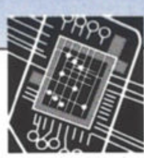

Unlike the prescription of some other medical devices, because of the lack of sufficient understanding of auditory system pathologies, the science of hearing aid fitting has not advanced to the level at which the identification of a specific pathology leads to the prescription of a hearing aid processing algorithm that is guaranteed to provide optimal benefit. No hearing aid algorithm has yet been found to be consistently superior for fitting one type of etiology or a particular audiometric configuration. For example, shown in Fig. 1b[35] are the speech recognition scores on a nonsense syllable test for two subjects with similar audiometric data (Fig. 1a) wearing a nonlinear hearing aid (adaptive high-pass filter plus expansion)[23] and a linear hearing aid with matched frequency response. Information obtained from the basic audiologic test battery is insufficient to predict which of these subjects would have performed significantly more poorly in noise with the linear than with the nonlinear signal processor.

The issue of candidacy may be further complicated by a lack of understanding of how a signal processing hearing aid functions and how to optimally set its parameters. Because of not having appropriate and sensitive electroacoustic techniques for testing nonlinear hearing aids, it has sometimes been difficult to determine what type of signal processing is being performed. This situation has been partially alleviated by testing hearing aids with a steady-state broad-band noise signal in accordance with a new ANSI standard[1]. For some hearing aids that change their frequency response as a function of input level, for example, this standard has been useful for more completely characterizing their steady-state electroacoustic behavior than testing them with a swept pure tone.

It is also clear that more appropriate behavioral protocols with sensitive and reliable test materials for evaluating hearing aids must be developed. This subject is receiving more attention because of the recent actions of the FDA in

David A. Preves, Ph.D., is vice president of research and development at Argosy Electronics. Carol A. Sammeth, PhD, is associate professor of audiology at Arizona State U, Tempe, AZ.

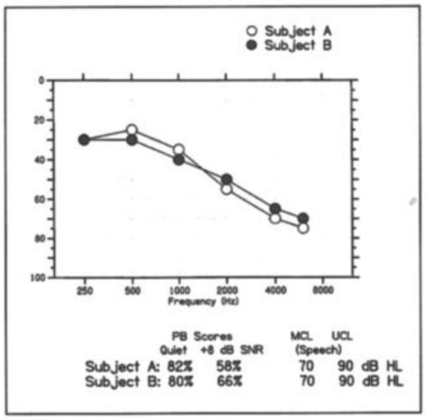

Fig. 1a *Audiometric data for two test subjects.*

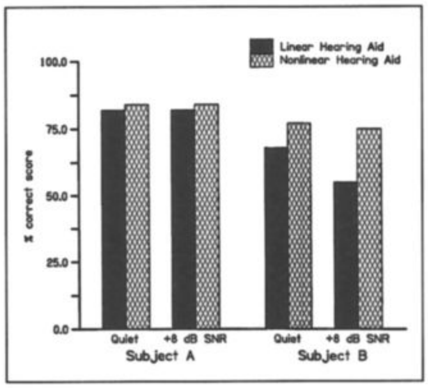

Fig. 1b *Speech recognition scores (for nonsense syllables) with a nonlinear hearing instrument (adaptive high-pass filtering plus expansion) versus a linear hearing instrument in quiet and in speech-spectrum noise.*[35]

regulating hearing aid marketing. In the future, it is likely that hearing aid manufacturers will be required to have data to support marketing claims about their products. Thus, development and marketing of new hearing aids may, by necessity, have to more uniformly follow a similar protocol to that used for higher-risk types of medical products.[e.g. 37]

Lack of a definitive selection and fitting process has perhaps contributed to the general "stigma" about amplification, and conceivably, therefore, contributed to keeping millions of hearing impaired persons from trying hearing aids.[15] Many first-time hearing aid wearers have apriori high expectations about what hearing instruments can accomplish, and, as a consequence, may experience post-fitting disappointment.[29] This disappointment is then spread by word of mouth to other prospective hearing aid wearers.

There are some areas in which technological advances in hearing aids are beginning to satisfy several important performance improvement needs. These include providing better sound quality and listener comfort with compression, adaptive frequency response circuits, and the use of push-pull and class D amplifier output stages[9, 14, 22]; finer resolution for shaping the frequency response with active multi-pole tone controls, wideband transducers and multi-band hearing aid amplifiers[28]; ensuring audibility of weak speech components[42]; and achieving more gain before the onset of acoustic feedback oscillation.[30] The most difficult task, however, has remained that of significantly improving speech recognition performance with hearing aids, particularly in the presence of background noise.

Problems in Improving and Assessing Speech Recognition in Noise

Attempts to improve speech understanding in the presence of background noise may be placed into two categories: reducing the noise itself and/or enhancing the speech signal.

Noise Reduction

Some of the approaches for noise reduction used in commercial hearing aids have been nonadaptive, such as directional microphones, fixed high-pass filters, and wearer-selectable multiple-modes in programmable hearing aids. These approaches have been somewhat successful, but a given fixed processing approach may only work well in a limited number of listening environments. However, a recent use of nonadaptive directional microphone arrays has shown promise for achieving large improvements in signal-to-noise ratio, even in reverberant environments.[31] As with single-microphone directional hearing aids, nonadaptive multi-microphone arrays rely on the line of sight of the hearing aid wearer to be in the same direction as the desired signal (unlike adaptive beamformers, the direction of their polar pattern nulls is not automatically changed). The problem of cosmetic packaging of such multi-microphone arrays (some with up to five microphones) may be the biggest challenge in getting them into production hearing aids.

At present, the majority of devices promoted as "noise reduction" hearing aids use forms of adaptive signal processing, including adaptive time constants, multiband compression, and adaptive filter responses. The research literature evaluating the benefits provided by these devices has been equivocal.[e.g. 27] Anecdotal evidence suggests, for example, that adaptive frequency response hearing aids that reduce low frequency gain automatically as input level increases (known as "BILL" hearing aids)[12]; may provide more comfortable listening in noisy environments. Significant benefit with these devices, however, has not always been demonstrated for a majority of subjects on objective speech recognition tasks.

Evaluation Issues

It is certainly possible that the negative results found in evaluations of some adaptive signal processors may have been due to the fact that they simply do not provide speech recognition improvements in noise (although there may be other benefits such as increased listening comfort). In at least some of these studies, however, the inability to demonstrate improved performance may have stemmed from the experimental protocols used. Inadequate detail is provided in some of these studies to be able to determine whether the equivocal performance results obtained were due to insensitive test methods, inappropriate protocol design, or the hearing aids themselves. Limitations of some research studies have included simulations of signal processors that do not accurately represent the function of commercial devices, too few subjects or the use of only group mean statistics, and the use of insensitive or unreliable speech test materials or procedures.

It is sometimes difficult to design a research experiment to evaluate underlying assumptions of signal processing approaches. For exam-

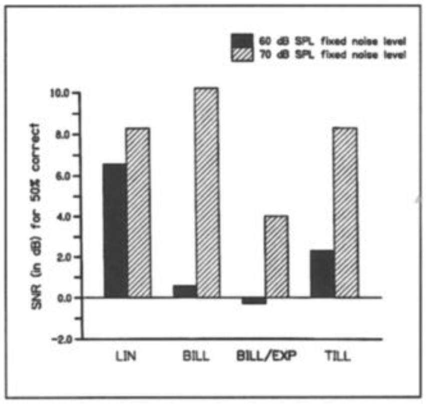

Fig. 2 *Performance of a hearing-impaired subject on an adaptive speech reception in noise test across several linear and nonlinear hearing aids. LIN = linear; BILL = Argosy Manhattan experimental; BILL / EXP = Manhattan experimental plus expansion; TILL = K-AMP. Speech-spectrum noise level was fixed; sentence level was varied for 50% correct identification[7].*

ple, one assumption in "BILL" type processing is that excess upward spread of masking thought to occur in cochlear-impaired ears at high input levels will be alleviated. Studies to examine this possibility by comparing masking patterns and performance for different frequency responses have either suffered from too much distortion in the linear mode to conduct the experiment[39] with actual hearing aids or have used fixed rather than adaptive low frequency reduction, often with laboratory equipment instead of hearing aid components.[4, 10, 26] This latter approach, while sometimes useful in exploring underlying concepts, does not fully simulate the time-varying adaptive nature of commercial "BILL" type hearing aids.

Perhaps one of the most critical issues in hearing aid research is the need to more closely examine individual as well as group mean differences in performance across hearing aids. Any benefit provided by a hearing aid being evaluated may be obscured by reporting only group mean data because usually some hearing aid wearers do better, some do the same, and some do more poorly with a given device relative to the reference condition. Because no hearing aid processing algorithm has been found as yet to be optimal for an identifiable set of patients, closer examination of individual results may lead to development of better candidacy criteria for particular devices. To this end, selecting the type of hearing aid processing and the fitting parameter values on an individual basis may be accomplished with single subject design.[32]

Another outstanding need in hearing aid fitting and research is the development of sensitive and reliable speech test materials and procedures. Many behavioral speech tests have proven insensitive in distinguishing differences in speech recognition performance between hearing aids.[33] For those speech tests that do appear to be relatively sensitive, more attention is needed to obtaining test-retest data to determine the reliability of the task, and to aid in examination of the significance of an individual subject's performance differences across hearing aid conditions.

One encouraging approach in evaluating speech recognition has been the use of various forms of speech reception threshold (SRT) in noise tests, which appear to be rela-

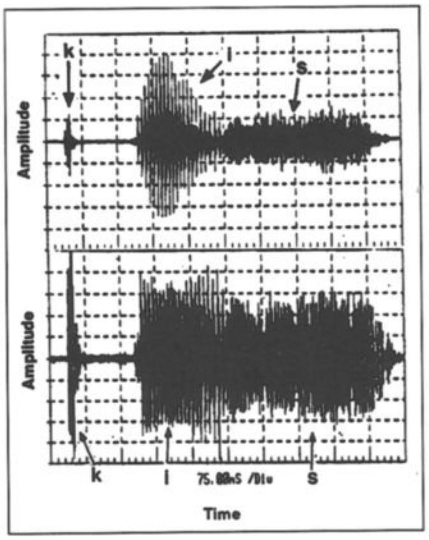

Fig. 3 *Effect of adaptive high pass filtering (Argosy Manhattan II ITE hearing aid) on amplitude of vowel and consonants in temporal speech waveform of phrase "..(you will mar/k /is/...". Upper: unprocessed input to hearing aid. Lower: processed hearing aid output on KEMAR.*

tively sensitive and reliable.[7, 13, 21, 40] In these tests, the noise level is held constant and the speech level is varied (or vice versa) until a certain percent correct score is obtained. Results can be displayed in percent correct as a function of signal-to-noise ratio (SNR) required. With this approach, performance can be evaluated at several speech and noise levels representing a range of quieter to noisier listening environments. Thus the adaptive nature of nonlinear aids can be more fully evaluated. An example is shown in Fig. 2[7] of performance of one hearing-impaired person across several linear and nonlinear hearing aids with an SRT- in-noise procedure (Hearing In Noise Test).[17] Plotted are the SNRs needed for 50% correct sentence recognition in a background of speech-spectrum noise fixed at 60 dB SPL and 70 dB SPL (note that a smaller or negative number indicates better performance). For this subject, best performance was obtained with both the "BILL" and the "BILL"/EXP at a 60 dB SPL noise level, but only with the "BILL"/EXP at a 70 dB SPL noise level.

Another means for determining speech recognition (or perceptual sound quality) is to use subjective judgments by the hearing aid wearer. This can be accomplished with magnitude estimation[16] or paired comparisons.[25] Sensitivity and reliability of subjective judgment tasks can vary depending upon the speech material and specific procedures employed. Improved ability for daily communication needs can be substantiated with reliable questionnaires such as the Profile of Hearing Aid Benefit (PHAB),[6] Profile of Hearing Aid Performance (PHAP)[5] and the Hearing Aid Performance Inventory (HAPI).[41]

Finally, in assessing the effectiveness of hearing aid fittings, particularly with respect to speech recognition in noise, the issue of "acclimatization" to amplification may be a consideration for some listeners. Some researchers have reported that patients perform better with a hearing aid after a sufficient period of wearing time, presumably because they become more able to use any new acoustical information that the hearing aids may provide.[8] The importance of this effect is still unclear, however, as Bentler, et al[3] found that performance of hearing aid wearers with several types of signal processing algorithms did not change significantly over time.

Speech Enhancement

A wide variety of futuristic speech enhancement algorithms have been tried in research laboratories, mostly implemented with digital signal processing. A sufficient suprathreshold presentation level for important speech cues is obviously a prerequisite for good speech recognition; however, simply ensuring the audibility of speech energy will not always result in maximal performance for every patient. Some researchers have suggested that perceptual distortions inherent in the damaged cochlea, as reflected in losses in temporal and frequency resolution, may reduce a patient's ability to utilize available speech cues.[20] Cognitive deficits in the elderly may also be a factor.[38] The goal in speech enhancement is to compensate for these perceptual distortions by intentionally manipulating certain speech cues so that they can be more easily used by the impaired ear.

Experimental approaches for speech enhancement have included increasing consonant-to-vowel ratios (CVRs), spectral sharpening, temporal modulation envelope modification or duration manipulation.[24, 43] The task of speech enhancement is complicated by the fact that a certain enhancement approach may work well for some phonemes but not for others. Recent research, for example, has suggested that moderate enhancement of the CVR of some syllables may improve speech recognition performance for some patients.[11] Regardless of what algorithms are used to accomplish speech enhancement (or environmental noise reduction), it is important to do so without introducing audible artifacts so that the hearing aid-processed signals do not sound further distorted or artificial.

The greatest difficulty in developing viable speech enhancement algorithms lies less in developing the technology than in determining which patients will benefit from which speech enhancement approaches. To do this, we will need to further differentiate cochlear impairments based on deficits in suprathreshold processing abilities. Several researchers have suggested that modified psychoacoustic and/or speech perceptual tests may need to be incorporated into evaluations for fitting nonlinear hearing aids in the future.[e.g., 18, 36] As discussed by Ochs,[18] any task incorporated into clinical procedures will need to be time-efficient, provide data that is not available from the basic audiologic test battery, and serve as a prognostic indicator for user success with a given signal processing approach. This will certainly not be an easy task. At present it is still unclear how differences in performance on psychoacoustic tasks relate to losses in speech recognition ability, and how to determine which cues in speech are most important to a given listener's performance.

As more sophisticated nonlinear signal processing algorithms appear on the market, there will be a need for new approaches to characterize them electroacoustically. In this light, the use of temporally-varying complex signals has been advocated in order to determine how well hearing aids function with a more speech-like signal.[2] Researchers have begun to more fully examine the waveforms of processed speech at the output of nonlinear hearing aids in order to determine possible effects on speech perception.[19,34] For example, as shown in Fig. 3, examination of the output of an ITE hearing aid with "BILL" type processing (lower trace) for the syllable /is/ reveals a higher CVR than found in the same stimulus unprocessed (upper trace).

Conclusions

As more knowledge of the interaction between pathological auditory systems and hearing aid processing algorithms is gathered, results for patients with a given etiology or with specific auditory system characteristics may hopefully be extended to predict the performance of other patients with similar characteristics. In order to realize that goal, better methods are needed to evaluate both the electroacoustic function of nonlinear hearing aids and behavioral performance with these devices. The availability of such methods, combined with an increased understanding of the auditory system, may also lead to the development of better hearing aid signal processing algorithms in the future. ◆

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