A Review of Digital Technology In Hearing Aids

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As digital technology is beginning to be incorporated into hearing aids, audiologists should be aware of the advantages and disadvantages of digital applications in the area of hearing aids. This paper provides: (a) an overview of the components of a successful hearing-aid-fitting strategy, with emphasis on how digital manipulations of speech signals might affect our thinking in this area; (b) a brief summary of the status of current analog hearing aids; (c) a short primer in digital signal processing; and (d) a description of some digital-analog solutions to purely analog hearing aids.

Walden (1982) described the stages of successful hearing aid fitting. Stage 1 is the identification and description of the hearing loss of the hearing aid candidate via a case history and audiometry. The validity of much of the audiometric information usually collected to fit a hearing aid has been suspect for many years (Chial & Hayes, 1974). Recently, Jesteadt (1985), Lufti (1985), Nelson (1985), Perkins and Wrightman (1985), and Stelmachowicz (1985) described abnormalities of impaired ears that could not be described directly using current clinical audiometric procedures. These included frequency resolution and selectivity, auditory nonlinearities, intensity relations, processing, masking effects, and the relationship between psychoacoustic tuning curves and speech perception. Knowledge in these areas will affect the results of Stage 1 and, in turn, limit the applicability of digital technology. Without adequate audiologic information, the potential of digital technology will remain latent and unavailable to the hearing-impaired individual. Clinical measures that more completely describe hearing-impaired persons' auditory capabilities and deficits need to be researched. The availability of technology may act as a catalyst in this area.

Walden's (1982) Stage 2 is the use of a specific hearing aid selection scheme including comparative methods, subjective judgments, and prescriptive formulae. Digital technology has the potential to be particularly useful in this area, given the validity of the information gathered in Stage 1. Several hearing aid evaluation systems have been developed and tested. Signor (1985) outlined many functioning systems and some that are still on the drawing boards. Some of the prospective systems will require re-thinking the hearing aid fitting process. The history of audiology has often seen the development of a better mouse trap that was little used because of resistance to change or difficulty understanding the concepts which made the trap better. Signor (1985) suggested that we will continue to be confronted with a need to evaluate our current practices.

Stage 3 (Walden, 1982) is hearing aid orientation and training to improve patients' communication with a hearing aid. Choices in Stage 3 are very dependent on the outcomes of Stages 1 and 2. Here, as well, Signor (1985) provided some insight into how our current procedures might be changed by digital technology. For example, Montgomery (cited in Signor, 1985) suggested that aural rehabilitation make use of speech enhancement techniques. Digital speech enhancement might take the form of a microprocessor system which would pick out consonants from an incoming message, stretch them out in time, and intensify them to give the hearing-impaired person a better opportunity to extract relevant acoustic features of speech. Development of such a speech enhancer would require an integration of technology with current knowledge of speech perception. Do we know now if an enhancement strategy would work for all hearing-impaired individuals, or only a subset? By what criteria can we decide to implement such signal manipulation? These are important research questions that must be answered if we are to use digital technology in the clinic.

Stage 4 (Walden, 1982) is the assessment of the benefit that patients obtain from a hearing aid in daily life. Typically this involves the use of a hearing handicap scale (Walden, 1982), the results of which may trigger some modification of the hearing aid fitting. The danger at Stage 4 is assuming that the prior three stages have been accomplished successfully. To aid in Stage 4, Signor (1985) suggested a programmable hearing aid whose characteristics might be changed by modifying a computer program even over a telephone line. There already are programmable speech processors, but size, power requirements, and speed of processing are problems (Levitt, 1985). Currently available microprocessors that are fast enough to perform this task are not appropriate because they are too large and require current not easily packaged in a wearable device (Plaut & Hecox, 1985; Preves, 1985). The rapid development of low-current drain complementary metal oxide semiconductors (CMOS), and optimization of component packaging through the use of large scale integration (LSI) technologies, may soon overcome these problems (Preves, 1985; Staab, 1985). Other limited, but smaller wearable units have been created as part of the development of cochlear implant devices (Kriewall, 1985). Implementation in hearing aids would depend on the development of a valid hearing aid fitting protocol and would require highly developed digital signal processing techniques. Neither appears to be on the horizon.

The purpose of this paper is to describe the status of hearing aid technology and some of the problems. Included is a short primer on digital signal processing, a description some of the analog-to-digital and digital solutions to problems encountered by hearing aid wearers, and speculations on the impact of this technology on hearing aids in the near future.

CAPABILITIES AND LIMITATIONS OF CONVENTIONAL HEARING AIDS

The modern hearing aid is no longer a bulky instrument with poor fidelity (Sigelman, 1985). Killion and Tillman (1982) showed that hearing aids can be produced with fidelity that is rated superior to that of commercial stereo systems, at least by normal-hearing listeners. In addition to being capable of providing uniform amplification through a wide range of frequencies, hearing aids have been designed which (a) can provide differential amplification for various frequency ranges; (b) compress the amplitude range of an incoming signal; (c) compress the frequency range of an incoming signal; (d) transpose signal frequencies; and (e) reduce the relative output level of sounds coming from a particular direction. All of these design elements are intended to compensate for the auditory perceptual deficits that are typically measured and assumed to be present in hearing-impaired individuals. These may include elevated thresholds, reduced dynamic range, abnormal frequency discrimination and selectivity, impaired temporal resolution, and poor localization ability (Scharf & Florentine, 1982).

Underlyng the process of hearing aid fitting is the assumption that the purpose of the hearing aid is to make everyday conversational speech audible and understandable to the hearing-impaired individual without exceeding that person's uncomfortable loudness level (Skinner, Pascoe, Miller, & Popelka, 1982). For many hearing-impaired individuals, making speech audible is not simply a matter of providing uniform gain across frequencies. Differential amplification across frequency is often needed (Levitt, 1985; Weiss, 1985); however, the best method for determining optimal gain at each frequency for an individual is an unsettled issue. Even if audiologists had a definitive means of prescribing frequency-specific gain, in most cases it would not be possible to find an analog hearing aid whose frequency response matched the prescription. Typically an aid is selected which comes closest to matching the prescription although frequency response curves do not differ significantly from one manufacturer to another.

The use of vents in earmolds has proven to be a convenient and economical means to modify frequency response. Cox and Alexander (1983) indicated that venting also produces a more pleasing sound for hearing aid users with high frequency losses. Unless carefully done, venting can cause a problem in the form of acoustic feedback.

One of the most common complaints of hearing-impaired individuals is that the hearing aid does not perform well in noise. In studies assessing hearing aid satisfaction in various listening situations, distinct differences have been found between reported hearing aid benefit in noise and in quiet (Kapteyn, 1977a, 1977b; Scherr, Schwartz, & Montgomery, 1983; Walden, Demorest, & Hepler, 1984). In situations characterized as noisy (such as parties and meetings), performance with most hearing aids is consistently rated poorer than it is in quiet. Common solutions to this problem include FM auditory trainers, binaural amplification, reduced amplification in the low frequencies (Gordon-Salant, 1984), and incorporation of directional microphones into the hearing aid (Hawkins & Yacullo, 1984). None of these solutions are without frequent exception. For example, Schreurs and Olsen (1985) reported that the majority of their 30 subjects preferred a monaural aid to binaural aids in the presence of competing noise.

In summary, although current hearing aids can provide benefit for hearing-impaired individuals, several serious technical problems remain to be solved: (a) adjustable frequency-specific gain, (b) the detrimental effects of environmental noise on speech perception, and (c) the occurrence of acoustic feedback at high hearing aid gain settings and with fittings utilizing large vents (Preves, 1982). A potentially more serious problem relates to our limited knowledge of how to manipulate a speech signal so that it maximizes the usefulness of an individual's impaired auditory-linguistic system. For example, it is not clear how a signal should be manipulated to compensate for abnormal effects of masking on speech perception, abnormal frequency or intensity resolution, or abnormal central auditory processing of important acoustic features of the speech code. With the development of digital algorithms to model the auditory system (Lyon, 1985), we may obtain a better understanding of the normal and impaired auditory-linguistic system. Full implementation of digital technology must await development of many of these rules for prescribing signal processing parameters.

A PRIMER ON DIGITAL SIGNAL PROCESSING

Digital signal processing (DSP) refers to the creation, changing, and/or detection of signals using digital technology (Staab, 1985). DSP is a sampling technique and not a hardware device. One of its functions is to eliminate the need for conventional analog components (e.g., transistors, resistors, capacitors, diodes). In hearing aid design, DSP uses software to emulate the functions of components such as filters, limiters, oscillators, modulators, and demodulators.

Regardless of the type of processing or application, DSP signifies certain basic procedures. DSP changes an analog waveform, such as voltage from a microphone, from a continuous signal into a sequence of binary numbers. These numbers represent estimates of voltage at specific points in time. It should be noted that only 0 or 1 are used in binary numbers. Figure 1 illustrates the manner in which an electrical voltage can be converted into a series of binary numbers. A computer can perform arithmetic operations on these numbers according to certain rule sets or algorithms. Just as analog devices can be designed to modify a signal in certain ways, algorithms can be designed to change the signal in specific ways as well, with an even greater degree of accuracy than the analog instruments. For example, algorithms are available for gating, peak or center clipping, rectification, frequency or noise filtering, and intensity and duration measurement. In addition to performing manipulations of the acoustic signal, DSP will allow or may even require us to consider new concepts of what is important in the acoustic signal for understanding speech.

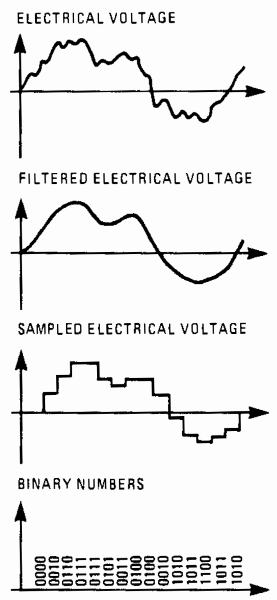

Figure 1. Schematic of how an electrical analog signal is digitally sampled and represented in binary numbers (from Staab, 1985).

Analog-to-Digital Conversion

The first step in preparing an analog signal for digital processing is to convert it from its analog form into a digital form. This is known as analog-to-digital conversion (A/D) and is carried out by a circuit board or chip known as the A/D converter. The A/D process involves two steps. The first step is sampling and the second is quantization of each sample. In actual operation these two steps take place simultaneously rather than sequentially.

In order to sample a signal at some predetermined rate, a device within the A/D converter or the computer acts as a clock to direct the A/D converter to sample the waveform at equal time intervals. The voltage obtained at each sample is then changed into a numerical value expressed as a binary number. The following section examines this process in more detail.

Sampling. Sampling rate is the number of times per second that a waveform will be sampled. It is chosen on the basis of how much distortion is acceptable and how much computer memory is available (Staab, 1985). The more rapid the sampling rate, the more pieces of information there are to store in the computer's memory. When the sampling rate is less than two times the highest frequency in the signal, the frequencies which are greater than two times the sampling rate will be shifted downward to a lower frequency. This distortion of the signal is called aliasing.

Figure 2 illustrates several important concepts of aliasing. Two waveforms (A and C) of equal frequency are presented. The vertical lines represent the points in time at which the A/D converter took a sample of the analog waveform. The interval between sample points, that is, X_1 to X_2 or Y_1 to Y_2, are Nyquist intervals. Waveforms B and D are reconstructions of waveforms A and C respectively. It is apparent that waveform B has the same fundamental frequency as waveform A from which it was derived. Seven samples were taken within each period of waveform A, exceeding the recommended minimum number of samples (two per period). Thus, the frequency of the original waveform is not shifted. The example illustrated by waveforms C and D shows the effect of too slow a sampling rate, less than two samples per period. In this case, only one sample per period was obtained. The result, waveform D, shows a frequency lower than that of the original waveform.

Aliasing in a complex waveform is shown in Figure 3. In this example, the higher harmonics of waveform A are omitted in the reconstructed waveform B. This occurred even though the sampling rate was at least two times as fast as the fundamental. However, the sampling rate was not two times faster than the harmonic which was present in the original waveform. Consequently, the fundamental frequency is maintained but the harmonics are lost. The greater the number of samples, the truer the reconstruction of the initial analog waveform, because the waveform between sample points is unknown and must be interpolated.

In order to prevent aliasing, there are two alternatives. One alternative is to make the sampling rate at least two times as fast as the highest frequency in the signal. For even finer reproduction of the analog waveform, the sampling rate should be on the order of 2.5 to 3 times as fast. The faster a waveform is

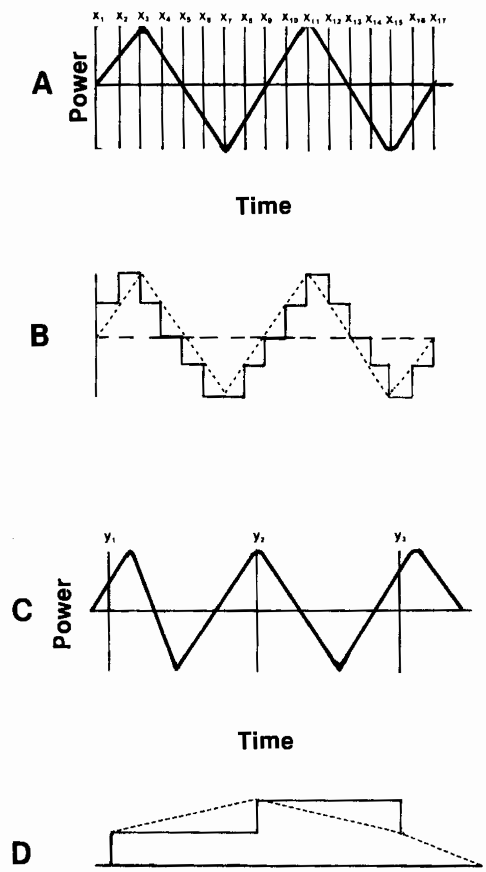

Figure 2. Analog waveforms A and C of equal frequency are shown digitally sampled in two ways. Waveform B resulted from sampling at a rate exceeding two times the fundamental frequency (no aliasing). Waveform D shows the effects of sampling at less than twice the rate for adequate reconstruction, producing aliasing and a much lower fundamental frequency than waveform C.

sampled, however, the more computer memory is required to store it. Thus, it is usually not reasonable to sample the entire signal fast enough to reconstruct its highest frequency. Input signals are usually band limited by a presampling, analog, low-pass filter and/or are shortened to limit the length of the signal to be digitized.

Quantization. When a waveform is sampled, the A/D process assigns a numerical value to the sample. The numerical value is a binary representation of an estimate of the signal strength at the point in time the sample was taken. Thus, the continuous analog waveform is transformed into a series of rounded-off values, sampled at uniform time intervals, and consecutively stored in computer memory (Staab, 1985).

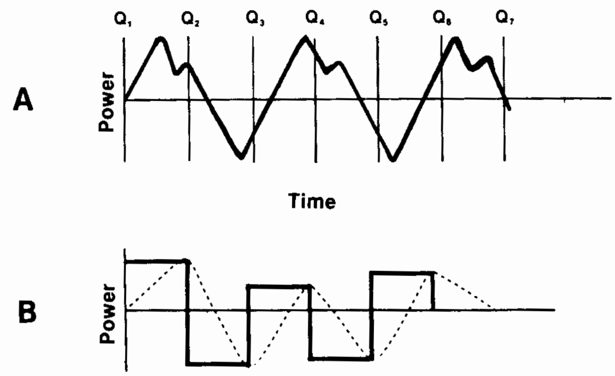

Figure 3. Sampling error due to aliasing in complex waveform A. The fundamental frequency of waveform A is adequately reconstructed in waveform B, but the harmonics of waveform A are omitted in waveform B. This happened because while the sampling rate was twice the fundamental of waveform A, it was less than twice the frequency of the harmonics of waveform A.

There are several methods of quantization of an analog sample. The amount of quantization noise or distortion obtained will differ according to the number of quantization steps. Figure 4 illustrates the quantization error resulting from sampling with an 8-bit and 4-bit quantizer. $S_1, S_2, \ldots, S_8$ are the Nyquist samples. Q represents a quantization level around some reference point, zero. Each quantization level, such as Q or -Q, is a discrete estimate of the voltage at the S_i point in time. Each quantization level is called a bit. The amount of voltage in the signal is rounded to the nearest quantization level. If the voltage exceeds either +Emax or -Emax, then the signal is rounded to the nearest quantization level. The lower part of the figure illustrates the error that is introduced into the digitized sample by the estimating process. By comparing the 8-bit quantizer with the 4-bit quantizer, we can see clearly that, when more quantization levels are used, the distortion, or error, is less.

In summary, the process of sampling and quantization of an analog waveform in order to digitize it can introduce error and, therefore, noise into the signal. Rabiner and Gold (1975) described mathematical procedures to measure the signal and the noise output so that a signal-to-noise ratio can be obtained. As a rule of thumb, Plaut and Hecox (1985) suggested that each bit of the analog-to-digital converter capacity improves the signal-to-quantization noise-ratio by 6 dB. For most practical applications, a 10-bit converter is sufficient because the quantization noise level is more than 60 dB below the signal level. At the least, one would want to keep quantization noise levels below ambient noise levels.

Digital-to-Analog Conversion. After the signal has been processed by the algorithms within the computer, the digital representation of the signal must be changed back into analog form. This conversion is accomplished by the

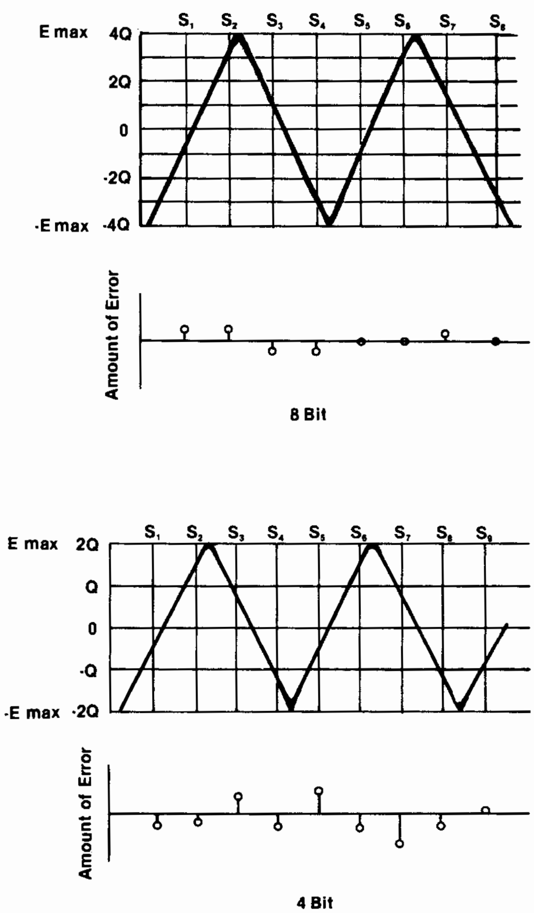

Figure 4. Quantization error is shown as a result of sampling with an eight and four bit quantizer. The larger the number of bits, the smaller the error.

digital-to-analog (D/A) converter. The D/A converter is a device that operates on the digital representation of the signal to reconstruct the original analog waveform. As a result of the D/A process, the analog waveform is not represented as a smooth function. The fewer bits used, the less like the analog signal is the resulting digital representation. This is referred to as quantization noise. Eight to twelve bits appear to adequately represent speech signals with acceptable quantization noise (Plaut & Hecox, 1985). What noise remains is usually handled by a low pass filter following the D/A converter which smooths the output of the digital section, removes the noise, and provides for accurate reproduction of intelligible speech (Rabiner & Gold, 1975; Staab, 1985). The whole analog-to-digital-to-analog process just described is shown schematically in Figure 5.

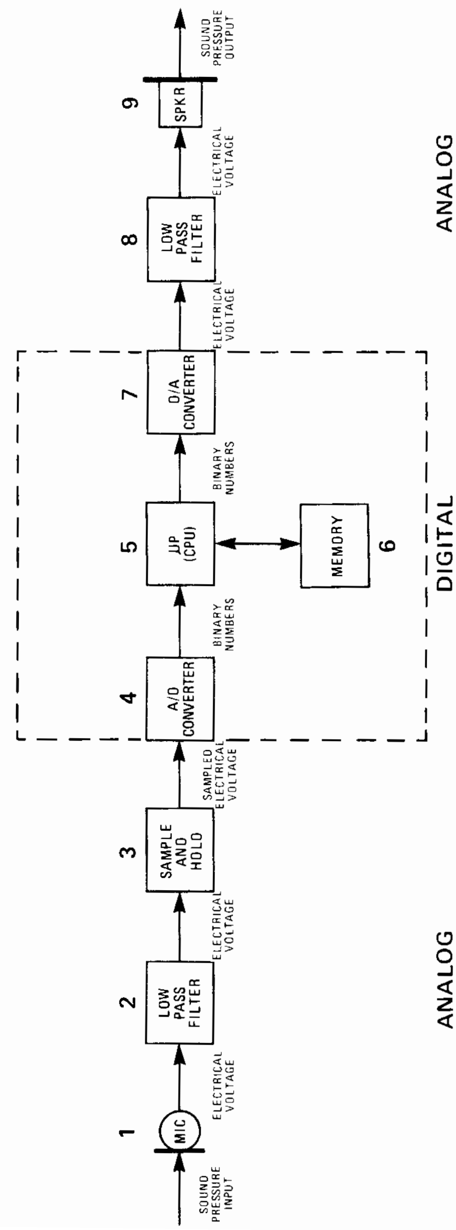

Figure 5. Summary of the stages involved in analog-to-digital-to-analog conversions required for digital signal processing (from Staab, 1985).

STATUS OF DIGITAL HEARING AIDS

The incorporation of digital technology into hearing aids may overcome the limitations of purely analog hearing aids as cited in the introduction. A microcomputer can quickly perform sophisticated mathematical operations and can conduct a number of functions in a short period of time. This creates the possibility of the hearing aid itself performing signal processing to compensate for hearing aid users' impairments in auditory signal processing. Algorithms have already been developed to suppress pure tones (Weiss, 1985) and continuous noise (Graupe & Causey, 1977) and to perform manipulations on the spectrum of the incoming speech signal (Levitt, 1985). Most of these systems consume more power and are currently larger than would be practical for hearing aid applications.

The only application of digital technology to hearing aids expected to be commercially available in the near future is the Graupe-Causey Self-Adaptive Filter (GCAF) (Graupe & Causey, 1977). This filter is designed to adaptively filter near-stationary noise out of a speech signal. That is, the frequencies of environmental noises of relatively long duration (more than about 3 seconds) and with frequency spectra that do not vary with time are attenuated in the incoming signal when and only when such noise occurs. The noise produced by cars, fans, and small engines is an example of targets of this particular filter.

The filter itself consists of a recognition subsystem and an adaptive filter subsystem. The recognition subsystem identifies the parameters of the incoming signal during each sampling period and then examines them to determine if they are stationary (similar from sample to sample). During speech intervals, the signal will be identified as not stationary. During pauses in speech when no noise occurs, the incoming signal will be below the pre-determined threshold for activation of the filter. Therefore, for these two conditions of speech and speech pauses, the filter will not be activated. During the pauses containing noise, the signal will exceed the threshold and be determined to be stationary; the frequency characteristics of the noise will then be conveyed to the adaptive filter. The filter will then attenuate the noise signal before passing it to the hearing aid amplifier. According to Stein and Dempesy-Hart (1984), if both speech and noise occupy the same frequency range, the filter will pass both speech and noise at some frequencies and reject both speech and noise at other frequencies.

As the characteristics of the noise change, they are identified during each pause and conveyed to the adaptive filter which adjusts the frequencies passed to block out the noise. A noise cessation circuit within the adaptive filter serves to detect the termination of noise within a speech interval and to allow the signal to pass unfiltered through the system. Thus the self-adaptive filter adapts not only to the spectral characteristics of near-stationary noise but also to the time course of that noise.

Stein and Dempesy-Hart (1984) assessed the effects of the GCAF on the intelligibility of NU-6 words in noise for five normal-hearing and 14 hearing-impaired subjects. The prototype adaptive filter used in this study was contained within a body-style hearing aid. Improvements were seen for the filter-on condition over the filter-bypassed condition.

As mentioned earlier, algorithms have also been developed which suppress pure tone stimuli. If hearing aids can be designed which eliminate tonal stimuli, the fitting problems created by acoustic feedback can be resolved.

Another advantage of digital technology within the hearing aid would be the capability of shaping the frequency response to conform to each individual's hearing loss. The amount of gain required at each frequency could be set independent of the gain at other frequencies. This would lessen the difficulties posed by hard-to-fit configurations as mentioned above. Individually-shaped frequency responses would also make the prescription method of selecting hearing aids more exact. At the present time, the audiologist determines the amount of gain desired at each frequency and then chooses the model of hearing aid which most closely matches this prescription. This process obscures subtle differences between different hearing aid prescription formulae and introduces error into the fitting procedure. With digital hearing aids, the fitting of hearing aid characteristics could be as precise as the methods of measuring hearing loss and hearing aid characteristics.

Digital technology in hearing aids would also permit compression characteristics to be set at each frequency independently. As a whole, research evaluating the advantage of multi-channel compression has been inconclusive. Most studies have not demonstrated superiority for compression systems (Abramovitz, 1980; Barfod, 1976; Lippmann, Braida, & Durlach, 1981). Other studies, however, have indicated potential benefit from multi-channel compression (Villchur, 1982; Laurence, Moore, & Glasberg, 1983). Braida, Durlach, Degennaro, Peterson, and Bustamante (1982) reviewed the problems with past studies of multi-channel compression and suggested reasons why these studies have been unsuccessful in realizing an advantage for multi-channel compression. The consensus of researchers in this area is that the potential for showing an advantage of multi-channel compression should not be ruled out. In their present analog form, experimental multi-channel compression aids add bulk to a hearing aid. Digital circuitry would allow multi-channel compression to be programmed into the hearing aid, if desired, without increasing the size of the aid.

Frequency-transposing or frequency-lowering hearing aids have been designed to pre-process the speech signal for listeners with high-frequency hearing impairment. Braida et al. (1980) reviewed many of the studies on these aids, most of which have not yielded promising results. Braida et al. attributed the negative results, in part, to the design of the aids. It is difficult to selectively alter the frequency characteristics of a signal in an analog hearing aid without introducing concurrent distortions in other essential aspects of the signal.

With digital hearing aid circuitry, however, it would be possible to change the variables of interest independently. This will permit isolation of the particular manipulations which would assist an individual hearing-impaired listener in speech recognition.

In summary, digital hearing aids present the prospect of improving signal-to-noise ratios for the hearing aid user, giving audiologists the opportunity to uniquely select amplification characteristics for each individual, and, ultimately, allowing signal processing of speech signals to aid hearing-impaired individuals in recognizing speech. The use of this extreme amount of flexibility afforded audiologists in programming hearing aid characteristics will, however, require a better understanding of which hearing aid characteristics are needed by each hearing-impaired individual.

SUMMARY AND IMPLICATIONS OF DIGITAL SIGNAL PROCESSING RESEARCH IN HEARING AIDS

In the previous sections, we indicated that analog hearing aid design has not been able to solve many of the problems encountered by hearing-impaired listeners, especially in noise. DSP algorithms (a step-by-step flow chart for the solution of a problem) can be quite simple or very complex. Algorithms can make rapid adaptive changes to DSP parameters through adaptive filtering to prevent acoustic feedback and to improve the signal-to-noise ratio of amplified sound in certain types of noise. While DSP parameters are changeable in such implementations, the range of change and number of changes possible are limited at this time. Each day, however, the number of changes that are possible grows.

Full implementation of DSP would permit many speech signal changes to occur simultaneously, each within a very large range of values. Another name for an algorithm might be a program; thus, the changes and ranges would be programmable for each individual hearing-impaired person. Although the required algorithm formulations and computer capabilities are not yet currently available for use in wearable amplification, they are in the realm of current laboratory research.

Continued work in DSP also raises a philosophical consideration. As we are challenged to develop algorithms and computer technology to implement models of impaired auditory-linguistic systems, we may need to view the systems in new ways that provide insight into their function and dysfunction beyond our current state of knowledge. We paraphrase what Einstein said of the first atomic explosion: Everything has changed, save man's thinking. The first DSP explosions have taken place, and it is up to us to determine if our audiologic concepts have been changed or not.

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REFERENCES

Abramovitz, R.S. (1980). Frequency shaping and multiband compression in hearing aids. *Journal of Communication Disorders, 13,* 483-488.

Barfod, J. (1976). *Multichannel compression hearing aids: Effects of recruitment on intelligibility* (Rep. No. 11). The Acoustics Laboratory, Technical University of Denmark.

Braida, L.D., Durlach, N.I., Degennaro, S.V., Peterson, P.M., & Bustamante, D.K. (1982). Review of recent research on multiband amplitude compression for the hearing impaired. In G.A. Studebaker & F.H. Bess (Eds.), *The Vanderbilt hearing-aid report.* Upper Darby, PA: Monographs in Contemporary Audiology.

Braida, L.D., Durlach, N.I., Lippmann, R.P., Hicks, B.L., Rabinowitz, W.M., & Reed, C.M. (1980). *Hearing aids — A review of past research on linear amplification, amplitude compression, and frequency lowering.* Rockville, MD: ASHA Monographs.

Chial, M., & Hayes, C. (1974). Hearing aid evaluation methods: Some underlying assumptions. *Journal of Speech and Hearing Disorders, 39,* 270-279.

Cox, R.M., & Alexander, G.C. (1983). Acoustic versus electronic modifications of hearing aid low-frequency output. *Ear and Hearing, 4,* 190-196.

Gordon-Salant, S. (1984). Effects of reducing low-frequency amplification on consonant perception in quiet and noise. *Journal of Speech and Hearing Research, 27,* 483-493.

Graupe, D., & Causey, G.D. (1977). Method of and means for adaptively filtering near-stationary noise from speech [United States Patent No. 4,025,721].

Graupe, D. (1985, May). *Use of adaptive filters in amplification devices.* Paper presented at the Conference on Signal Processing Strategies for the Hearing Impaired, Madison, WI.

Hawkins, D.B., & Yacullo, W.S. (1984). Signal-to-noise ratio advantage of binaural hearing aids and directional microphones under different levels of reverberation. *Journal of Speech and Hearing Disorders, 49,* 278-286.

Jesteadt, W. (1985, May). *Frequency selectivity and auditory nonlinearities.* Paper presented at the Conference on Signal Processing Strategies for the Hearing Impaired, Madison, WI.

Kapteyn, T.S. (1977a). Satisfaction with fitted hearing aids. I. An analysis of technical information. *Scandinavian Audiology, 6,* 147-156.

Kapteyn, T.S. (1977b). Satisfaction with fitted hearing aids. II. An investigation into the influence of psycho-social factors. *Scandinavian Audiology, 6,* 171-177.

Killion, M.C., & Tillman, T.W. (1982). Evaluation of high-fidelity hearing aids. *Journal of Speech and Hearing Research, 25,* 15-25.

Kriewall, T. (1985, October). *Possible future directions of cochlear prostheses.* Paper presented at the Meeting of the Minnesota Speech-Language-Hearing Association, Minneapolis, MN.

Laurence, R.F., Moore, B.C.J., & Glasberg, B.R. (1983). A comparison of behind-the-ear high-fidelity linear hearing aids and two-channel compression aids, in laboratory and in everyday life. *British Journal of Audiology, 17,* 31-48.

Levitt, H. (1985, May). *Use of computer simulation in hearing aid research.* Paper presented at the Conference on Signal Processing Strategies for the Hearing Impaired, Madison, WI.

Lippman, R.P., Braida, L.D., & Durlach, N.I. (1981). Study of multichannel amplitude compression and linear amplification for persons with sensorineural hearing loss. *Journal of the Acoustical Society of America, 69*(2), 524-534.

Lufti, R. (1985, May). *Masking in the hearing impaired.* Paper presented at the Conference on Signal Processing Strategies for the Hearing Impaired, Madison, WI.

Lyon, R. (1985, May). *Computational models of hearing.* Paper presented at the Conference on Signal Processing Strategies for the Hearing Impaired, Madison, WI.

Nelson, D. (1985, May). *Abnormalities of frequency resolution in impaired listeners.* Paper presented at the Conference on Signal Processing Strategies for the Hearing Impaired, Madison, WI.

Perkins, M., & Wrightman, F. (1985, May). *Intensity processing.* Paper presented at the Conference on Signal Processing Strategies for the Hearing Impaired, Madison, WI.

Plaut, D., & Hecox, K. (1985, May). *Introduction to digital signal processing and demonstration of algorithms on a single chip signal processor.* Paper presented at the Conference on Signal Processing Strategies for the Hearing Impaired, Madison, WI.

Preves, D. (1982). The potential of computers and signal processing for hearing aids. *Hearing Instruments, 31* (7).

Preves, D. (1985, October). *Application of signal processing for hearing aids.* Paper presented at the Meeting of the Minnesota Speech-Language-Hearing Association, Minneapolis, MN.

Rabiner, L.R., & Gold, B. (1975). *Theory and application of digital signal processing.* Englewood Cliffs, NJ: Prentice Hall.

Scharf, B., & Florentine, M. (1982). Psychoacoustics of elementary sounds. In. G.A. Studebaker & F.H. Bess (Eds.), *The Vanderbilt hearing-aid report* (pp. 3-15). Upper Darby, PA: Monographs in Contemporary Audiology.

Scherr, C.K., Schwartz, D.M., & Montgomery, A.A. (1983). Follow-up survey of new hearing aid users. *Journal of the Academy of Rehabilitative Audiology, 16,* 202-209.

Schreurs, K.K., & Olsen, W.O. (1985). Comparison of monaural and binaural hearing aid use on a trial period basis. *Ear and Hearing, 6,* 198-202.

Sigelman, J. (1985, October). *The future of custom aids.* Paper presented at the Meeting of the Minnesota Speech-Language-Hearing Association, Minneapolis, MN.

Signor, M.B. (1985). Hearing aids: Future prospects. *ASHA, 27*(7), 29-32.

Skinner, M.W., Pascoe, D., Miller, J.D., & Popelka, G.R. (1982). Measurements to determine optimal placement of speech energy within the listener's auditory area: A basis for selecting amplification characteristics. In G.A. Studebaker & F.H. Bess (Eds.), *The Vanderbilt hearing-aid report* (pp. 161-169). Upper Darby, PA: Monographs in Contemporary Audiology.

Staab, W. (1985). Digital hearing aids. *Hearing Instruments, 31* (7), 14-25.

Stein, L.K., & Dempesy-Hart, D. (1984). Listener-assessed intelligibility of a hearing aid self-adaptive noise filter. *Ear and Hearing, 5,* 199-204.

Stelmachowicz, P. (1985, May). *Psychoacoustic tuning curves in the prediction of speech intelligibility performance.* Paper presented at the Conference on Signal Processing for the Hearing Impaired, Madison, WI.

Villchur, E. (1982). The evaluation of amplitude compression processing for hearing aids. In G.A. Studebaker & F.H. Bess (Eds.), *The Vanderbilt hearing-aid report* (pp. 141-143). Upper Darby, PA: Monographs in Contemporary Audiology.

Walden, B.E. (1982). Validating measures for hearing aid performance. In G.A. Studebaker & F.H. Bess (Eds.), *The Vanderbilt hearing-aid report* (pp. 188-192). Upper Darby, PA: Monographs in Contemporary Audiology.

Walden, B.E., Demorest, M.E., & Hepler, E.L. (1984). Self-report approach to assessing benefit derived from amplification. *Journal of Speech and Hearing Research, 27,* 49-56.

Weiss, M. (1985, May). *Noise reduction techniques for implusive noise.* Paper presented at the Conference on Signal Processing Strategies for the Hearing Impaired, Madison, WI.