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SPEECH PROCESSING EFFECTS ON INTELLIGIBILITY FOR HEARING-IMPAIRED LISTENERS

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ABSTRACT

Alternative hearing aid amplification and signal processing schemes have been developed (a) to suppress noise, improving listeners' subjective assessments of the resulting sound quality and/or (b) to improve speech intelligibility for hearing-impaired listeners. This paper presents selected studies of the effects of various types of processing on both perceived quality and measured intelligibility. One study examined the effects of low- and high-cut filtering, superimposed on a suitable hearing aid gain function, using simulations running on a digital-signal processing (DSP) board. Another study compared two types of amplification systems, using purpose-built hearing aids in which the gain functions had been precisely matched to the needs of the individual hearing-impaired listener.

I. INTRODUCTION

Hearing aid signal processing is intended primarily (a) to improve the listener's ability to understand speech or (b) to improve the subjective quality of the processed sound. Killion, Staab, & Preves [11] have distinguished two general hearing aid signal processing strategies designed to assist hearing aid wearers when listening to speech in a background of noise. In both strategies, the gain function (i.e., the amplification provided at each frequency) changes in response to changes in the level of the input signal to the hearing aid. In one strategy, the relative amount of amplification provided at the low frequencies is reduced as the level of the input signal increases; this approach has been termed "BILL" for "base increases at low levels" [11]. In the other strategy, reduced amplification is provided at the low frequencies when the input level is low, but the gain function flattens as the input signal increases; this approach has been termed "TILL" for "treble increases at low levels" [11]. Figure 1 presents a schematic summary of the two types of processing.

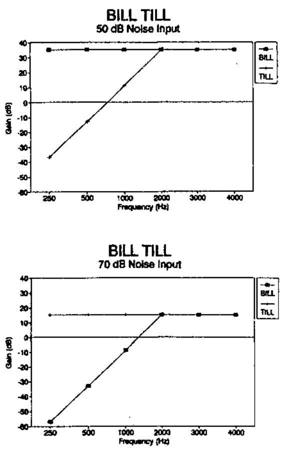

Figure 1. Example of the gain specified by two hearing aid signal processing schemes as a function of level of the input signal. At low levels (top panel), a constant 35 dB of gain is provided under the BILL scheme; under the TILL scheme however, gain is reduced below 2 kHz. At high input levels (bottom panel), gain is reduced overall and the constant gain is specified by the TILL (not BILL) scheme; under the BILL scheme, gain is further reduced below 2 kHz.

The "BILL" approach has been realized in several "noise-reduction" circuits, including the "Zeta Noise Blocker" [14,8]. The "TILL" approach has been realized in the "K-AMP" circuit[1] [10].

The evaluation of these and other ASP schemes when implemented in actual hearing aids is necessarily complicated by a variety of factors, including other electroacoustic characteristics of the hearing aid, (eg., physical/acoustical characteristics), and the appropriateness of the gain function which is implemented by the hearing aid under test -- and on which the ASP characteristic is superimposed -- for the hearing loss of each particular listener. An alternative approach is to study alternative ASP systems through a precise simulation of the ASP functions. Such an approach was followed here in Study 1, where the effects of low- and high-cut filtering were simulated on a digital-signal processing (DSP) board. Study 2 examined the possible benefits of K-AMP vs linear amplification, using purpose-built hearing aids in which the gain functions had been precisely matched to the needs of the individual hearing-impaired listener.

II. STUDY I

II.A. Approach

This study simulated "BILL" and "TILL" processing for normal and hearing-impaired listeners. The effects of the two types of processing on intelligibility and perceived quality were compared.

II.B. Subjects

Three groups participated. Groups 1 and 2 each contained four (different) normal hearing listeners (NHL), aged 27 to 33. All were unpaid volunteers from the Hearing Health Care Research Unit, whose hearing sensitivity was confirmed by audiometric screening (pure tone thresholds $\leq$ 15 dB HL at 500-4000 Hz).

Group 3 contained six hearing-impaired listeners (HIL), aged 31 to 62 years. Each HIL had a bilateral mild-to-moderate sensorineural hearing impairment, confirmed by conventional pure tone audiometry. HILs were paid for their participation. Data were collected on the right ear of NHLs and on both ears of HILs.

II.C. Materials and Equipment

Speech intelligibility was measured using an adaptive speech reception threshold testing procedure [2;3] which required listeners to identify individual spondaic words. Signals were presented in each of four noise conditions: broadband noise (lowpass filtered at 7.8 kHz) vs narrowband noise (lowpass filtered at 1 kHz) each presented at 50 dB and 70 dB.

Subjects wore a custom earmold attached to an ER-3A insert earphone which delivered the test stimuli. For testing, signals were replayed over a DT2801A digital to analog converter (DAC), sampled by the ADC of an ARIEL DSP-16 board, filtered using the board's TMS320C25 chip, output over the board's DAC, amplified (Crown D-75), and low-pass filtered (KEMO VBF25) at 7.8 kHz. To create the background noise, white noise produced by a Bruel and Kjaer Type 1027 Sine/Random Generator was low-pass filtered, using a KEMO Model VBF25 programmable filter, with the cutoff set to 1 kHz for the narrowband condition or to 7.8 kHz for the broadband condition. A

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[1] The K-AMP circuit has other characteristics as well, including compressive amplification.

digital filter was designed and implemented on the TMS320C25 chip of the ARIEL board; this filter was applied to both signal and noise, to ensure that the simulated adaptive signal processing functions were accurately represented, independently of the system response. For each test session, the amplification/ attenuation levels for each channel were set and adjusted under computer control to ensure that the speech and noise signals matched the target levels.

II.D. Procedure

Testing occurred in a double-wall sound attenuating test booth (IAC), using computer-controlled testing procedures. For each listener, the signal delivery system provided a specified amount of gain at each frequency. This individualized gain function reflected several factors: 1) the required real-ear insertion gain (REIG), specified by the NAL-revised formula [1] (for HILS; no REIG was provided for NHLs); 2) the filter function specified for a particular listening condition by each ASP scheme; 3) the measured deviation of the full physical/electroacoustic path from the input to the amplifier to the individual listener's tympanic membrane. A separate finite impulse response filter, taking all three factors into account, was designed and implemented on the TMS320C25 chip for each listener.

Speech Reception Thresholds (SRTs) were measured using an efficient, adaptive test procedure [3]. This test requires listeners to identify which of several alternative words has been presented; each word is presented in a background of competing noise. Within each test session, the type and level of the noise is fixed, while the level of the speech signal is varied according to a well-defined set of rules. By monitoring the sequence of correct and error responses throughout testing, the speech level can be adjusted in an efficient manner to obtain an estimate of the level that would lead to a given level of accuracy.

The procedure involves a "closed-set" test, in which the set of possible responses is displayed on a computer monitor, for the listener to read. The subject uses a computer mouse to select the desired response from the set of words displayed. Each time the subject makes an error (i.e., when the selected response does not correspond to the word presented), the signal level is raised by 1.5 dB. Following any two correct responses at one level, the signal level is reduced by 1.5 dB. This process continues until a total of eight reversals (i.e., a decrease in signal level following one or more consecutive increases in signal level, or the converse) has been achieved. The application of this rule causes the signal level to converge on the point where the proportion of correct responses is equal to 70.7% [12].

II.E. Results

II.E.1. Normal Hearing Subjects. The spondee threshold for each NHL in Group 1 is shown in Figure 2 for each condition. In each case, the amount of speech attenuation is measured in dB re: noise level; increased attenuation (of the speech signal) indicates better performance, since a given level of identification accuracy is achieved at a less favourable S/N ratio. In effect, this means that the listener achieved the same level of performance *under less favorable listening conditions*, as a consequence of the subsequent processing of the speech-plus-noise signal by the hearing aid.

Clearly, performance in the narrowband noise condition was better whenever a low-cut filter was applied to the (Signal+Noise) input, *regardless of input level*. Performance in the wideband noise condition was also marginally better whenever a flat filter was applied -- again, regardless of input level. Presumably, the low-cut filter *improved the signal to noise ratio* when listening in the narrowband noise condition; applying the same filter did not improve and may in fact have *reduced the signal to noise ratio* when listening in the wideband noise condition.

For 50 dBA input noise, intelligibility was higher for all listeners following TILL processing. However, at the 70 dBA input noise level, all listeners performed better in the BILL condition. Modest differences in the opposite direction were found when listening in broadband noise. At the 50 dBA input noise level, intelligibility was higher for all listeners following BILL processing. However, at the 70 dBA input noise level, all listeners performed better in the TILL condition.

This pattern of performance across conditions was also found with NHLs from Group 2 (cf. Figure 3): the ASP conditions differed substantially in the *narrowband* noise conditions, but showed minimal differences in the wideband noise conditions, regardless of the input noise level. Moreover, performance in the narrowband noise condition was better whenever a low-cut filter was applied to the (signal-plus-noise) input, *regardless of input level* (i.e., at the 50 dB input noise level, intelligibility was higher for all listeners following "TILL" processing, while at the 70 dB input noise level, all listeners performed better in the "BILL" condition). In the wideband noise condition, performance tended to be better when a flat filter was applied, particularly for subjects 2 and 3 -- again, regardless of input level. Presumably, in the narrowband noise condition, the low-cut filter *improved* the effective signal to noise ratio for normal hearing subjects. However, applying the same filter did not improve, and in some instances may in fact have reduced the signal to noise ratio, when listening in the wideband noise condition.

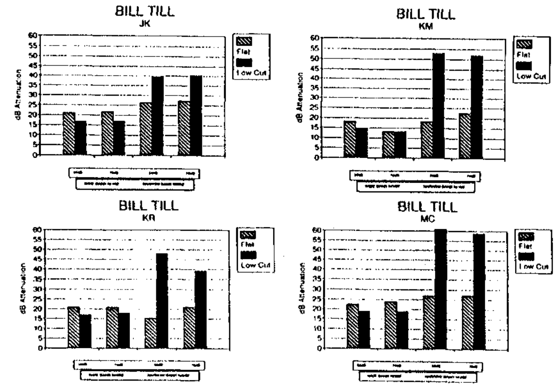

Figure 2. Each panel presents the final attenuation levels in the SRT task for an individual NHL in Group 1. Data are arranged so that adjacent bars represent performance under the flat-filter (striped bar) and low-cut filter (solid bar) signal processing conditions. From the left of each panel, the pairs of bars are for data obtained in in wideband noise at 50 dB and 70 dB and in narrowband noise at 50 and 70 dB, respectively. Each panel presents data for one of the four listeners.

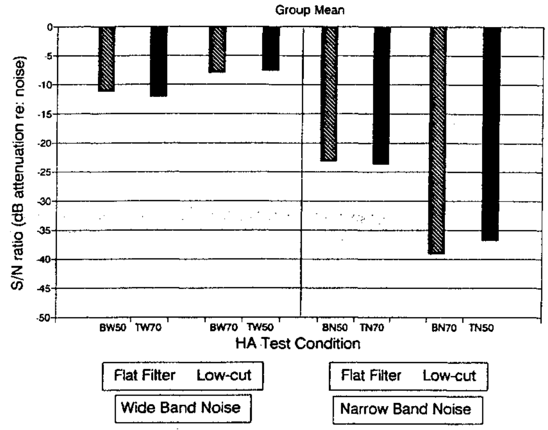

Figure 3. Group mean SRT thresholds for normal hearing listeners in Group 2. Data are arranged so that a direct comparison can be made for performance under the Flat-filter and Low-cut filter signal processing conditions.

II.E.2. Hearing-Impaired Subjects. The HIL Group mean spondee threshold in each condition is shown in Figure 4. The pattern of performance across conditions was clear: listeners tended to perform better in those ASP conditions which implemented a flat filter regardless of the noise conditions. With the exception of the left-ear results for subject EW when listening in narrowband noise, the benefits that low-cut filtering provided to normal hearing subjects in the narrowband noise condition were not obtained by HILs. The results obtained by HILs in the wideband noise test conditions are similar to but more extreme than those obtained by normal hearing subjects. That is, in the presence of wideband noise, applying a low-cut filter to a speech signal *reduces* performance for all listeners.

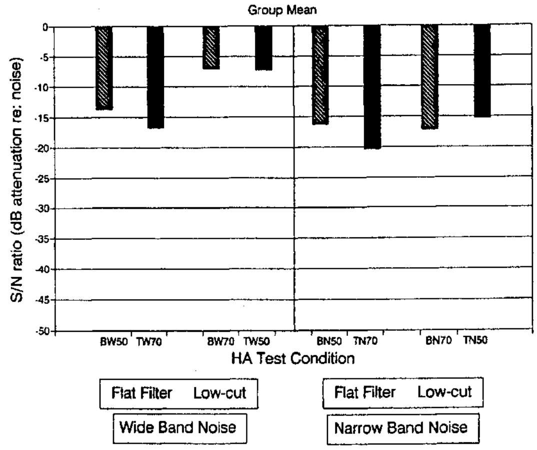

Figure 4. Group mean spondee thresholds for hearing-impaired listeners (Group 3). Data are arranged so that a direct comparison can be made for performance under the Flat-filter and Low-cut filter signal processing conditions.

II.F. Discussion

Processing a mixture of speech-plus-noise, using the two signal processing strategies considered here, leads to significant differences in intelligibility and in listener preference. Most clearly, speech intelligibility can be improved substantially for those with normal hearing who are listening in the presence of narrowband noise when the processing scheme applies a low-cut filter to the speech and noise mixture; in our analog, this is implemented as a "BILL" strategy at high noise levels and a "TILL" strategy at low noise levels. Applying such a low-cut filter removes the predominantly low-frequency noise (together with part of the speech input signal).

When listening in wideband noise, intelligibility and listener preference were marginally superior after flat ASP filtering, rather than low-cut filtering. This result apparently reflects a degradation of the signal when low-cut filtering is applied.

Our hearing-impaired listeners were less able to benefit from low-cut filtering in the narrowband noise condition, presumably because they were less able to use the enhanced high-frequency cues which remain in the processed speech-plus-noise signal. This result likely reflects both the relative inaudibility of these high-frequency cues for listeners who have a high-frequency hearing loss and -- because in our study, a real-ear-insertion-gain appropriate for each individual's loss was applied prior to signal processing -- the relative inexperience of these listeners in using these high-frequency cues. Indeed, there is evidence [6,7] to suggest that hearing-impaired listeners can learn to use such high-frequency cues, following several weeks of experience with such appropriate amplification.

III. STUDY 2

III.A. Approach

Five listeners were tested over 17 weeks with two pairs of hearing aids (Linear and K-AMP), fitted in sequence. Testing occurred prior to fitting the hearing aids (i.e., unaided), immediately after fitting the hearing aids, and during a period in which the hearing aids were worn in everyday life.

III.B. Method

Two types of speech in noise tests were used to measure speech intelligibility in sound field. One was the adaptive SRT measure described in Study 1. The test indicates that a given type of hearing aid processing provides an advantage if it permits the listener to achieve the specified level of performance when the environmental listening conditions *are less favorable* (i.e., when the S/N is lower), than those required by a competing system.

The second was a modification of the Distinctive Features Differences Test [2; 3; denoted DFD(m)]. This measure indicates the listener's ability to identify the different consonant speech sounds (in VCV context), at a given level and S/N. The test indicates that a given type of hearing aid processing provides an advantage if it permits the listener to achieve a higher level of performance under identical environmental listening conditions than can be achieved using another hearing aid system.

Two types of self-report questionaires were used to quantify communication performance in everyday life. The PHAP [4] assesses experience with amplification in terms of speech communication in three types of listening situations and in terms of reactions to environmental sounds. The CPHI [5] is a self-assessment inventory that provides scales of communication performance (CP), communication importance (CI), communication environment (CE), communication strategies (CS), and personal adjustment (PA).

III.B.1. Subjects. Five subjects with mild-to-moderately-severe sensorineural hearing loss, aged between 30 and 65 years, participated in the study. Of these, three subjects (EC, EW, VB) completed all phases of the experiment, while two subjects (GW, HW) completed some phases of the experiment, but declined to participate at later points in the testing.

Pure tone thresholds and Loudness Discomfort Levels were obtained using an insert ER-3A earphone. Subjects were paid an hourly rate and given both sets of hearing aids at the end of the experiment.

III.B.2. Hearing Aid Prescription. At the outset, detailed audiometric assessments were conducted to support the selection and fitting of each listener's hearing aids. The result of this selection process was that the hearing aid manufacturer was provided with target full-on gain and frequency specific SSPL-90 curves for each ear, specified in a 2-cc coupler so compliance to the specification could be confirmed by both manufacturer and our laboratory. Identical prescriptions were applied for both sets of hearing aids (i.e., the K-AMP and Linear), with the aids equated at both real-ear insertion gain (REIG; 60 dB input) and real-ear aided response (REAR; 90 dB input).

III.B.3. Design. Both behavioral and subjective measures of hearing aid performance were obtained. Behavioral Speech Reception Thresholds (SRT) were measured in noise, under both aided and unaided conditions, at each test session, using the approach described for Study 1, above. Speech identification was measured using the DFD(m) [3] in noise; unaided DFD(m) performance was assessed at the beginning and end of the experiment, and aided performance was assessed during each testing session. Subjective measures included responses to the PHAP [4], and the CPHI [5].

The experiment required subjects to return for six test sessions (one pre-aided and five aided) over a 17 week time period. Three subjects were first fitted with the K-AMP hearing aids and two subjects were first fitted with the Linear hearing aids. Two subjects (EW and HW) had no prior experience with an amplification device while the other three subjects were experienced hearing aid users. Both the behavioral and subjective measures were obtained at various points in time during the experiment, as outlined further below.

III.B.4. Timeline. Session 0 -- all subjects completed the PHAP and CPHI questionnaires prior to hearing aid fitting; Session 1 -- one week later, subjects were fitted with their first pair of hearing aids and both aided and unaided performance was measured using the SRT and DFD(m); Session 2 -- four weeks later, aided performance on the SRT and DFD(m) tasks was measured, the SRT was assessed unaided, and the PHAP questionnaire was completed; Session 3 -- eight weeks later, aided performance on the SRT and DFD(m) tasks was measured, the SRT was assessed unaided, and the CPHI, and PHAP questionnaires were completed. Subjects were then fitted with the second pair of hearing aids, and aided performance on the SRT and DFD(m) was measured.

Session 4 -- four weeks later, aided performance on the SRT and DFD(m) tasks was measured, the SRT was assessed unaided, and the PHAP questionnaire was completed; Session 5- eight weeks later, performance on the SRT and DFD(m) tasks was measured aided and unaided and subjects completed the CPHI and PHAP questionnaires. The subjects were then asked to indicate which of the two sets of hearing aids they would choose to wear (if they could only keep one set).

III.C. Results and Discussion

Table 1 summarizes the profile of results obtained for each listener with three types of comparisons: (1) K-AMP amplification vs unaided/pre-aided performance; (2) Linear amplification vs unaided/pre-aided performance; and (3) K-AMP amplification vs Linear amplification.

Table 1 Comparisons of ultimate (8-week) performance for each pair of hearing aid conditions

I. Comparison of K-AMP hearing aids with unaided/pre-aided performance

	SRT-50	SRT-70	DFD(m)-50	DFD(m)-70	PHAP	CPHI
EC	**K**	u	**K**	**K**	-	-
EW	-	-	k	u	**K**	k
VB	-	-	**K**	u	**K**	k
GW	**K**	-	-	-	**K**	-
HW	**K**	**K**	-	-	**K**	k

II. Comparison of Linear hearing aids with unaided/pre-aided performance

	SRT-50	SRT-70	DFD(m)-50	DFD(m)-70	PHAP	CPHI
EC	-	-	**L**	**L**	l	-
EW	-	u	-	**U**	**L**	l
VB	-	-	**L**	**L**	**L**	**L**
GW	l	l	-	-	-	-
HW	l	-	-	-	p	-

K-AMP vs Linear hearing aids

III. Comparison of K-AMP hearing aids with Linear hearing aids

	SRT-50	SRT-70	DFD-50	DFD-70	PHAP	CPHI	CHOSE	FIRST
EC	**K**	**L**	**K**	-	-	-	**K**	K
EW	-	**K**	k	k	-	-	**L**	L
VB	-	k	l	**L**	-	**L**	**K**	L
GW	**K**	-	-	-	**K**	-	kl	K
HW	k	**K**	l	-	**K**	-	**K**	K

Notes:

Uppercase, bold, type indicates a difference which is statistically significant beyond the 5% level of chance in favor of the K-AMP hearing aid (**K**), the Linear hearing aid (**L**), the unaided condition (**U**; i.e, for inexperienced/novice hearing aid users), or previously aided (**P**; i.e., for experienced hearing aid wearers, where judgements were made in relation to the personal hearing aid they were using at the start of the study). Lowercase, unbolded type indicates a difference which is not statistically significant, but which is at least one standard deviation in size, for the K-AMP hearing aid (k), the Linear hearing aid (l), unaided (u), or preaided (p). A dash (-) indicates that no statistically significant comparison could be made or that no tendency could be discerned. Hearing aid data were collected at the conclusion of a listener's eight-week trial with the indicated hearing aid. Unaided data were collected at the corresponding time. Pre-aided data were collected prior to the initial hearing aid fitting. Columns indicate measured speech intelligibility with the Speech-Reception Threshold test in a background of 50 dB or 70 dB background noise (SRT-50 and SRT-70, respectively), measured speech intelligibility with the Distinctive Features Differences Test in a background of 50 dB or 70 dB background noise (DFD-50 and DFD-70, respectively), self-ratings of performance using the Profile of Hearing Aid Performance (PHAP), self-ratings of performance and attitudes using the Communication Profile for the Hearing Impaired (CPHI), the hearing aid selected as preferred at the conclusion of the experiment (CHOSE), and the hearing aid which was the first one provided to the listener.

IV. GENERAL DISCUSSION

When a listener is required to listen to a speech signal in noise, applying an appropriate adaptive signal processing strategy can lead to *both* increased intelligibility and greater listener preference. The key remains selecting the *appropriate* strategy -- and this involves an interaction between the listening condition (type and level of signal and of noise), type of processing applied, the listener's residual auditory capability, and the listener's experience with the processing. In the present work, the appropriate strategy was clearly the application of low-cut filtering when listening in a background of narrowband, low frequency noise -- for those having the residual auditory capacity to benefit from such processing. Similar "spectral-subtraction" procedures, in which an estimate of the noise spectrum is used to design a filter which is then applied to the signal-plus-noise complex, bear additional investigation in the design of hearing aids.

The benefits of similar processing as part of a complete, wearable, hearing aid instrument are less clear. This is particularly true when evaluated with respect to a *carefully fitted*, linear hearing aid. In Study 2, some of our listeners strongly preferred K-AMP hearing aids; others preferred the Linear aids, or different aids in different conditions. This result is consistent with informal reports from clinicians and listeners. However, because of the careful controls used in the present experiments, there is greater confidence that these effects are in fact attributable to some aspect of K-AMP processing. In particular, the result cannot be attributed to differences in the overall gain function between the different hearing aids (at the target input level).

The factors which *cause* the differences and/or preferences observed here cannot presently be determined. Additional work should be directed toward *pre-dicting* which listeners are likely to benefit most from, and/or most prefer, a hearing aid with K-AMP, rather than linear, circuitry. This work would require (a) the administration of an extensive battery of psychoacoustical tests, to characterize precisely, each listener's residual auditory capacity, prior to the fitting of a hearing aid; and (b) additional measures of speech understanding ability.

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