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Perceptual Evaluation of Digital Signal Processing Strategies In a Modem Hearing Instrument Across Noisy and Reverberant Environments

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Abstract

Speech intelligibility and quality scores were evaluated across four different hearing aid settings that differed in the strength of directional microphone (DM), digital noise reduction (DNR), and level dependent speech enhancement (LDSE) features, in quiet and noise, and in low and high reverberation environments. Twenty-two listeners with bilateral sensorineural hearing loss and ten normal hearing listeners participated in our study. Results indicated that the directional microphone condition provided significant improvement for speech recognition in noise, at both levels of reverberation. Addition of SE and DNR processing to directional microphone had both beneficial and detrimental effects on speech perception and sound quality depending upon the strength of processing, type of environment, and noise condition. Specifically, SE and DNR features operating at maximum strength degraded speech intelligibility in the high reverberation environment. The same processing condition was, however, rated as having higher sound quality especially when the masker was stationary noise at 0 dB signal to noise ratio in low reverberation. Clinical implications of these results are discussed.

Keywords: Speech enhancement, Directional microphone, Digital noise reduction, Reverberation, Speech intelligibility, Speech quality

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Introduction

Difficulty understanding speech in noise has been and continues to be a common complaint of hearing impaired listeners (Kochkin, 2002; Kochkin, 2010). Background noise reduces speech understanding by masking the highly redundant acoustic and linguistic cues important for speech recognition (Smaldino, Crandell, Kreisman, John, & Kreisman, 2008). Smaldino et al. (2008) also indicated that the overall level of background noise was not the primary factor for speech understanding, but rather it was the signal-to-noise ratio (SNR) of the listening environment. In general, speech recognition reduces as SNR of the listening environment decreases (Nabelek & Nabelek, 1994). Listeners with sensorineural hearing loss (SNHL) generally require more favourable SNRs to achieve the same speech perception scores as listeners with normal hearing (Killion, 1997; Ricketts, 2001). Reduced audibility due to SNHL is not the only factor affecting speech recognition, as other temporal and spectral aspects of the auditory mechanism play critical roles in speech perception (Killion, 1997).

Digital signal processing (DSP) algorithms have been applied in hearing aids to improve SNRs for hearing-impaired listeners. Various attempts have been made at designing new DSP algorithms to improve auditory perception for hearing impaired listeners. DSP schemes have been implemented in both microphone-based and processing-based applications (Bentler, Wu, Kettel, & Hurtig, 2008). Directional microphone (DM) technology employs two microphones (front and rear), the combination of which substantially improves the SNR in situations where the signals of interest and competition are spatially separated (Ricketts & Mueller, 1999; Ricketts, 2001). By appropriately delaying the signal transduced by the rear microphone and subtracting it from the signal acquired from the front microphone, DM reduces sounds coming from behind or beside of the listener relative to those arriving from the front, resulting in an improved SNR (Ricketts & Mueller, 1999). In addition, a number of modern hearing aids implement multiband adaptive directionality, where the internal delay is adapted in a frequency-specific manner to optimally reduce the background noise originating in the rear hemisphere of the listener. Benefits of directional hearing aids compared to omnidirectional (OM) hearing aids have been well established (e.g., Ricketts & Hornsby, 2003; Ricketts & Mueller, 1999).

In situations where the desired signal and background noise are not spatially separated, additional noise reduction strategies are required. Processing-based noise reduction algorithms are examples of those strategies, which reduce output of hearing aids in background noise (Bentler & Chiou, 2006). These manufacturer-specific algorithms analyze the incoming signal, and modify the gain/output characteristics based on pre-calculated rules. In general, these algorithms analyze a multitude of temporal and spectral features (e.g., modulation depth and frequency) to detect the presence of speech and to estimate the SNR in different frequency channels (Bentler & Chiou, 2006). Frequency regions with unfavourable SNRs (< 0 dB) are attenuated by a prescribed amount. Benefit of DNR algorithms on speech recognition in noise has not been clearly proven (Bentler et al., 2008). Nordrum, Erler, Garstecki, and Dhar (2006) compared the performance of DM and DNR algorithms in four different hearing aids, and concluded that there was no significant improvement in speech understanding with DNR turned on. Bentler, Palmer, and Mueller (2006) also suggested that the effects of DM and DNR on speech recognition were independent each other. Walden, Surr, Cord, Edward, and Olson

(2000) evaluated the effects of DNR with DM on speech recognition, sound comfort, and sound quality. They found better sound comfort with DNR and DM than with DM alone. Ricketts and Hornsby (2005) also found sound quality preferences for DNR but no improvement in speech recognition. Similarly, Bentler et al. (2008) stated no benefit of DNR on speech recognition. Laboratory-based ratings of ease of listening, however, indicated better ratings for the DNR-on conditions even though sound quality ratings were not affected. The Bentler et al. (2008) self-report evaluation also revealed significantly higher aversiveness in the DNR-off condition compared to the pre-test measures (unaided condition). More recently, Sarampalis, Kalluri, Edwards, and Hafter (2009) demonstrated the lack of benefit of DNR on speech perception at low SNRs, but a decrease in listening effort to free up cognitive resources for other tasks.

In summary, the main effects of DNR relate to the ease of listening and acceptability (or lessened aversiveness) of high-level and/or noisy signals. DNR has not, however, been shown to produce increased speech recognition. A plausible reason for the lack of DNR benefit on objective measures of speech recognition is the typically overlapping nature of speech and noise frequency bands (Bentler, Palmer, & Mueller, 2006). As gain is reduced in the frequency regions where noise is detected, gain for speech in the same frequency range is reduced, resulting in no SNR improvement but rather an overall level reduction. In contrast, subjective listening comfort appears to increase due to this level reduction by DNR for noisy stimuli.

More recently, new processing-based algorithms called speech enhancement (SE) have been designed for hearing aids to further boost frequencies where speech energy is greater than noise energy. Peeters, Kuk, Lau, and Keenan (2009) studied objective and subjective efficacy of a new SE algorithm in a commercial hearing aid. This algorithm calculates the Speech Intelligibility Index (SII) based on estimates of the noise spectrum, speech spectrum, and hearing thresholds. The results of their study revealed that the SII-based speech enhancer in conjunction with directional microphones significantly improved the SNR compared to the directional condition alone on subjective measures, but no improvement in objective measures was observed. Hayes (2006) also evaluated a new SE algorithm called level dependent speech enhancement (LDSE). This algorithm distinguishes speech in both quiet and noisy environments based on modulation properties, and adds more gain to the frequency bands of modulated signals compared to those with little modulation. This is designed to provide more gain to speech and not to noise (Hayes, personal communication). Hayes (2006) also indicated that the level dependency of speech enhancement provided more gain to softer speech than to louder speech resulting in reduced loudness and distortion issues related to traditional speech enhancement algorithms.

The effectiveness of the aforementioned DSP algorithms in real-world listening environments is still under investigation. In real-world listening environments, listeners are typically exposed to a combination of direct and reverberant energy from both speech and competing noise sources. Synergistic effects of noise and reverberation could significantly reduce speech perception due to different masking effects of noise and reverberation on speech (Nabelek & Nabelek, 1994). The masking effect of reverberation causes impulsive noise energy to become more steady-state, resulting in a poorer SNR. Nabelek and Nabelek (1994) also stated that the interaction of noise and reverberation distorted speech features of voicing, manner, and place of articulation for consonants leading to diminished phonemic cues. Typically, speech recognition does not change considerably in normal hearing adult listeners until the reverberation time (RT) exceeds about 1 s (reverberation time is defined as the time required for the sound pressure level (SPL) to decay 60 dB after the termination of a signal). Listeners with SNHL, however, require significantly shorter RT (e.g., 0.4 to 0.5 s) in order to obtain maximum speech recognition (Smaldino et al., 2008). As with noise and reverberation in isolation, Smaldino et al. (2008) also suggested that listeners with hearing impairment had poorer speech understanding in noise and reverberation than those with normal hearing.

The effectiveness of directional microphones in reverberant environments is varied. Some studies have indicated little to no directional benefit in some noisy and reverberant environments (Ricketts, Henry, & Gnewikow, 2003; Ricketts & Hornsby, 2003). Other studies demonstrated that the effectiveness of DM in the real-world listening environments was not clear (Cord, Surr, Walden, & Dyrlund, 2004; Gnewikow, Ricketts, Bratt, & Mutchler, 2009). Benefit of DM was reported to diminish as listening environment became more reverberant (Ricketts & Dhar, 1999), and the speaker to listener distance increased (Ricketts & Hornsby, 2003). Directional benefit also reduces when multiple noise sources, as opposed to a single noise speaker, are used (Ricketts, 2000). Ricketts (2000) also suggested that speaker configuration of 0°/180° (signal emanating from the front and a single noise behind the listener) significantly impacted the directional benefit and the rank order of benefit across hearing aid brands. The evidence therefore suggests that laboratory experiments may overestimate directional benefit compared to more realistic acoustic environments (Gnewikow et al., 2009).

Very few studies have investigated the performance of processing-based noise reduction algorithms in reverberant environments. Recently, Luts et al. (2010) evaluated objective and subjective benefits of five different signal enhancement algorithms in both low and high reverberation environments. The results revealed more preferences for most of the algorithms over the unprocessed condition (omnidirectional) at all tested SNRs even though speech intelligibility scores did not improve.

In summary, there is a paucity of studies investigating the benefit of processing-based algorithms on speech recognition and sound quality in different reverberant environments. In addition, no studies have systematically evaluated the combined effects of different DSP strategies across anechoic and reverberant environments. Few studies have also reported the performance of hearing-impaired listeners compared to that of normal listeners in varied signal processing algorithms.

The purpose of our study was therefore to evaluate the synergistic effect of three DSP algorithms (DM, DNR, and LDSE) as implemented in a commercial hearing aid (Unitron Hearing, "Passport") in a variety of noisy and reverberant environments. The Passport hearing aid is accessorized by a remote control (SmartFocus™) which provides the hearing aid user with real-time control over the combination of three adjustable parameters including microphone strategy (omnidirectional versus degrees of directionality), strength of speech enhancement, and strength of noise reduction. The simultaneous adjustment of multiple parameters is designed to eliminate unpredictable interactions between features that occur when all of them act independently (Hayes, 2009). In clinical use, the SmartFocus control is implemented as a trainable control, logging the user's preferences across a variety of environments, thereby allowing the aid to "learn" which setting is preferred. After an initial training period, the user can cease training and allow the hearing aid to automatically choose a setting along the SmartFocus continuum in response to the hearing aid's classification of the current environment.

Our study aimed to evaluate the relative effectiveness of varying the SmartFocus control on speech intelligibility and sound quality. Our primary question was whether the combined use of multiple processors provided additional benefits with speech intelligibility and sound quality over and above those obtained with directional microphone alone. Our second question was whether the synergistic effect of those three adaptive features interacted with room reverberation. Finally, our third question was whether hearing-impaired listeners performed similarly to normal hearing listeners when three DSP algorithms were combined. We hypothesized that SmartFocus with three fully active parameters would provide the maximum speech intelligibility and quality across all listening environments. We also hypothesized that the combined effect of three features complemented the function of DM alone in reverberant environments. Our last hypothesis was that the full strength of three adaptive features would result in speech understanding performance similar to that of normal hearing listeners.

Methods

Participants

Twenty-five participants with hearing impairment were recruited from the Translational Research Unit of the National Centre for Audiology for this study (8 women and 17 men). The sample size for this study was estimated using the Horatio software package (Lee, 2004). Parameters for the sample size estimation included an alpha level of .05, a medium effect size, a within-subjects design with four levels of repeated measurement, and 90% power to detect a significant change. For repeated measures, this software also assumed a 30% correlation between repeated measures by default. Previous research on improvement in speech recognition and listening preferences with different hearing aid signal processing across environmental conditions has revealed a medium to large effect sizes (Ricketts & Hornsby, 2003; Amlani, Rakerd, & Punch, 2006). With these parameters, the Horatio software reported that a sample size of 20 would provide power of 90% at a critical F ratio of 2.76. Therefore, 20 participants should meet the power requirements. Additional participants were recruited to compensate for some level of participant withdrawal in case this should occur.

From the initial 25 participants, three participants dropped out of the study leaving 22 participants (6 women and 16 men). Of the 22 subjects,15 participants were experienced hearing aid users (wearing hearing aids more than a year), 2 subjects were new hearing aid users (wearing hearing aids less than a year), and 5 subjects were non users of hearing aids. The age range of the 22 participants was 38-85 years with a mean age of 71. Participants in this study met the following criteria: 1) mild to moderately severe sensorineural hearing loss, predominantly downward sloping, with no significant air-bone gap (<10 dB per frequency), 2) normal tympanogram defined as compensated static admittance between 0.35 and 1.65 mmho measured from the positive tail with tympanometric peak pressure between -100 and +100 daPa, and 3) bilateral hearing loss with 4 frequency pure tone average (0.5,1,2,4 kHz) asymmetry of less than or equal to10 dB .

In addition, 10 normal hearing listeners (thresholds better than 15 dB HL) and ages 23-28 were recruited and tested in the unaided condition. Test results from this group provided the reference data for use in interpreting the scores from listeners with hearing loss.

The testing protocol was approved by the Office of Research Ethics at The University of Western Ontario (UWO) (Appendix A). All participants signed an informed consent form after an explanation of the purpose of the study as well as the benefits and risks prior to their participation. Participants were financially reimbursed for their time and/or parking expenses, and were provided with sufficient rest periods to prevent fatigue during testing.

Hearing aids

Two Unitron Passport behind-the ear (13 BTE series) hearing aids were evaluated in our study (participants had never experienced Passport hearing aids before). The passport is a 20-channel hearing aid with 125 dB SPL peak output and 60 dB peak gain, four automatic programs, and three manual programs. Multiple microphone options in Passport include omnidirectional, fixed directional, and multiband adaptive directional processing. Passport also features digital noise reduction and level dependent speech enhancement. As indicated, SmartFocus is a unique setting in Passport hearing aids, which provides a single control to adjust a combination of the three adaptive features (DM, DNR, and LDSE) in both automatic and manual programs. In our study, we evaluated the effects of adaptive feature combinations from the neutral setting of SmartFocus to its maximum setting. While the control allows for a gradual change of the feature strength across this range, for the purpose of this study we tested listeners at four discrete points in this range: a) SmartFocus Omni, b) SmartFocus Directional, c) SmartFocus Partial Strength, d) SmartFocus Full Strength. Table 1 defines the specific characteristics of each SmartFocus setting.

It must be noted here that the dB values specified in the last two columns of Table 1 represent the maximum gain and attenuation values respectively. The exact amount of gain or attenuation at any given time depends on a combination of input signal level, frequency-specific SNR, and the hearing loss configuration (Cornelisse, personal communication).

For each of these four settings, perceptual measures of speech recognition and sound quality were measured under two conditions of reverberation as described below. The order of speech lists, hearing aid settings, and rooms were counterbalanced among participants. A single-blind design was used in which participants were not told what hearing aid conditions were being tested.

Rooms

Two test environments were used: a double- walled sound booth and a reverberant chamber, both located at the UWO National Center for Audiology. The internal dimensions of the double- walled room were 2.8 m X 3.0 m X 1.9 m. The room's measured reverberation time (RT_{60}) was 0.1 s. Internal dimensions of the reverberant chamber were 6.1m X 4.0 m X 2.6 m. RT_{60} in this room was tuned to 0.9 s by placing an acoustic curtain and an acoustic foam panel on the walls. RT_{60} was measured in both rooms using the SpectraPlus software with a pink noise stimulus. SpectraPlus reports both frequency-specific and wideband RT_{60} values; the wideband RT_{60} values were used in this study to characterize room reverberation. RT measured in the reverberant room was higher than real RT in the living rooms and offices (0.4-0.8 s), but lower than that in large classrooms, small auditoriums, and places of worship (Nabelek & Nabelek, 1994).

SmartFocus setting	Microphone directionality	LDSE (max. gain in dB)	Noise reduction (max. attenuation in dB)
Omni	Omnidirectional	Off	Off
Directional	Fully adaptive directional	Off	Off
Partial strength	Partial adaptive	5.6	6
Full strength	Fully adaptive	7	7.5

Table 1. Summary of adaptive features across SmartFocus settings

Therefore, the $RT_{60} = 0.9$ s selected in our experiment was representative of a more difficult real-world listening environment and consistent with previous studies of signal processing performance in reverberation (Luts, et al., 2010; Ricketts & Hornsby, 2003).

Speech stimuli

Speech recognition was measured in noise using the Hearing in Noise Test (HINT: Nilsson, Soli, & Sullivan, 1994). Speech quality was assessed using the Multiple Stimuli with Hidden Reference and Anchors (MUSHRA) protocol (Stoll & Kozamernik, 2000). Each of these measures was administered for each hearing aid condition within subjects.

The HINT is an adaptive psychometric procedure that requires the listener to repeat recorded sentences of a male talker in the presence of speech-shaped noise. The level of the sentences is adapted, with the noise level fixed at an overall level of 65 dBA. This test measures the reception thresholds for sentences (RTSs) as the SNR required to obtain 50% correct recognition of sentences. In HINT testing, all the key words of a sentence must be accurately repeated in order for the sentence to be considered correct. The HINT stimuli consist of 250 sentences, which can be presented as 25 equivalent ten-sentence lists or as 12 equivalent 20-sentence lists. Two presentations of the 20 sentence lists were presented in our study. The SNRs over the last 17 sentences were averaged together to obtain the dB SNR for RTS. For this study, the HINT test was implemented using an in-house custom software that automatically scored the results. The software also modified the HINT noise to be continuous (rather than paired with the sentences) and to have 10 seconds of noise-only presentation prior to sentences. This allowed the DNR and adaptive directionality to be fully activated prior to testing.

The MUSHRA protocol has been developed for the systematic rating of general sound quality for speech and audio coding technologies (Stoll & Kozamernik, 2000). The procedure allows the listener to rate a set of stimuli by their perceived sound quality on a 0-100% scale (poor to excellent ratings). A modified version of the MUSHRA protocol was employed in this project with no hidden reference and anchors. Although reference and anchor stimuli are typically used within MUSHRA, this protocol required comparison across aided conditions only, therefore reference and anchor stimuli were not used. Custom-developed computer software displayed four sliders that were used to make the sound quality ratings. These four sliders corresponded to the four hearing aid settings in our study. The software was connected to the hearing aids, and it automatically selected and randomized the hearing aid settings. Listener instructions for the task are provided in Appendix A. Listeners completed the MUSHRA ratings at a presentation level of 65 dBA (speech) and at three SNRs (-5, 0, and +5 dB). SNRs of 0 dB and +5 dB were selected in our study to simulate real-world SNRs (Ricketts & Hornsby, 2005; Gnewikow et al., 2009). The -5 dB SNR was also included to investigate whether the combination of DM, DNR, and LDSE improved the quality of speech even in harsher noisy environments. Both multi-talker babble (BKB-SIN noise: Etymotic Research, 2005) and speech-shaped noise (HINT noise) were used at each SNR to determine the synergistic effects of different signal processing strategies on speech quality under differing noisy environments. The speech stimuli were the concatenated sentences of list one from the standard HINT. Table 2 provides a summary of the behavioural tests conducted for this study.

For all measures, speech and noise were presented from directly in front of the listener at zero degrees (0°) azimuth via two separate speakers; the speech speaker was positioned directly above the noise speaker. Additional three noise speakers were also placed around the listener at 90^{o}, 180^{o}, and 270^{o} azimuth. This speaker arrangement was specifically chosen to simulate a difficult listening situation where noise and speech were presented from the front as in many real world environments.

The speech and noise stimuli were presented from Di5 DC Tannoy loudspeakers in the reverberation chamber, and from Anthony Gallo Acoustics Nucleus loudspeakers in the sound booth. The speech loudspeaker and the four uncorrelated noise loudspeakers were placed at approximately ear level using speaker stands (132cm from the floor) at a distance of 1.2 m from the listener. Critical Distance (CD) is the distance at which the direct and reflected sound energies are equal. Similarly to previous studies (Hawkins & Yacullo, 1984; Leeuw & Dreschler, 1991; Ricketts & Dhar, 1999; Amlani et al., 2006), CD was estimated using Peutz's (1971) formula: $CD = 0.2\sqrt{(V/RT)}$, where CD = critical distance (m), V = volume of the room (m^3), and RT = reverberation time (s). Using this formula, the CDs were approximately 2.53 m and 1.68 m for low and high reverberant environments respectively. Thus, the speaker-to-listener distances were within the CD in both rooms. Previous studies have shown that the DM is most effective when the listener is within the CD (Ricketts & Hornsby, 2003), thus this arrangement allowed us to evaluate the effectiveness of combined DM, DNR, and LDSE over DM.

	HINT		MUSHRA														
	Low reverberation	High reverberation	Low reverberation							High reverberation						
			Quiet	-5 dB		0 dB		5 dB		Quiet	-5 dB		0 dB		5 dB	
				N	B	N	B	N	B		N	B	N	B	N	B
Unaided	✓	✓														
Omni	✓	✓	✓	✓	✓	✓	✓	✓	✓	✓	✓	✓	✓	✓	✓	✓
Directional	✓	✓	✓	✓	✓	✓	✓	✓	✓	✓	✓	✓	✓	✓	✓	✓
Partial Strength	✓	✓	✓	✓	✓	✓	✓	✓	✓	✓	✓	✓	✓	✓	✓	✓
Full Strength	✓	✓	✓	✓	✓	✓	✓	✓	✓	✓	✓	✓	✓	✓	✓	✓

Table 2. Outcome measures across listening conditions and rooms [N = Noise (HINT), B = Babble (BKB-SIN)].

The same calibration equipment, operated by the same researcher, was used at both test sites to verify the presentation levels. Prior to testing, daily calibration was also performed. A sound level meter (Larson Davis system 824) was placed on a tripod in the center of the room with the microphone at the position of the listener's head. The output of each loudspeaker was independently calibrated with a sample of the speech noise filtered to have the same long-term average spectrum as the HINT sentences.

Procedure

Full audiological assessment consisting of otoscopy, tympanometry, and pure-tone audiometry was conducted. Pure tone thresholds were obtained at 250, 500, 1000, 2000, 3000, 4000, 6000, and 8000 Hz with ER-3A insert earphones using two audiometers (Interacoustics AC40, Grason-Stadler, GSI 16*)*. If complete audiometric evaluation had recently been completed at the UWO H.A. Leeper Speech and Hearing Clinic (< 6 months), clinical records were used for the purposes of defining audiometric threshold and candidacy. In addition, frequency- specific loudness discomfort levels were measured using Hawkins' procedure (Hawkins, 1984). The mean hearing thresholds of the participants are shown in Figure 1. Participants were then fitted bilaterally with Unitron Passport behind-the-ear hearing aids. The hearing aids were coupled to the ear using a custom skeleton earmold with standard #13 tubing and 1.5 mm vent. The aids were fitted using the DSL 5 adult prescriptive algorithm (Scollie et al., 2005). Fittings were verified in the ear using a Verifit hearing aid analyzer to match the target. Hearing aids were further adjusted based on participant's comfort levels if necessary. Hearing aid responses were within an average of 3.5 dB of DSL 5 targets from 500 Hz to 6 kHz.

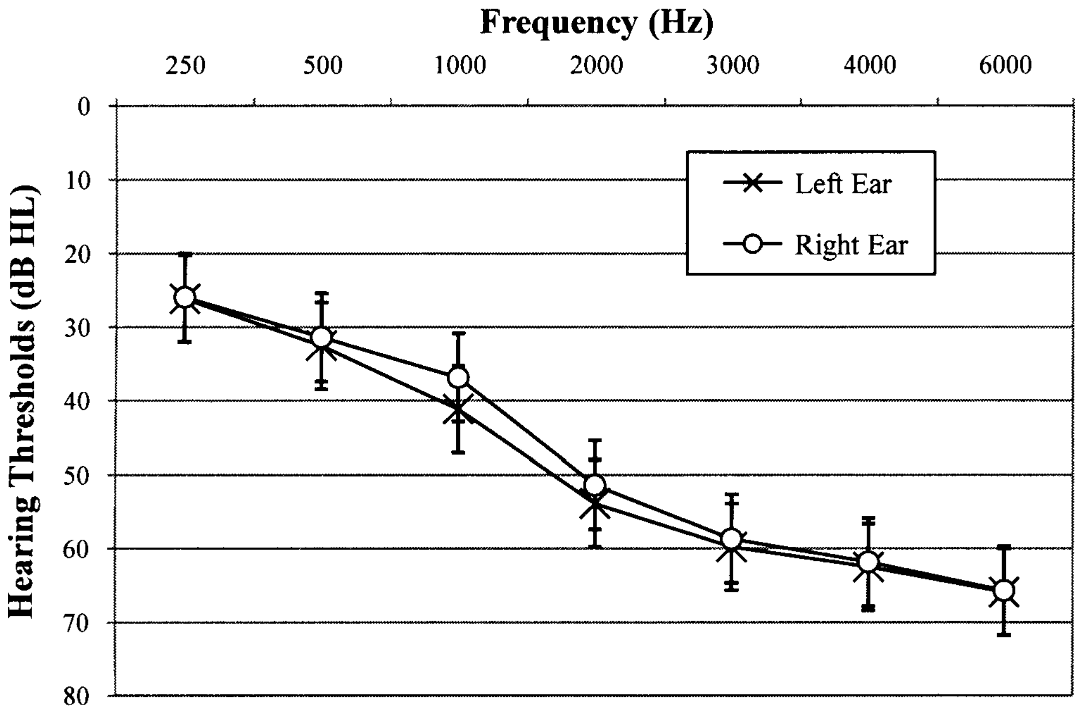

Figure 1. Average pure-tone thresholds (with one standard deviation bars) for the right and left ears.

Results

This study evaluated the synergistic effect between LDSE, DM, and DNR on speech perception and sound quality in two listening environments (low and high reverberation).The combined effect of three adaptive features was assessed on a user control (SmartFocus) in omni, directional, partial strength, and full strength. The dependent variables of speech perception scores and sound quality ratings were analyzed as a function of the independent variables of hearing aid settings and listening environments using repeated- measures analysis of variance (ANOVA). Post-hoc analyses of significant findings were completed using a modified Bonferonni procedure, known as the False Detection Rate (FDR; Benjamini & Hochberg, 1995), in which the tests were sorted in descending order of significance. The Greenhouse Geiser epsilon correction was used to adjust the analysis of variance for lack of sphericity (Max & Onghena, 1999). The maximum critical alpha per contrast was the overall alpha of 0.05. This controls the overall error rate per family to 11.4%.

Speech recognition in noise

During the HINT test, two participants out of 22 were considered outliers as they had unusually high (poor) RTSs. Participants were excluded from group analyses if the listener's mean aided score on the HINT test was poorer than the mean aided score from the group + 2 standard deviations averaged across the two rooms.

The HINT data were analyzed for the effects of room and condition as well as the interactions between these. The main effects of room, $F(1,19) = 134.24, p < 0.001$, and condition, $F(1, 32) = 21.52, p < 0.001$ were significant. The interaction of room and listening condition, $F(2, 56) = 4.81, p = 0.005$ was also significant. Paired comparisons of the conditions (t-test) were completed independently for each room. An a priori decision was made to investigate the listening conditions against one another per task and listening environment, as detailed below. Paired comparisons on five contrasts of interest were completed on1) unaided versus omni, 2) omnidirectional versus directional, 3) directional versus partial strength, 4) partial strength versus full strength, 5) directional versus full strength.

In addition, independent samples t-tests were completed to compare the normal-hearing listeners' results to unaided as well as all aided hearing-impaired participants' results for both listening environments.

In low reverberation

HINT results for hearing-impaired listeners in the soundbooth demonstrated significant improvement of speech perception as DM was applied. Addition of DNR and LDSE did not significantly change the speech intelligibility scores. HINT results for normal hearing subjects also indicated significant differences between normal hearing and each hearing aid condition. The results are shown in Figure 2.

In high reverberation

The same five contrasts of interest were compared in high reverberation room. The results graphed in Figure 2. HINT results indicated that there was significant improvement for HINT scores when a hearing aid was used in omnidirectional versus unaided listening, and that the use of directionality versus omnidirectional processing provided additional improvement. In addition, scores worsened overall when the full strength processing was used in addition to the directionality. Differences between partial strength versus directionality and full strength versus partial strength were not significant.

In addition, all contrasts between normal hearing and all conditions for hearing- impaired listeners were significant. Once again, the results demonstrated that hearing-impaired listeners performed differently from the normal hearing subjects across all the hearing aid conditions. The results of the post hoc analysis for HINT in low and high reverberation are shown in Table 3.

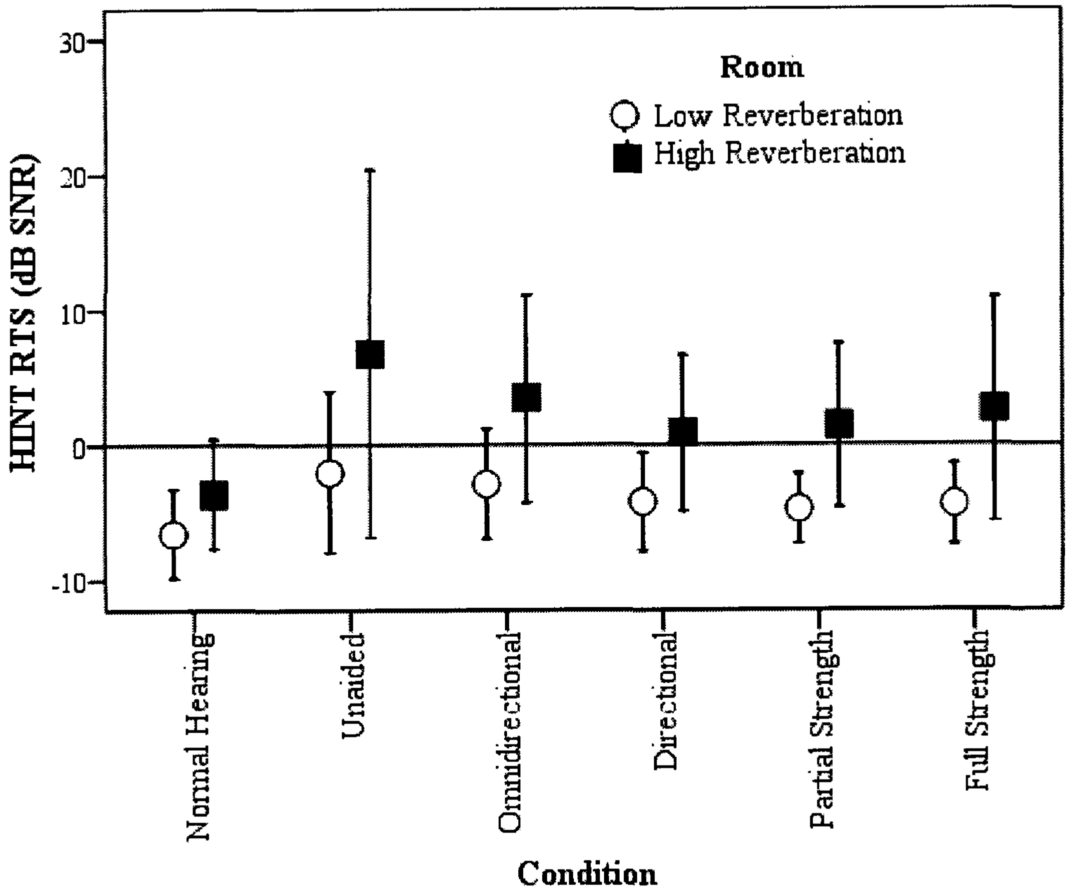

Figure 2. Means and two standard deviation bars of HINT RTS scores (dB) for five listening conditions across two listening environments.

		Unaided		Omni		Directional		Partial strength		Full strength	
		Low Rev	High Rev	Low Rev	High Rev	Low Rev	High Rev	Low Rev	High Rev	Low Rev	High Rev
Normal	*p*	**<0.001***	**<0.001***	**<0.001***	**<0.001***	**0.003***	**<0.001***	**0.005***	**<0.001***	**0.002***	**<0.001***
	t	**-4.32**	**-4.56**	**-4.75**	**-5.23**	**-3.21**	**-4.16**	**-3.05**	**-4.51**	**-3.38**	**-4.28**
Unaided	*p*			0.136	**0.005***						
	t			1.55	**3.21**						
Omnidirectional	*p*					**<0.001***	**0.001***				
	t					**4.44**	**4.11**				
Directional	*p*							0.204	0.316	0.655	**0.001***
	t							1.32	-1.03	0.45	**-3.70**
Partial strength	*p*									0.265	0.074
	t									-1.15	-1.89

Table 3. Summary of HINT paired sample t-test post hoc analysis for multiple comparisons across two listening environments. Significant values are marked with bold font and *.

Sound quality ratings

In general, sound quality ratings improved as more signal processing was added. Specific sound quality rating data are displayed across processing conditions, rooms, and masker types in Figures 3 and 4. Evaluation of significant differences among these various factors is presented below.

One participant out of 22 could not reliably perform sound quality ratings. This participant was not among the HINT outliers. This participant rated all the hearing aid settings as either all 0 or all 100 depending on the noise level (i.e., ratings in noise were all 0, ratings in quiet were all 100), despite re-instruction on the task. This person was therefore excluded from the final analyses, and MUSHRA ratings were analyzed for the remaining 21 participants (including HINT outliers). Repeated measures of ANOVA were analyzed separately for the MUSHRA ratings in quiet and noise and in each room. Pairwise comparisons on four contrasts of interest were completed: 1) omnidirectional versus directional, 2) directional versus partial strength, 3) partial strength versus full strength, 4) directional versus full strength.

Sound quality ratings in quiet

Results for sound quality testing in quiet indicated a significant effect of room type, $F(1, 20) = 12.5, p = 0.002$, listening condition, $F(1, 39) = 3.63, p = 0.036$, and a significant interaction of room by condition, $F(2, 44) = 3.38, p = 0.038$. In the sound booth, pairwise contrasts revealed no significant differences between different conditions. In the reverberation chamber, the listeners rated the directional microphone condition more highly than the omnidirectional microphone condition. No significant differences were noted between other paired comparisons. These results are shown in Figure 3.

Sound quality ratings in noise

Ratings for sound quality in noise showed significant effects of noise, $F(1, 20) = 5.28$, $p = 0.032$, signal to noise ratio, $F(1, 28) = 113.63$, $p < 0.001$, and condition, $F(1, 24) = 47.06$, $p < 0.001$. In addition, there was a significant noise by SNR interaction, $F(1, 39) = 3.80$, $p = 0.031$, noise by condition interaction, $F(1, 38) = 19.20$, $p < 0.001$, and SNR by condition interaction, $F(4, 81) = 4.48$, $p = 0.002$. Significant three-way interactions were also found for the room by noise by condition, $F(2, 50) = 3.55$, $p = 0.027$, and the room by noise by SNR interaction, $F(1, 38) = 4.41$, $p = 0.020$. There were no significant effects of room $F(1, 20) = 0.75$, $p = 0.396$, room by noise, $F(1, 20) = 0.73$, $p = 0.403$, room by SNR, $F(1, 29) = 0.17$, p = 0.777, and room by condition, $F(1, 36) = 3.18$, $p = 0.057$. Similarly, we found no significant effects of room by SNR by condition, $F(3, 79) = 0.39$, $p = 0.812$, noise by SNR by condition, $F(3, 74) = 0.33$, $p = 0.839$, and noise by SNR by room by condition, $F(3, 78) = 0.46$, $p = 0.756$. Further analyses were therefore focused on the interaction of listening condition with noise and room, collapsed across SNRs, and the interaction of listening condition with SNR, collapsed across rooms and noise types.

Sound quality ratings in stationary noise

Sound quality ratings improved significantly in the soundroom as more signal processing was added. Results were similar in the reverberation room except for partial strength and full strength conditions which were not significantly different. These results are shown in Figure 3.

Sound quality ratings in four talker babble

Across room types, addition of some amount of LDSE and DNR significantly improved sound quality over and above the use of directionality alone, although the difference between the partial strength and full strength was not considerable. These results are shown in Figure 3. The results of the post hoc analysis for sound quality ratings in low and high reverberation are also shown in Tables 4 and 5 respectively.

Sound quality ratings in varied SNRs

In general, decreases in SNR acted to reduce sound quality ratings. As more signal processing was added at -5 and +5 dB SNRs, the sound quality ratings improved, with no measurable change between partial and full strength. The results were slightly different at 0 dB SNR. Sound quality ratings were enhanced as more adaptive features were added at 0 dB SNR, indicating benefit of full strength of DNR and LDSE on subjective ratings. These results are shown in Figures 4. The results of post hoc analysis for sound quality ratings across three SNRs are also shown in Table 6.

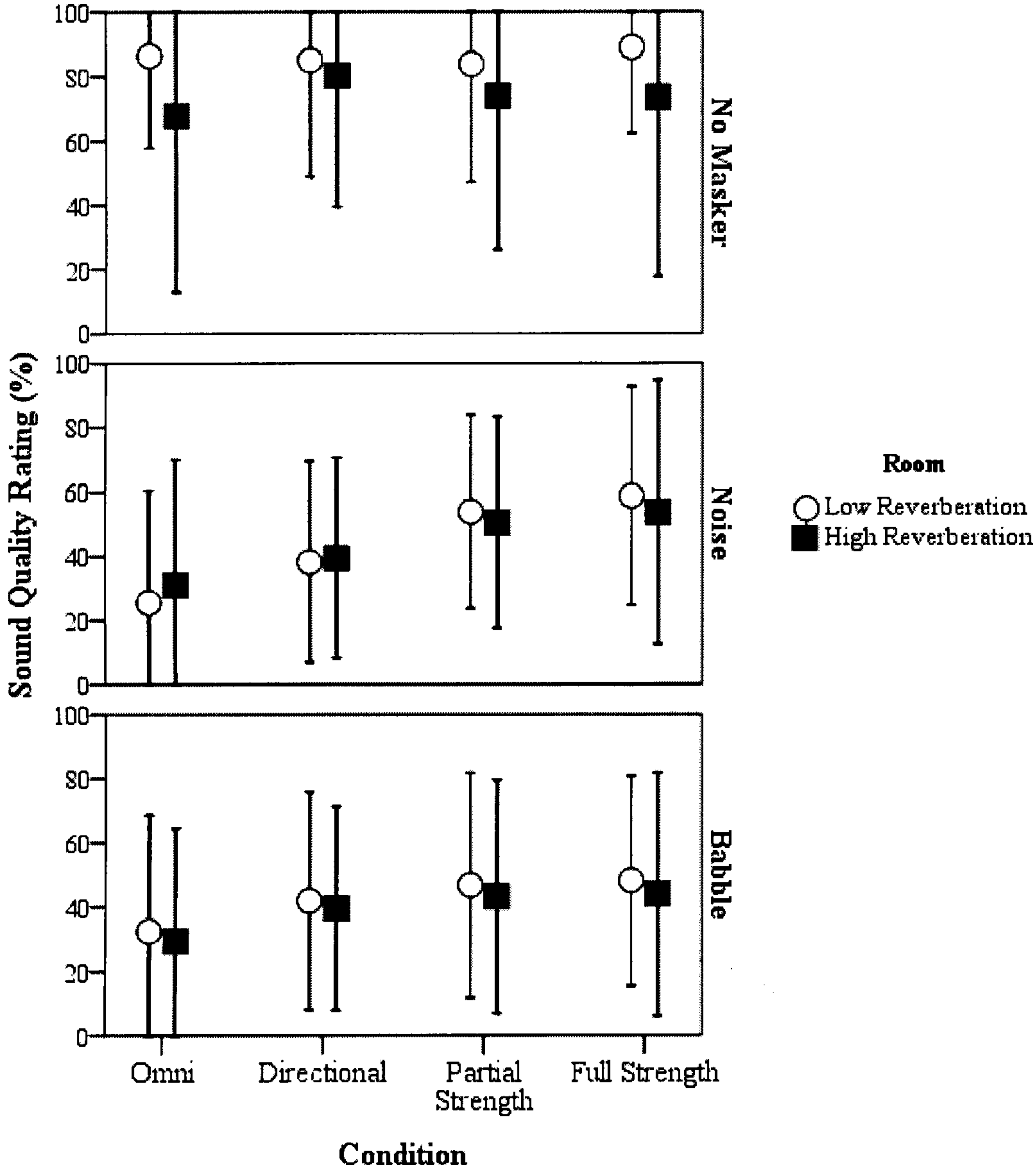

Figure 3. Means and two standard deviation bars of sound quality ratings (MUSHRA) across four listening conditions, two listening environments, and three masker types collapsed across SNRs.

		Directional			Partial strength			Full strength		
		Q	N	B	Q	N	B	Q	N	B
Omni	*p*	0.605	**<0.001***	**<0.001***						
	t	0.52	**-5.42**	**-6.04**						
Directional	*p*				0.604	**<0.001***	**0.007***	0.057	**0.001***	**<0.001***
	t				0.52	**-6.71**	**-3.00**	-2.02	**-6.33**	**-4.24**
Partial strength	*p*							0.014	**0.010***	0.467
	t							-2.70	**-2.86**	0.74

Table 4. Summary of MUSHRA paired sample t-test post hoc analysis for multiple comparisons across masker types in low reverberation (Q = Quiet, N = Noise, B = Babble). Significant values are marked with bold font and *.

		Directional			Partial strength			Full strength		
		Q	N	B	Q	N	B	Q	N	B
Omni	*p*	**0.004***	**<0.001***	**<0.001***						
	t	**-3.29**	**-4.51**	**-5.53**						
Directional	*p*				0.094	**<0.001***	**0.007***	0.199	**<0.001***	0.038
	t				1.76	**-5.73**	**-3.01**	1.32	**-3.85**	-2.22
Partial strength	*p*							0.14	0.224	0.650
	t							0.886	-1.25	-0.46

Table 5. Summary of MUSHRA paired sample t-test post hoc analysis for multiple comparisons across masker types in high reverberation (Q = Quiet, N = Noise, B = Babble). Significant values are marked with bold font and *.

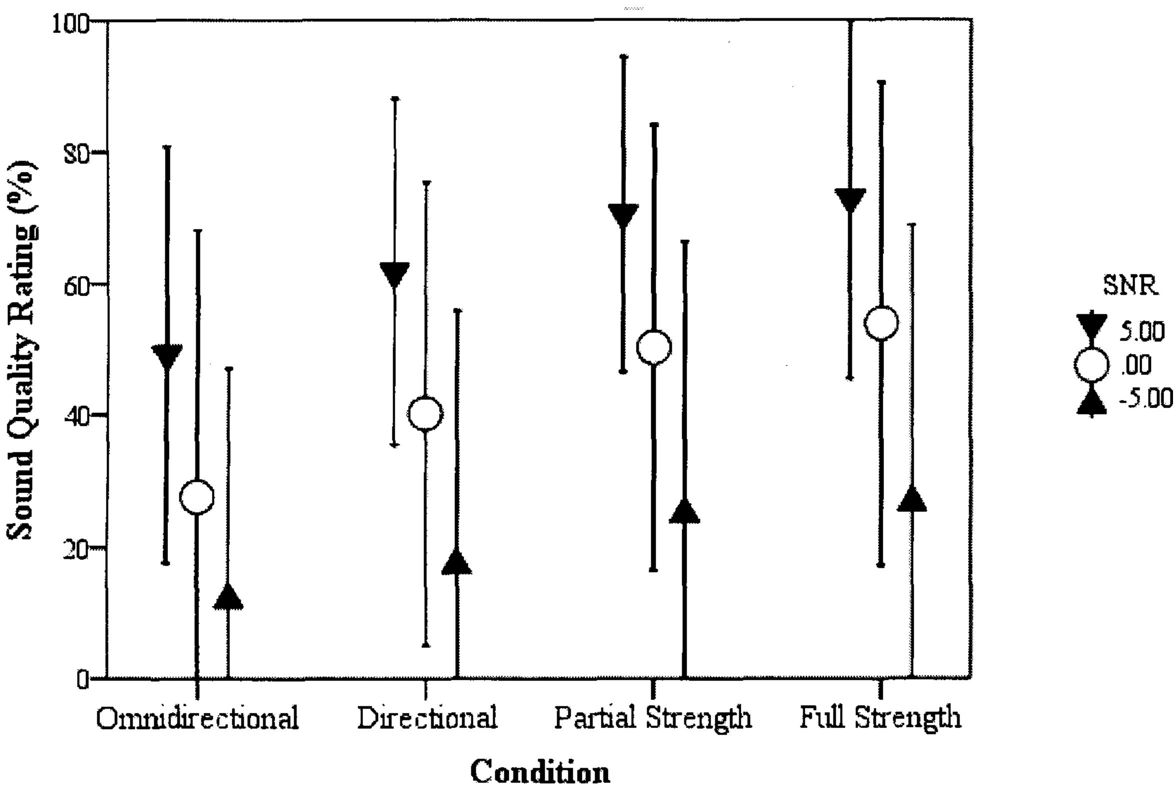

Figure 4. Means and two standard deviation bars of sound quality ratings (MUSHRA) for four listening conditions and three SNRs collapsed across listening environments and masker types.

		Directional			Partial strength			Full strength		
		-5	0	5	-5	0	5	-5	0	5
Omni	*p*	**0.003***	**<0.001***	**<0.001***						
	t	**-3.39**	**-5.91**	**-4.94**						
Directional	*p*				**0.002***	**<0.001***	**<0.001***	**0.001***	**<0.001***	**<0.001***
	t				**-3.69**	**-4.67**	**-5.32**	**-4.11**	**4.80**	**4.25**
Partial strength	*p*							0.193	**0.024***	0.156
	t							-1.34	**-2.43**	-1.47

Table 6. Summary of MUSHRA paired sample t-test post hoc analysis for multiple comparisons across three SNRs. Significant values are marked with bold font and *.

Discussion

Comparisons of unaided to aided performance

The results of this study clearly indicate the benefit of hearing aid use compared to the unaided condition especially in the reverberant environment. Speech recognition in noise results revealed that participants could obtain benefit from the hearing aids regardless of the effects of digital signal processing systems. The reason that omnidirectional processing improved speech perception was due to increased audibility of speech (Ricketts, 2001). This benefit was larger for the reverberation room than soundroom. As previously indicated, the adverse effects of noise and reverberation on speech recognition were synergistic due to the different masking impacts of noise and reverberation. The masking effect of reverberation is greater at low frequencies (Smaldino et al., 2008). This low frequency masking effect is added to the broadband effect of noise, and the loss of audibility due to SNHL primarily at high frequencies. Amplification provides the hearing aid user with increased audibility especially at higher frequencies resulting in improved speech recognition. Our results also support a recent study by Gnewikow et al. (2009) stating that hearing aid users would receive significant improvements in speech understanding with the hearing aids compared to unaided condition even in more challenging and reverberant listening environments. In summary, the results of this study indicated that individualized nonlinear amplification improved significantly the listener's ability to understand speech in noise and reverberation.

Aided performance: Effects of DSP and listening environments

The speech in noise results indicated that the directional microphone improved speech perception significantly in both rooms. This benefit of DM versus OM across varying RTs is consistent with previous studies (e.g., Leeuw & Dreschler, 1991; Ricketts, 2000; Ricketts & Hornsby, 2003; Amlani et al., 2006). The addition of DNR and LDSE to directional microphone conditions did not, however, improve speech perception in noise. In fact, the condition with the strongest setting of DNR and LDSE (full strength) reduced speech intelligibility in the reverberant environment. This may have been due to an erroneous function of the speech/noise classification algorithm. The speech/noise detector algorithms must distinguish frequency regions of speech and noise in order to apply the speech enhancement (gain) and noise reduction (attenuation). Typically, this detection is based on a combination of features such as the modulation frequency and depth, periodicity, and spectral profile (Bentler & Chiou, 2006). In reverberation, most of these features are likely distorted and the algorithm may not reliably identify speech/noise components across frequencies. As such, it may erroneously boost noise frequencies and reduce gain for speech frequencies. Because the full strength condition applies more gain/attenuation, erroneous application of DNR and/or LDSE would be more likely to occur in the full strength setting. Also, erroneous classification of signals as either speech or noise may be more likely in a reverberant environment, as reflected signals may be received at the hearing aid microphones. For example, frontal speech could be reflected and received by the hearing aid rear microphone. If this occurred, it could have contributed to the observed significant decrease in speech recognition associated with full strength processing.

Results in the low reverberation environment were somewhat different from those in the highly reverberant environment. Recall that no significant improvement in speech recognition was observed in low reverberant room, with either partial or full DNR and

LDSE. One possible reason for the lack of speech recognition benefit from DNR and LDSE in low reverberation may be the specific temporal behaviour of DM, DNR, and LDSE. During the HINT, noise was presented continuously, but speech was played during specific intervals. This may have caused an interaction between the signal processing and the moment-to-moment signal versus noise alternations within the test signal. This may have reduced the performance of DNR and LDSE resulting in no significant improvement in speech intelligibility. Level dependency of speech enhancement could also affect the benefits listeners obtain from this algorithm. As previously noted, LDSE provided more benefits for softer speech than for louder speech. In our study, we presented speech at 65 dBA to be representative of average conversational speech. This level may not be suitable level for LDSE to be properly activated. Hayes (2006) indicated that LDSE provided a 3 dB improvement in SNR when 53 dB SPL speech and 50 dB SPL traffic noise were presented simultaneously to the hearing aid. However, presenting 70 dB SPL speech and 50 dB SPL traffic noise resulted in only a 1 dB improvement in SNR. Hayes (2006) also evaluated the subjective benefit of LDSE in different presentation levels. The result of his study indicated more preferences for LDSE in soft and average speech compared to loud speech. These results are consistent with those of the present study, when considering the level-dependent nature of the speech enhancement processing.

More generally, the results of the present study are consistent with many previous studies of DNR algorithms indicating lack of their objective benefit (e.g., Walden et al., 2000; Ricketts & Hornsby, 2005; Bentler et al., 2008). The recent study of Peeters et al. (2009) also suggested no significant improvement in speech perception when a speech enhancer was added to a directional microphone. Similarly, Luts et al. (2010) stated the general lack of improvement in speech recognition using different signal enhancement algorithms.

Other test environment factors may also influence the magnitude of benefit with different digital processing in both rooms. As previously indicated, distance from the listener to speaker (Ricketts and Hornsby, 2003) as well as the number and placement of speech and competing noise sources (Ricketts, 2000) might affect directional benefit. Ricketts and Hornsby (2003) stated that directional microphones provided greater benefits when listening occurred within CD in the low reverberation time. Ricketts (2000) also demonstrated less directional benefit as speaker arrangements varied relative to 0°/180°. As shown earlier, all the speakers were located within CD in both rooms in our study. Speech and noise were also presented from two separate speakers in the front (0°) and the noise from other three speakers around the listener (90°, 180°, and 270°). The results of our study revealed that DM significantly improved speech perception in both low and high reverberation, even though some competing signals were presented from the front. One potential reason for this result may be due to the number of competing noise sources in the back and sides of the listener (three speakers) compared to that in the front of the listener (one speaker). Attenuation of competing noise and reflected energy in the rear provided by DM was still more than the amplification of those in the front. In summary, directional benefit was found in our study even though the number and placement of our speakers differed from previous investigations (Ricketts & Hornsby, 2003; Amlani et al., 2006).

Furthermore, placement of speech and noise speakers in front of the listeners did not improve the effectiveness of DNR and LDSE over DM on speech perception. As previously mentioned, DNR and LDSE algorithms were designed to improve the efficacy of DM by reducing gain for the frequency bands of noise (DNR) and increasing gain for frequency regions of speech (LDSE). Our results agree with those from previous studies in which noise was not presented from the front (e.g., Ricketts & Hornsby, 2005; Bentler et al., 2008; Peeters et al., 2009). No improvement was found in speech intelligibility in previous studies or the present study from adding DNR and/or SE to DM.

Finally, speech recognition scores for the normal hearing and hearing-impaired listeners were analyzed across signal-processing schemes (omnidirectional, directional, partial strength, and full strength) across rooms. In all hearing aid conditions and test environments, hearing-impaired listeners performed significantly poorer than normal-hearing individuals. Bentler, Palmer, and Dittberner (2004) demonstrated that directional hearing aids resulted in speech perception performance similar to normal hearing individuals in a low reverberation environment. Our study did not replicate this finding, which may be due to differences with configuration of speech and noise speakers. Bentler et al. (2004) placed one speaker with speech in the front in one corner, and six speakers with background noise at the top and bottoms of other three corners. This speaker arrangement probably yielded an increased directional effect. Another reason for this discrepancy may be due to differences between the directional processing capabilities of hearing aids (directivity index) across studies. Finally, the range of hearing loss was slightly different between two studies which may be another reason for the differences between findings. Bentler et al. (2004) evaluated individuals with mild to moderate sensorineural hearing loss, but we studied those with mild to moderately severe sensorineural hearing loss.

Since different signal processing algorithms did not restore normal speech perception in hearing-impaired listeners in the low reverberation environment, we should not expect different results in the more reverberant listening environment. As previously indicated, the synergistic effect of noise and reverberation considerably decreased speech perception in highly reverberant environments. This effect is greater in hearing-impaired individuals compared to normal hearing listeners (Nabelek & Nabelek, 1994). In general, the results of the present study suggested that hearing aid use, with or without different signal processing algorithms, resolved audibility loss but did not overcome the suprathreshold deficits associated with sensorineural hearing loss.

Sound quality

In general, the listeners assigned higher quality ratings to the combined effect of directional microphone, noise reduction, and speech enhancement. Across room types, addition of some amount of speech enhancement and noise reduction significantly improved sound quality over and above the use of directionality alone. Preference of DM to OM processing in the quiet but reverberant environment was also considerable. DM seemed to provide better sound quality by reducing reflected energy from the rear azimuths. Addition of noise to reverberation clearly demonstrated the synergistic benefit of DM, DNR, and LDSE over DM on sound quality ratings. The higher ratings assigned to the effect of DNR in our subjective test were in agreement with previous studies indicating the benefit of DNR on sound quality and sound comfort (e.g., Walden et al., 2000; Ricketts & Hornsby, 2005; Bentler et al., 2008). Our findings also supported the recent study of Peeters et al. (2009) and Luts et al. (2010) stating subjective benefit of speech enhancement algorithms. As more signal processing was added, the sound quality ratings improved even though the difference between partial strength and full strength was not measurable when the data were collapsed across signal to noise ratios.

Our findings also indicated that listeners rated significantly higher sound quality for the full strength condition compared to the partial strength in stationary noise only in the low reverberant environment. The same effect was not significant in four talker babble in either room. The performance of the DNR and LDSE algorithms was better in stationary noise due to distinct differences in modulation properties between stationary noise and speech (Bentler & Chiou, 2006). As a result, channels dominated by either speech or noise were better isolated and processed accordingly. Subjectively, this should translate into appropriate gain reduction for noisy frequencies and preservation of audibility for channels dominated by speech energy. While this did not contribute to an objective benefit in speech perception, it clearly demonstrated a subjective benefit. In high reverberation, the same trend was observed even though the relative differences between partial and full strength conditions were not significant. The sound quality of the DNR and LDSE algorithms was degraded in the presence of reverberation even with stationary noise.

Furthermore, overall sound quality ratings were improved as the SNR of listening environments was raised from -5 to +5 dB. Across rooms and noise types, addition of signal processing significantly improved the sound quality ratings at -5 and +5 dB SNRs. The difference between partial and full strength conditions was measurable only at the 0 dB SNR condition.

Our results clearly indicated that listeners preferred DM to OM in all three SNRs. It was interesting that DM still increased sound quality ratings even at low SNR (-5 dB), which was in agreement with some of previous studies (Preves, Sammeth, & Wynne, 1999; Amlani et al., 2006). Many other studies, however, have suggested that subjective measures do not demonstrate a clear advantage for DM despite clear objective benefit (Palmer, Bentler, & Mueller, 2006; Gnewikow et al., 2009).

As indicated above, addition of some amount of signal processing to the DM condition (partial strength condition) improved the sound quality ratings at all SNRs. Once again, this finding supports previous investigations indicating the benefit of processing- based algorithms on sound quality (Ricketts & Hornsby, 2005; Luts et al, 2010). In addition, our results demonstrated that the full strength of DNR and LDSE enhanced sound quality ratings at 0 dB SNR. Our results did not however support it at 5 dB SNR, as listeners did not rate higher sound quality for the full strength condition. It appears that the full strength of DNR and LDSE algorithms did not provide a significant improvement in sound quality, although the highest absolute sound quality ratings were obtained in this listening condition. Similarly, the full strength setting did not improve sound quality ratings at -5 dB SNR. Electroacoustically, it is possible that the DNR/LDSE system could have misclassified speech and noise components, resulting in distorted sound quality.

Conclusion

Taken together, the objective and subjective data indicated that combined application of different signal processing algorithms provided benefit for sound quality, but was less beneficial for speech perception in noise over and above a directional microphone. The most and only benefit of full strength of LDSE and DNR algorithms was demonstrated on sound quality ratings especially in the stationary noise, low reverberation, and 0 dB SNR. In contrast, the least benefit of those algorithms was observed on speech perception in high reverberant environments. The discrepancy between our objective and subjective results is consistent with the literature, and may be attributable to the multidimensional aspect of sound quality. Listeners judge sound quality based on perceptual dimensions of clarity (intelligibility), fullness, brightness, loudness, spaciousness, nearness, and extraneous sounds (Preminger & Van Tasell, 1995). Addition of DNR and LDSE to DM may have influenced dimensions of sound quality other than speech intelligibility in our study.

Considering objective and subjective data, we can conclude that "Partial Strength" DSP preserves speech intelligibility while enhancing speech quality across a wide range of acoustic environments. Clinical application of this setting may ensure the best compromise between sound quality benefit and benefit for speech recognition across environments, competing noise types, and signal to noise ratios. In future hearing aid design, improved algorithms for use in reverberation could support improved signal processing, and therefore address the major area of performance deficits observed in these data.

Further investigation is also required to determine if individual variability such as audiometric configuration and aging are related to the benefit participants receive from different signal processing in different acoustic environments. As our participants were mainly older adults, cognitive measures might also provide valuable information. George, Goverts, Festen, and Houtgast (2010) demonstrated that in older hearing-impaired listeners, cognitive factors in addition to auditory temporal processing mechanism might affect speech perception differences in noise and reverberation. Recent studies have also started to use cognitive measures to assess success with different signal processing and hearing aid designs in order to make them more beneficial to the auditory and cognitive performance of patients (Lunner, Rönnberg, & Rudner, 2009; Pichora-Fuller, 2009).

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Appendix A: UWO ethics approval

Western

Office of Research Ethics

The University of Western Ontario
Room 4180 Support Services Building, London, ON, Canada N6A 5C1
Telephone: (519) 661-3036 Fax: (519) 850-2466 Email: ethics@uwo.ca
Website: www.uwo.ca/research/ethics

Use of Human Subjects - Ethics Approval Notice

Principal Investigator: Dr. P. Allen
Review Number: 17091E **Review Level:** Expedited
Review Date: May 05, 2010 **Approved Local # of Participants:** 500
Protocol Title: Evaluation of hearing aid technology in adults (NCA Translational Research Unit 1)
Department and Institution: Communication Sciences & Disorders, University of Western Ontario
Sponsor: ONTARIO RESEARCH FUND #RE-03-009
Ethics Approval Date: May 19, 2010 **Expiry Date:** August 31, 2014
Documents Reviewed and Approved: UWO Protocol, Letter of Information and Consent, Participation in Research Opportunities, Telephone Script, Poster.
Documents Received for Information:

This is to notify you that The University of Western Ontario Research Ethics Board for Health Sciences Research Involving Human Subjects (HSREB) which is organized and operates according to the Tri-Council Policy Statement: Ethical Conduct of Research Involving Humans and the Health Canada/ICH Good Clinical Practice Practices: Consolidated Guidelines; and the applicable laws and regulations of Ontario has reviewed and granted approval to the above referenced study on the approval date noted above. The membership of this REB also complies with the membership requirements for REB's as defined in *Division 5 of the Food and Drug Regulations.*

The ethics approval for this study shall remain valid until the expiry date noted above assuming timely and acceptable responses to the HSREB's periodic requests for surveillance and monitoring information. If you require an updated approval notice prior to that time) ust request it using the UWO Updated Approval Request Form.

During the course of the research, no deviations from, or changes to, the protocol or consent form may be initiated without prior written approval from the HSREB except when necessary to eliminate immediate hazards to the subject or when the change(s) involve only logistical or administrative aspects of the study (e.g. change of monitor, telephone number). Expedited review of minor change(s) in ongoing studies will be considered. Subjects must receive a copy of the signed information/consent documentation.

Investigators must promptly also report to the HSREB:

a) changes increasing the risk to the participant(s) and/or affecting significantly the conduct of the study;
b) all adverse and unexpected experiences or events that are both serious and unexpected;
c) new information that may adversely affect the safety of the subjects or the conduct of the study.

If these changes/adverse events require a change to the information/consent documentation, and/or recruitment advertisement, the newly revised information/consent documentation, and/or advertisement, must be submitted to this office for approval.

Members of the HSREB who are named as investigators in research studies, or declare a conflict of interest, do not participate in discussion related to, nor vote on, such studies when they are presented to the HSREB.

Chair of HSREB: Dr. Joseph Gilbert
FDA Ref. #: IRB 00000940

Ethics Officer to Contact for Further Information			
Janice Sutherland (jsutherl@uwo.ca)	☐ Elizabeth Wambolt (ewambolt@uwo.ca)	☑ Grace Kelly (grace.kelly@uwo.ca)	☐ Denise Grafton (dgrafton@uwo.ca)

This is an official document. Please retain the original in your files. cc: ORE File

UWO HSREB Ethics Approval - Initial
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Appendix B: Listener instructions

You are going to hear some noise. When you are ready, press START on the screen. A man's voice will play. You do not have to repeat the sentences he is saying. You just have to listen to how the man's voice sounds. On the screen, there will be four sliders (labeled A, B, C, and D). The hearing aids you are wearing will change settings each time you touch A, B, C or D on the screen. You must touch the letters to change the hearing aid settings. Each time you change settings, listen to the sound for at least 30 seconds before rating the hearing aid setting. Your task is to listen carefully to the speech voice, and indicate the overall quality of each setting in relation to each other by adjusting the corresponding sliders. Note that the overall quality stands for your overall impression of the speech, which includes speech clarity, presence of distortion, background noise, and other artifacts. Rate the sound of the hearing aid setting by moving the slider up towards excellent or down towards poor. To change the hearing aid setting, press A, B, C, or D. Each time you change settings, please listen to the sound for at least 30 seconds before making your rating. You can always go back to a previous setting and change your rating just by touching the A, B, C, or D buttons and moving the slider again. You have to move each slider up or down to move on to the next set. The program will stop automatically when you are done. If you need us to stop the program at any time, please raise your hand. Any questions?