J Am Acad Audiol 17:681–707 (2006)

Effect of Multichannel Digital Signal Processing on Loudness Comfort, Sentence Recognition, and Sound Quality

Karen M. Mispagel*
Michael Valente*

Abstract

This study evaluated the effect of increasing the number of processing channels from 32- to 64-signal processing channels on subjects' loudness comfort and satisfaction, sentence recognition, and sound quality of his or her own voice. Ten experienced hearing aid users with mild-to-moderate sensorineural hearing loss wore behind-the-ear (BTE) hearing aids with Adaptive Dynamic Range Optimization (ADRO™) signal processing for a period of six weeks in the 32-channel and 64-channel conditions. Results revealed no significant differences in loudness comfort or satisfaction for the majority of sound samples as measured by the *Subjective Loudness Test* and *Environmental Sounds Questionnaire.* No significant differences in sentence recognition between the two processing conditions were found as measured by the *Hearing In Noise Test* (HINT). Additionally, no subjective differences in sound quality of subjects' own voice were determined by the *Listening Tasks Questionnaire*.

Key Words: Adaptive dynamic range optimization (ADRO), *Hearing In Noise Test* (HINT), multichannel compression, R-Space™ noise

Abbreviations: ADRO™ = Adaptive Dynamic Range Optimization; ANSI = American National Standards Institute; BTE = behind the ear; CLL = comfortable listening level; DSP = digital signal processing; HINT = *Hearing In Noise Test*; IRB = institutional review board; MIL = most intelligible level; OB = octave band; PTAlf = low-frequency pure-tone average; RTS = reception threshold for sentences; SNR = signal to noise ratio

Sumario

Este estudio evaluó el efecto de incrementar de 32 a 64 el número de canales de procesamiento de la señal, sobre el nivel agradable de intensidad subjetiva del sujeto, y la satisfacción, el reconocimiento de frases y la calidad del sonido de su propia voz. Diez sujetos con experiencia en el uso de auxiliares auditivos, con hipoacusias sensorineurales leves a moderadas, utilizaron auxiliares retroauriculares (BTE) con Optimización Adaptativa del Rango Dinámico (ADRO™) para procesamiento de la señal, por un periodo de seis semanas en condiciones de 32 y 64 canales. Los resultados no revelaron diferencias significativas en el nivel confortable de intensidad subjetiva o en la satisfacción para la mayoría de la muestra de sonidos, medidos por medio de la *Prueba de Intensidad Subjetiva* y el *Cuestionario de Sonidos Ambientales*. Tampoco de encontraron diferencias significativas en el reconocimiento de frases entre las dos condiciones de procesamiento, medidas con la *Prueba de Audición en Ruido* (HINT). Además, no se determinaron diferencias subjetivas en la

*Department of Otolaryngology-Head and Neck Surgery, Division of Adult Audiology, Washington University School of Medicine, St. Louis, Missouri

Karen Mispagel, Washington University School of Medicine, Department of Otolaryngology-Head and Neck Surgery, Division of Adult Audiology, 660 South Euclid Ave., Campus Box 8115, St. Louis, MO 63110; Phone: 314-362-7490; Fax: 314-747-5593; E-mail: mispagelk@ent.wustl.edu

calidad de sonido de la propia voz del sujeto, determinada por el *Cuestionario de Tareas de Audición*.

Palabras Clave: Optimización adaptativa del rango dinámico (ADRO), *Prueba de Audición en Ruido* (HINT), compresión multicanal, ruido R-Space™

Abreviaturas: ADRO™ = Optimización adaptativa del rango dinámico; ANSI = Instituto Americano Nacional de Estándares; BTE = retro-auricular; CLL = Nivel confortable de audición; DSP = procesamiento digital de la señal; HINT = Prueba de Audición en Ruido; IRB = comité institucional de revisión; MIL = nivel más inteligible; OB = banda de octava; PTAlf = promedio tonal puro a bajas frecuencias; RTS = umbrales de recepción para frases; SNR = tasa señal/ruido

Digital signal processing (DSP) algorithms have become increasingly more complex since DSP hearing aids became widely commercially available in the mid-1990s. DSP hearing aids are now available with as many as 32 channels of signal processing. As the number of processing channels has increased, studies have investigated the potential advantages of multichannel signal processing (Moore and Glasberg, 1986; Kiessling and Steffens, 1991). Relative to single-channel compression, multichannel processing can increase intelligibility because of the increase in the audibility multichannel compression provides for low level input sounds (Dillon, 2001). Additionally, multichannel compression allows the frequency response of hearing aids to be more easily controlled by providing programming flexibility not available in single-channel processing (Kuk, 2002). Multichannel processing has also been critical in the development of more effective noise suppression and feedback management strategies. With an increased number of processing channels, noise reduction and feedback strategies can more precisely decrease gain in channels in which noise or feedback is occurring with less reduction in the adjacent frequency channels or those containing speech.

Although advantages of multichannel signal processing are documented, potential negative side effects of increasing the number of channels have been investigated (Crain and Yund, 1995; Moore et al, 1999; Stone and Moore, 1999, 2002, 2005; Agnew and Thornton, 2000). Possible disadvantages of multichannel signal processing include channel summation, temporal (or spectral) smearing, and increased group delay.

Channel summation occurs when the output in each channel of a multichannel hearing aid combines, resulting in a wider bandwidth than any individual channel and an increase in overall output (Dillon, 2001). In general, the more channels present and the higher the compression ratio in each channel, the greater the summation effect (Kuk and Ludvigsen, 2003). If unaccounted for, this increase in output as a result of multichannel processing could lead to greater loudness discomfort for the hearing aid user when compared with hearing aids with fewer channels. Kuk and Ludvigsen (2003) demonstrated this by evaluating the output of four hearing aids with different numbers of channels. Results revealed that the output of the 15-channel hearing aid was almost 10 dB greater than the single-channel aid and at least 5 dB greater than the two- and three-channel hearing aids. The results from this investigation suggested that channel summation may need to be accounted during the fitting of hearing aids. Dillon (2001) recommended a reduction in real ear saturation response (RESR) levels for multichannel hearing aids in which the output is controlled independently in each channel. The amount of recommended reduction increased as the number of channels increased: two channels—5 dB reduction; three channels—7 dB reduction; four channels—9 dB reduction; and five channels—10 dB reduction. No recommendations were made for hearing aids containing more than five channels.

As mentioned above, another major concern of increasing the number of signal processing channels is temporal (or spectral) smearing. Temporal smearing occurs when the intensity difference between the peaks and troughs in the speech envelope is reduced. In multichannel hearing aids, as the number of channels increases, the intensity difference decreases (Kuk, 2002). This reduction in temporal contrasts could cause a reduction in speech recognition, especially for those with greater than a moderate hearing loss who rely on temporal contrasts for speech recognition (Van Tassell et al, 1987). Numerous studies investigating the effect of the number of channels on speech recognition have been undertaken, and the results from these studies have been variable (Summerfield, 1992; Crain and Yund, 1995; Moore et al, 1999; Stone and Moore, 1999, 2002, 2005; Agnew and Thornton, 2000). Crain and Yund (1995) investigated the "degradation of vowel and stop-consonant discrimination as a function of the number of channels and compression ratios" (p. 530). The results for hearing-impaired subjects indicated when the multichannel compression processing strategy was customized to the subjects' hearing loss utilizing subject specific threshold and loudness discomfort information, no significant change in vowel discrimination performance was demonstrated as a result of increasing the number of channels. Additionally, it was reported that significant discrimination errors with vowel spectra were present only when compression ratios were high (value not reported by the authors) in each channel and when the number of channels was greater than eight.

Moore et al (1999) evaluated the effectiveness of multichannel compression with one, two, four, and eight channels by measuring subjects' speech reception threshold (SRT) with the *Hearing In Noise Test* (HINT) sentences. Although only a slight benefit of multichannel compression was seen in this study, the authors theorized that "further increases in the number of compression channels, with corresponding reductions in bandwidth of each channel, might lead to a system that was more effective in improving the detectability of portions of the speech target falling in the spectral dips in background sound" (p. 409).

Yund and Buckles (1995) investigated the effect of increased number of processing channels on speech recognition of mild-to-moderately severe hearing-impaired subjects. Reported results indicated a highly significant effect for number of channels. Increasing from four to eight processing channels improved speech recognition, but above eight channels, no further improvement was found.

The final potential disadvantage of increasing the number of signal processing channels is group delay. "Processing time" or "group delay" is defined as the finite time delay created as an input signal passes through a hearing aid from the microphone to the receiver (Agnew and Thornton, 2000). The group delay in digital hearing aids is considerably longer in comparison to analog hearing aids due to the complex conversion of the input sound signal into discrete quantities for signal processing. Whereas the time required for analog hearing aids to process input signals is very short, a few tenths of a millisecond (msec), the time needed for DSP can vary widely depending on the DSP algorithm. In general, as the amount of processing increases, so does the processing time or group delay (Frye, 2001).

Previous research has demonstrated that long group delay can negatively affect speech production and perception for normal-hearing and hearing-impaired patients (Summerfield, 1992; Stone and Moore, 1999, 2002, 2005; Agnew and Thornton, 2000). Specifically, concerns of auditory confusion (Summerfield, 1992) and degradation of speech production and perception of subjects' own voice (Stone and Moore, 1999, 2002, 2005; Agnew and Thornton, 2000) as a result of delay have been investigated.

Auditory confusion can occur when there is a delay between the hearing aid user observing the movement of the talker's lips and hearing the sound of his or her voice. Summerfield (1992) reported that sound can lag the visual image by more than 80 msec before confusion will occur. Therefore, he recommended that processing for hearing aid users with severe-to-profound hearing loss be as short as possible, but group delays as long as 40 msec would be acceptable.

Stone and Moore (1999) reported on the effect of delay on a subject's own speech production and perception of his or her own voice for normal-hearing populations using a simulation of hearing loss. They reported that delays greater than 20 msec can lead to the perception of an "echo" in the subjects' own voice, whereas delays less than 10 msec might lead to a perception of a subtle change in the timbre of the sound. In a follow-up study, Stone and Moore (2005) utilized hearing-impaired subjects to measure the effect of group delays (13–40 msec) on perception of the subject's own voice and speech production. It was concluded that subject disturbance to the sound of his or her voice increased with increasing group delay. Additionally, subjects with low-frequency (500, 1000, and 2000 Hz) hearing loss greater than 50 dB HL were significantly less disturbed than those subjects with less low-frequency hearing loss. Specifically, the results showed that delays greater than 15 msec can be unacceptable to listeners with low-frequency hearing loss around 35 dB HL, but those with more moderate-to-severe hearing loss or very mild hearing losses in the low frequencies may be able to tolerate longer delays.

Stone and Moore (2002) analyzed objective and subjective measures of effects of hearing aid delay on speech production and perception in two different environments with the goal of defining an upper limit to permissible processing delay. They concluded that normal-hearing subjects reported that disturbing effects on perception become significant when delays exceeded 15 msec in an office environment and 20 msec in a test booth. Objective measures of speech production did not show any significant negative effects of delay until the delay reached 30 msec. As a result of these findings, Stone and Moore (2002) recommended DSP hearing aids, which should be able to incorporate delays as long as 15 msec with few negative side effects. Additionally, the amount of tolerable processing delay increased by 4 msec in reverberant environments compared to a near anechoic environment.

Agnew and Thornton (2000) investigated the amounts of delay that were just noticeable and considered objectionable with 18 normal-hearing engineers to determine a worse case limit for DSP hearing aid design. The listeners in this study reported that time delays greater than 10 msec were objectionable 90% of the time, a significantly shorter time delay than what was published by Stone and Moore (1999, 2005).

Overall, results from past research that examined the effect of increasing the number of channels in DSP hearing aids suggest that the issues of channel summation, temporal smearing, and group delay need to be addressed when fitting multichannel hearing aids. These effects have been well researched in other compression strategies, yet how these issues affect loudness comfort and satisfaction, sentence recognition, and sound quality of a hearing aid user's own voice has not been addressed in Adaptive Dynamic Range Optimization (ADRO™) signal processing.

The current study utilizes Adaptive Dynamic Range Optimization (ADRO™) signal processing in 32- and 64-channel processing strategies. ADRO™ is a slowly adapting DSP that controls the output level of a set of narrow-frequency bands so that the levels fall within a specified dynamic range. The 32-channel processing strategy has a 250 Hz bandwidth for each channel from 125 to 8000 Hz. The 64-channel processing strategy has a bandwidth of 125 Hz for each channel. By using narrow channels in the hearing aid, there is great flexibility to shape the maximum gain, maximum output levels, comfort targets, and audibility targets in each channel allowing ADRO™ to be fit to a wide range of hearing losses.

The dynamic range of ADRO™ processing is defined by the threshold of audibility and a comfortable level within each frequency channel for an individual. ADRO™ measures the peaks and troughs of the output signal unlike most amplifiers that measure the average level of the input signal. Since ADRO™ does not make an assumption regarding the input signal dynamic range, it can maintain comfort and audibility of a wide variety of sounds, not just speech (Blamey, 2005). A set of rules is implemented to control the output levels with the goal of keeping the output signal level within the optimum dynamic range. First, the "comfort rule" requires 90% of the output levels to be below the comfort target level in each frequency channel. This rule ensures that sounds are not too loud. Next, the "audibility rule" requires 70% of the output levels to be above the audibility target in each channel. This ensures that sounds are not too soft. The "audibility rule" is applied only if the "comfort rule" is satisfied. The magnitude of approximate increase or decrease in gain or "slew rate" can be changed from the default of 3 dB/sec to 6 dB/sec through the manufacturer's software. The "hearing protection rule" limits the output level in each channel so that it never exceeds the maximum output level. Finally, the "background noise rule" limits the maximum gain in each channel to ensure low-level background noise is not overamplified (Blamey, 2005). As a result of the ADRO rules, gain does not change unless the comfort or audibility rule is violated.

The studies described previously all have examined the impact of increasing the number of processing channels in other compression strategies, but it is unknown if similar results would occur with ADRO™ processing. Martin et al (2001) compared ADRO™ signal processing to a linear fit hearing aid utilizing open-set sentences at multiple intensity levels. Subject performance with ADRO™ processing was significantly better than the linear processing at 55 and 65 dB SPL (15.9% improvement at 55 dB and 36% dB improvement at 65 dB). In a reverse-block design study by Blamey et al (2004), the difference in sentence recognition in quiet and noise between a nine-channel-wide dynamic range compression (WDRC) and a 64-channel ADRO™ signal processing strategy was examined. The results revealed that the subjects' mean performance with ADRO™ processing was statistically significantly better in quiet and noise than WDRC processing, although it is unclear if the slight improvement (7.85% word score and 6.41% phoneme score in quiet and 7.25% in noise) in performance is a result of increasing the number of processing channels, of difference in fitting strategy, or of differences in amplification strategies.

The current study utilized a hearing aid with ADRO™ processing in 32- and 64-channel strategies programmed using the same in situ fitting method. This comparison between 32- and 64-channel ADRO™ processing examined whether detrimental side effects (i.e., channel summation, temporal smearing, or group delay) as result of increasing the number of processing channels occurred that could lead to decreased loudness comfort or satisfaction, poorer sentence recognition, or decreased sound quality of the subject's own voice. Furthermore, if laboratory benefits of increasing the number of ADRO™ processing channels are present, these same benefits should ideally be accompanied by increased real-world benefits (i.e., external validity) in order to establish the effectiveness of the processing strategy. If, on the other hand, laboratory benefits of increasing the number of ADRO™ processing channels are not present, then it can be assumed that increasing the number of channels would not provide any significant benefit, and performance with a 32-channel processor would provide the same level of performance as a 64-channel processor.

The primary objectives of the present study were to determine if:

1. Significant differences in loudness comfort were present between 32- and 64-channel processing strategies as measured by the *Subjective Loudness Test* in the aided condition and the *Environmental Sounds Questionnaire* in the unaided and aided conditions.
2. Significant differences in satisfaction were present between 32- and 64-channel processing strategies as measured by the *Subjective Loudness Test* in the aided condition and the *Environmental Sounds Questionnaire* in the unaided and aided conditions.
3. Significant differences were present between 32- and 64-channel processing strategies in an adaptive directional microphone mode for the reception threshold for sentences (RTS in dB) required for 50% performance on the *Hearing In Noise Test* (HINT) sentences presented at 0° and diffuse R-Space™ noise (eight loudspeaker array) fixed at 65 dBA.
4. Significant differences were present between 32- and 64-channel processing strategies in the adaptive directional microphone mode for reception threshold for sentences (RTS in dB) required for 50% performance on the *Hearing In Noise Test* (HINT) sentences presented at 0° in quiet.
5. Subjective differences in sound quality of subjects' own voice quality were present between the 32- and 64-channel processing strategies as measured by the *Listening Tasks Questionnaire*.

PROCEDURES

Subjects

Ten adults (8 males, 2 females; mean age = 70.8 years; SD = 11.4 years) with mild-to-moderately severe bilateral symmetrical sensorineural hearing loss (ANSI [American National Standards Institute], 1996) participated in this investigation. Symmetry was defined as no greater than a 15 dB HL difference in interaural thresholds at 250–4000 Hz. The magnitude of hearing loss was within the recommended fitting range for the experimental behind-the-ear (BTE) hearing aids. Pure-tone thresholds and acoustic immittance were measured during the first test session. All subjects exhibited no significant air-bone gap at any frequency and normal tympanograms. Mean word recognition scores under earphones in quiet at the most intelligible level (MIL) were 75.6% (SD = 10.7%) and 77.4% (SD = 13.1%) for the right and left ears respectively. The presentation level to assess word recognition at MIL is determined by monitored live voice presentation (voice peaking at 0 on the VU [volume units] meter) of conversational speech and asking the subject to indicate when the presentation level was comfortably loud and most intelligible.

Figure 1 illustrates the mean hearing thresholds (average of right and left ears) at 250 to 8000 Hz. All subjects had prior experience with binaural digital, adaptive directional amplification for at least one year with their current hearing aids. See Table 1 for subject specific hearing aid information.

The subjects were recruited from the Washington University School of Medicine Adult Division of Audiology. When subjects were recruited for the study, they were asked to sign the institutional review board (IRB) approved consent form. Subjects were told the purpose of the study was to evaluate two different processing strategies, but they were not informed about the signal processing or any other aspect of the experimental hearing aid. Finally, to compensate the subjects for his or her efforts, subjects were provided $200 at the conclusion of the study.

Fitting the Experimental Hearing Aids

ADRO™ processing in 32- and 64-channel amplification strategies was placed

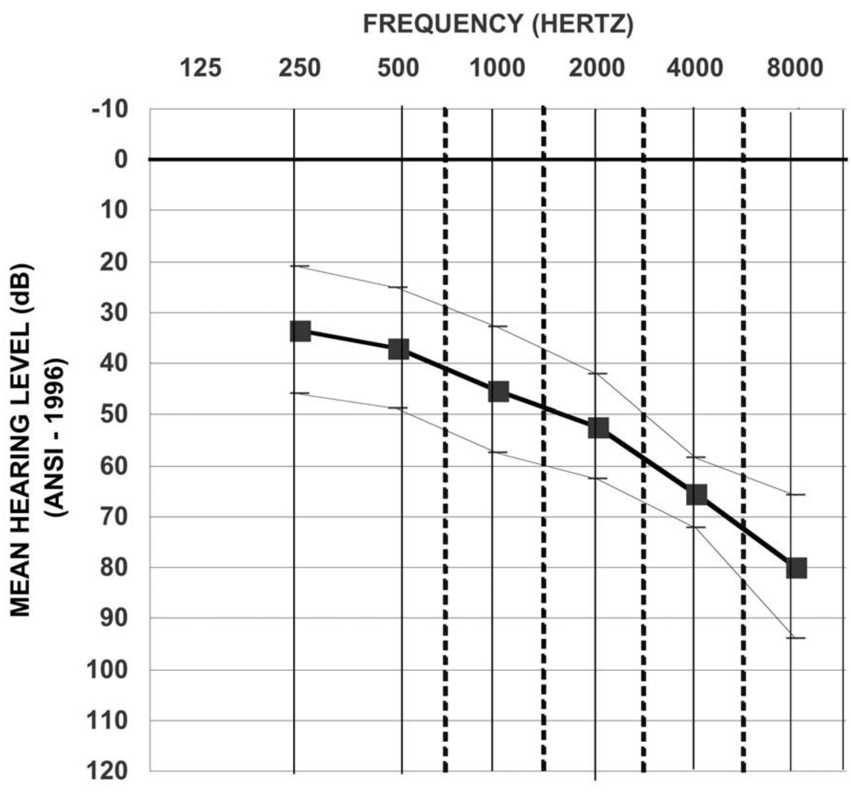

Figure 1. Mean hearing thresholds averaged for the right and left ears. Also provided is ±1 SD.

Table 1. Subject Hearing Aid and Earmold Information

Subject	Current Hearing Aid	Yrs. with Current Aid	Yrs. of Use (Total)	Earmold	Vent	Tubing
1	Diva	2.0	10.0	shell	2.4 mm	3 mm
2	Claro	3.0	3.0	1/2 shell	pressure	4 mm
3	Diva	2.5	10.0	skeleton	1.5 mm	3 mm
4	Canta	1.6	5.4	skeleton	4.0 mm	3 mm
5	Diva	2.5	6.0	skeleton	R-2.4; L-4.0	3 mm
6	Diva	1.0	12.0	3/4 shell	4.0 mm	3 mm
7	Diva	1.8	8.2	skeleton	4.0 mm	3 mm
8	Diva	2.0	22.0	skeleton	2.4 mm	3 mm
9	Diva	1.4	4.0	shell	2.4 mm	3 mm
10	Claro	4.0	10.0	shell	1.8 mm	3 mm

into a BTE hearing aid supplied by one of the sponsors. The aid had a volume control wheel and push button to access multiple programs. A software program was utilized to alternate between the two signal processing strategies throughout the study. After changing strategies, coupler and real ear measures using the Frye 6500CX were performed to ensure accurate hearing aid performance. Additionally, the group delay was measured on the Frye 6500CX test box. The measurement was performed by using a broadband impulse signal and a 20 msec time window for each amplification strategy (Frye, 2001). The mean group delay for the 32-channel processing was 6.9 msec. The mean group delay for the 64-channel processing strategy was 12.8 msec.

The experimental hearing aids were initially fit using the manufacturer's (Dynamic Hearing, LTD) recommended fitting procedure. Briefly, the hearing aids were coupled to NOAH utilizing a QuickCOM (manufactured by AVR Communications LTD) interface box and the aids placed in the ear canal. The QuickCOM box is a manufacturer specific interface box that allows for faster communication between NOAH software and hearing aids. The subject's own earmolds were used to ensure a comfortable fit. See Table 1 for subject earmold information. The results from the audiometric test were used to predict the initial comfortable levels at seven frequencies spaced at half-octave intervals from 500 Hz to 4000 Hz. Then, the individual in-situ comfortable listening levels (CLL) were measured using sixth-octave bands of noise in a bracketing procedure (2 dB and 4 dB step sizes for up and down, respectively). To measure CLL, a seven-point categorical loudness scale was used. The seven categories were (1) very soft, (2) soft, (3) comfortable but slightly soft, (4) comfortable, (5) comfortable but slightly loud, (6) loud but OK, and (7) uncomfortably loud. Subjects were instructed to assign a loudness category when each stimulus was presented. The 50% intensity level that the subject judged to be at the highest level of "comfortable" on the loudness scale was selected as the comfort target for each of the seven channels. Then the noise was presented in a sweep across frequencies to ensure balance in comfortable loudness judgments across frequencies. If necessary, the comfort target was adjusted until the subject judged the stimuli to be equal in loudness across the frequency channels. Each hearing aid was programmed with an omnidirectional microphone in Program #1, adaptive directional microphone in Program #2, and telecoil in Program #3. The volume control was programmed to provide a 20 dB SPL range (10 dB SPL up and 10 dB SPL down). Finally, fine-tuning adjustments were made to the initial frequency response to address any concerns related to feedback and the occlusion effect.

After the initial fit, subjects were instructed to complete the *Listening Tasks Questionnaire* (see Appendix 1) before returning for the next appointment. This was a 22-item questionnaire targeting loudness comfort and sound quality for a variety of stimuli and listening situations typically encountered in the real world. Seven environmental sounds identified in the questionnaire, three loud sounds, two average sounds, and two soft sounds were used to assess loudness comfort. The *Listening Tasks Questionnaire* also included questions regarding the subjects' own voice quality and use of the directional microphone and telecoil programs. The subjects' responses were used as a guide for fine-tuning the frequency response of the study hearing aid at the subjects' fine-tuning appointments (see Figure 2 for subject visit schedule) and to gather subjective information regarding subjects' perceptions of their own voice quality. The *Listening Tasks Questionnaire* was also completed prior to returning for the evaluation appointments of each of the two processing conditions to ensure subjects had no loudness or sound quality concerns before objective testing was performed.

Subjects wore the hearing aids with the initial fit for one week and returned for fine-tuning to address any subjective concerns identified by the *Listening Tasks Questionnaire*. Of the 10 subjects, four required fine-tuning one week following the 32-channel signal processing fitting, and six required fine-tuning one week after the 64-channel signal processing fitting. The most common fine-tuning performed in each of the processing strategies was to reduce the high-frequency maximum gain at 6k Hz to eliminate feedback. Following the one-week fine-tuning appointment, subjects were given the *Listening Tasks Questionnaire* and *Environmental Sounds Questionnaire* (see Appendix 2) to complete before the next visit.

The *Environmental Sounds Questionnaire* addressed loudness comfort and satisfaction for sounds such as car noise, washing machine, phone ringing, and so forth. This questionnaire differed from the *Listening Task Questionnaire* in that it identified 18 environmental sounds, and subjects were asked to assign an eight-point categorical loudness scale and five-point satisfaction rating for unaided and aided conditions. The eight-point loudness categorical scale was (1) did not hear, (2) very soft, (3) soft, (4) comfortable but slightly soft, (5) comfortable, (6) comfortable but slightly loud, (7) loud but okay, and (8) uncomfortably loud. The five-point satisfaction scale included (1) not good at all, (2) not too good, (3) okay, (4) pretty good, and (5) just right. At this point, subjects wore the aids for four weeks before returning for measuring sentence recognition in quiet and noise with the HINT and completion of the *Subjective Loudness Test*. The protocol was repeated for the alternate fitting rationale.

After subjects wore the hearing aids in both processing strategies for five weeks, the hearing aids were reprogrammed to the first randomly assigned rationale for another one-week trial. The purpose of this crossover design was to refamiliarize subjects with the processing strategy prescribed in the first trial period. Following this one-week trial, subjects were asked to report any subjective differences between the two processing strategies.

At the conclusion of the study, the directional microphone performance of each hearing aid was verified by measuring the front-to-back ratio via probe-tube measures in Program 2. To perform this measurement, each hearing aid was coupled to a nonvented earmold and placed into an artificial ear in a double-walled sound booth. A 70 dB SPL, ANSI composite noise was presented at 0° in front of the hearing aid, and the output was saved. Then, the hearing aid was rotated so that the signal was at the maximum angle of reduction, and again the output was saved. Figure 3 illustrates the mean front-to-back difference and one standard deviation for

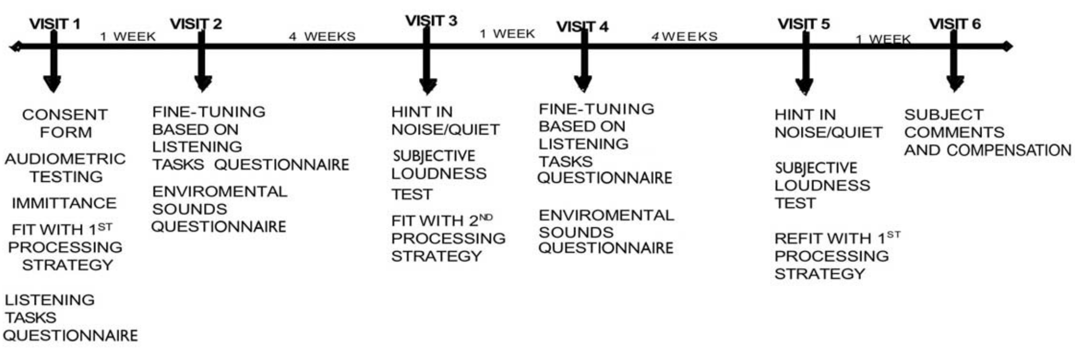

Figure 2. Subject visit schedule.

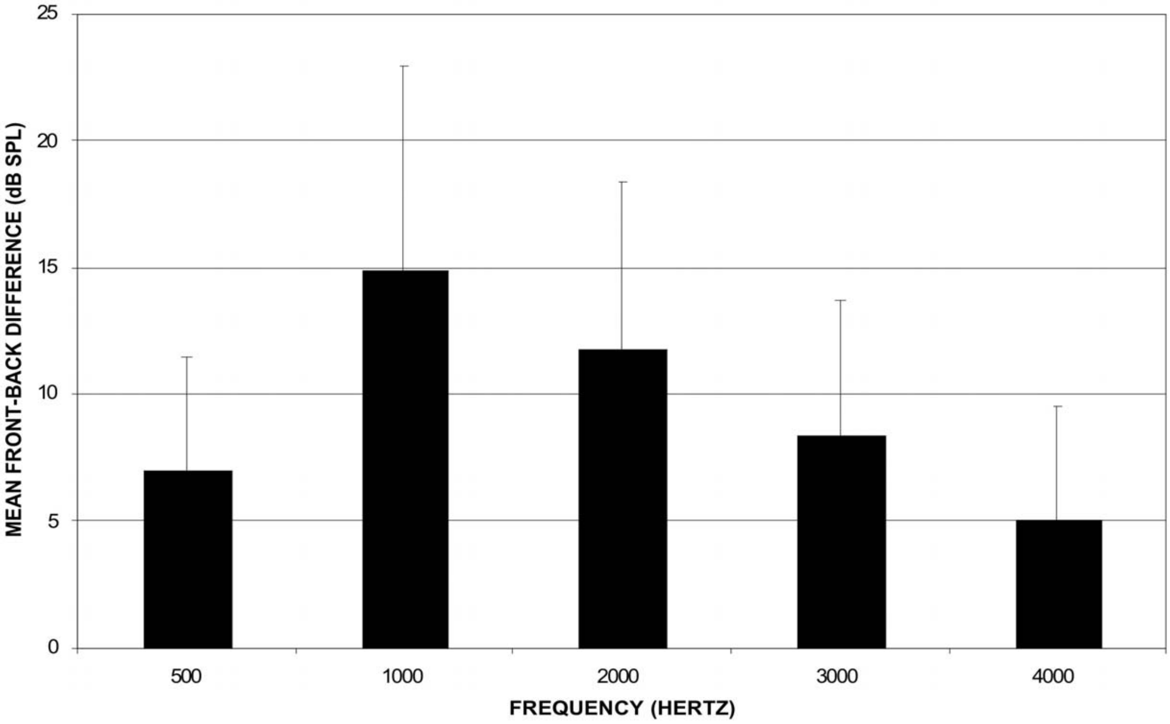

Figure 3. Mean front to back difference (in dB SPL) at 500, 1000, 2000, 3000, and 4000. Error bars represent ±1 SD.

the 20 study hearing aids across five discrete frequencies.

Hearing In Noise Test (HINT)

The HINT (Nilsson et al, 1994) consists of 250 sentences (25 lists of 10 sentences per list) read by a male speaker. The sentences are of approximately equal length (six to eight syllables) and difficulty (first-grade reading level) and have been digitally recorded for standardized presentation. The HINT estimates the signal-to-noise ratio (SNR) at which the sentences, embedded in noise, can be repeated correctly 50% of the time. This type of measure is useful because it enables accurate, reliable estimation of speech recognition in noise for context-rich speech materials.

The administration of the HINT requires two lists to be presented (ten sentences each) for each experimental condition. The first sentence was presented 10 dBA below the attenuator setting necessary for the noise to be presented at 65 dBA. The first sentence is repeated, increasing the level of presentation by 4 dB, until repeated correctly by the subject. Subsequently, the intensity level is decreased by 4 dB, and the second sentence is presented. The stimulus level is raised (incorrect response) or lowered (correct response) by 4 dB after the subject's response to the second, third, and fourth sentences. The step size is reduced to 2 dB after the fourth sentence, and a simple up-down stepping rule is continued for the remaining 15 sentences. The calculation of the SNR necessary for 50% sentence recognition is based on averaging the presentation level of sentences 5 through 20, plus the calculated intensity for the 21st presentation.

HINT reception threshold for sentences (RTS) was obtained for two conditions for each signal processing strategy: (a) quiet and (b) diffuse R-Space™ restaurant noise. Conditions were randomly assigned to avoid order effects. No subject received the same sentence list twice, eliminating the potential for learning effects. Before HINT testing began, subjects were instructed to adjust their volume control to a comfortable level for a 65 dBA noise signal presented from a speaker at 0° at one meter.

Recording the R-Space™ Restaurant Noise

A known noisy restaurant (noise floor of 58 dBA at the recording position, but the level of the noise created by the assemblage of people was significantly higher), with carpeted floors, wooden walls, and a wooden cathedral ceiling, was secured for a private party. The dimensions of the room where the recording was made were 36 feet (length) x 36 feet (width) x 8.5 to 17.5 feet (height with a sloping roofline). Thus, the volume of the room was 22,000 cubic feet. The reverberation time was unknown but is probably of limited interest here, because the test materials (HINT sentences) were not spoken in the restaurant and therefore were not subject to any possible masking effects of reverberation. Finally, it was determined that the critical distance for the recording was about five feet. Some of the tables (those nearest the recording position) were partially at or within the critical distance of the recording microphones, but many of the tables were beyond. Therefore, the restaurant simulation was a combination of direct and diffuse elements (L. Revit, personal communication). About 45 people were seated and served breakfast in the main seating area of the restaurant, which, when completely full, could accommodate over 100 customers. A table at the center of the main seating area had been removed and replaced by an array of recording microphones. The eight main recording microphones were of the highly directional, "shotgun" (interference-tube) variety typically used in the movie-making industry to record sounds from a distance. Because each shotgun microphone had a frontal pick-up pattern spanning approximately 45° (±22.5°) around its axis, the eight microphones, when placed in an equally spaced, horizontal, circular array, picked up sounds arriving from all horizontal directions around the center of the array. The presumed pick-up points (diaphragms) of the shotgun microphones were located two feet from the center of the array. A ninth, omnidirectional microphone was placed at the center of the array for calibration purposes.

Each microphone was connected via a preamplifier to a separate track of a multitrack, digital audio tape (DAT) recorder (Tascam DTRS system). In this way, direct and reverberated sounds were captured (recorded) from around the restaurant "on their way" to the center of the two-foot-diameter microphone array. Later, using the R-Space™ playback system in the laboratory, these "captured" sounds were then released by the eight loudspeakers of the two-foot-diameter playback array. In this way, the sounds that had been captured at two feet from the center of the array in the restaurant would now complete their paths toward the central listening position, although now in a different time and place.

Calibration of the R-Space Restaurant Noise

Before the recording of the breakfast party, calibration signals were recorded individually through each microphone, so that playback levels could later be established to reflect the sound levels recorded in the restaurant. Separately for each shotgun microphone, an equalized loudspeaker (flat from 100 to 16,000 Hz in 1/3-octave bands, ±3 dB) was held at a distance of two feet in front of the diaphragm, along the center of the pickup axis of the microphone. A pink-noise signal was delivered to the loudspeaker and adjusted to produce 84 dB SPL at the center of the array. For each shotgun microphone, the individual, pink-noise calibration signal was recorded onto the corresponding tape channel. In subsequent playback, the gain of the amplifier for each R-Space loudspeaker was adjusted to produce 84 dB SPL at the center of the loudspeaker array, thus mirroring the calibration recording condition. On average, the sound pressure level of the breakfast party, as measured at the calibration point in the restaurant, was 75 dBC, or 72 dBA. Therefore, when properly calibrated, the playback system created corresponding average sound pressure levels.

The HINT materials (sentences) and the "R-Space restaurant noise" were transferred to a Macintosh hard drive using Toast 5.0 software, before being imported into AudioDesk software. Then, in AudioDesk, the right track was separated from the left track, and the two tracks were digitally spliced end-to-end to form one long "sound bite." This concatenated sound bite was then repeated as many times as was necessary to provide noise long enough for the longest presentation for the first HINT sentence.

For subsequent HINT lists, the same noise sound bite was used, but with the starting time differing from that of the previous list by several seconds. Offset times of several seconds were digitally edited and placed in the appropriate channels, thus producing uncorrelated noise. Compton-Conley et al (2004, figure 4, p. 447) recently reported that the long-term speech spectrum of the R-Space restaurant noise was very similar to the long-term speech spectrum of the HINT sentences and noise.

Figure 4 illustrates the signal presentation system consisting of eight Boston Acoustics CR-65 loudspeakers (dimensions: 257 mm x 162 mm x 200 mm; frequency response (±3 dB): 65–20,000 Hz; crossover frequency: 4200 Hz; woofer: 135 mm copolymer; tweeter: 20 mm dome; nominal impedance: 8 ohms) placed in an equally spaced array at ear level, one meter from the test subject in a 1.97 x 2.54 x 2.73 meter double-walled sound suite (volume = 14.05 m^3) with a reported reverberation time of 0.19 seconds (personal communication with Industrial Acoustics Company). The radius of the circle was one meter plus the depth of the loudspeaker (200 mm).

Prior to testing, two measurements were made using narrow bands of pink noise centered at 250, 500, 1000, 2000, and 4000 Hz from each of the eight loudspeakers. One measure was made at one meter and the second measure at a half a meter. As expected, the SPL measured at a half meter was 6 dB (±1 dB) greater than the SPL measured at one meter with the exception of 250 Hz for the loudspeakers at 45, 90, 270, and 317°. Thus, for the majority of loudspeakers and frequencies between 500 and 4000 Hz, the subject's head was within the critical distance in this test environment. Finally, signals (sentences and noise) were fed from a Macintosh-driven digital audio workstation, using MOTO AudioDesk software and a MOTU Model 828 eight-channel FireWire A/D-D/A converter. The 0° loudspeaker was driven by an Alesis Model RA-150 amplifier in bridge-mono mode. Individual channels of Carvin DC-150 amplifiers drove the remaining loudspeakers.

To ensure that the overall presentation level was 65 dBA for the noise condition, a .5 in microphone connected to a Quest 1900 precision sound level meter and OB-300 1/3-1/1 octave band (OB) filter was placed at ear

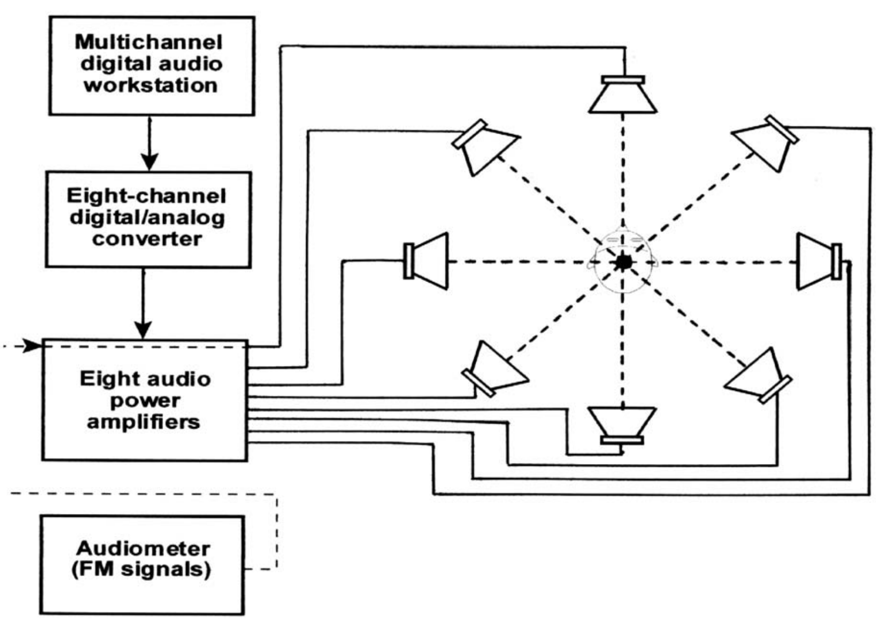

Figure 4. Illustration of the signal delivery and loudspeaker array used in the present study. For the diffuse condition, the noise was delivered from all eight loudspeakers, and the HINT sentences were delivered from the loudspeaker at 0° azimuth. For the quiet condition, the HINT sentences were delivered for the loudspeaker at 0° azimuth.

level, with the subject absent, one meter from the loudspeakers. Because the noise from each loudspeaker was uncorrelated to each other in the diffuse condition, the output level of each loudspeaker can be easily adjusted to yield the same overall output for each test-loudspeaker condition. Calibration of the loudspeakers was completed weekly, and the measured output was within ±1 dB of 65 dBA throughout the course of the study. For the noise condition, the overall output from each loudspeaker was 56 dBA (10 $\log_{10}[8]$ where 8 denotes the number of loudspeakers or 9 dB). Thus 65 dBA - 9 dBA = 56 dBA at each loudspeaker so when summed, the output from the eight loudspeakers at one meter was 65 dBA.

The purpose for using this continuous noise rather than the gated noise provided by the HINT recording was that the noise approximates more closely many real-world noisy situations. Finally, a lavaliere microphone was placed near the subject's mouth so the examiner could hear the subject's response to the HINT sentences.

Subjective Loudness Test

The *Subjective Loudness Test* is made up of 15 recorded environmental and speech stimuli derived from a Phonak sound compact disc (CD). Included in the CD were representative sounds of varying intensities and spectral properties. Sound samples included speech in quiet and background noise, music and various environmental sounds. Fifteen stimuli of varying intensity (50, 65, and 80 dBA) and spectral characteristics (low, mid, and high frequency) were chosen to be included in the questionnaire (see Appendix 3). Subjects were seated in the middle of a 1.97 x 2.54 x 2.73 meter double-walled sound suite with a loudspeaker at 0°, one meter away. After each sound sample was randomly presented, the subject was instructed to assign a loudness category and satisfaction category utilizing the *Environmental Sounds Questionnaire* rating scale described earlier. The evaluation of loudness comfort and satisfaction was completed only in the omnidirectional program.

Subjective Loudness Test Calibration

To assure that the presentation levels were correct, a .5 in microphone connected to a Quest 1900 precision sound level meter and OB-300 1/3-1/1 octave band filter was placed at ear level one meter from the loudspeaker. A 50, 65, and 80 dBA signal was played through the audiometer via an external CD track. The external output was adjusted on the audiometer until the desired signal level was read on the sound level meter. That position was then marked on the external output control on the audiometer. Calibration was completed weekly.

RESULTS

Main Effect of Signal Processing

Figures 5 and 6 illustrate the mean RTS (in dB) for signal processing (32-channel and 64-channel) and listening (R-Space™ restaurant noise; quiet) conditions. An RTS of 0 dB for the noise condition means the subject required the intensity level of the sentences to be equal to the level of the noise (65 dBA) in order to correctly repeat back 50% of the sentences. Thus, a higher RTS reflects poorer performance, and a lower RTS reflects better performance. For the quiet condition, a lower value represents the subject is able to repeat back the sentence at a lower intensity and, therefore, better performance.

A repeated randomized block ANOVA (Kirk, 1982) was performed on the data appearing in Table 2. The ANOVA reveals no significant main effects for signal processing

Table 2. Mean, Standard Deviation, Standard Error, Lower Boundary, and Upper Boundary for HINT Results in the Noise and Quiet Conditions for 32- and 64-Channel Signal Processing

	NOISE		QUIET	
	32 CHANNEL (dB SNR)	64 CHANNEL (dB SNR)	32 CHANNEL (dB RTS)	64 CHANNEL (dB RTS)
MEAN	3.6	3.5	51.2	50.4
SD	2.8	2.7	6.6	6.4
SE	0.9	0.9	2.1	2.0
LB	1.6	1.5	46.4	45.8
UB	5.6	5.4	56.0	55.0

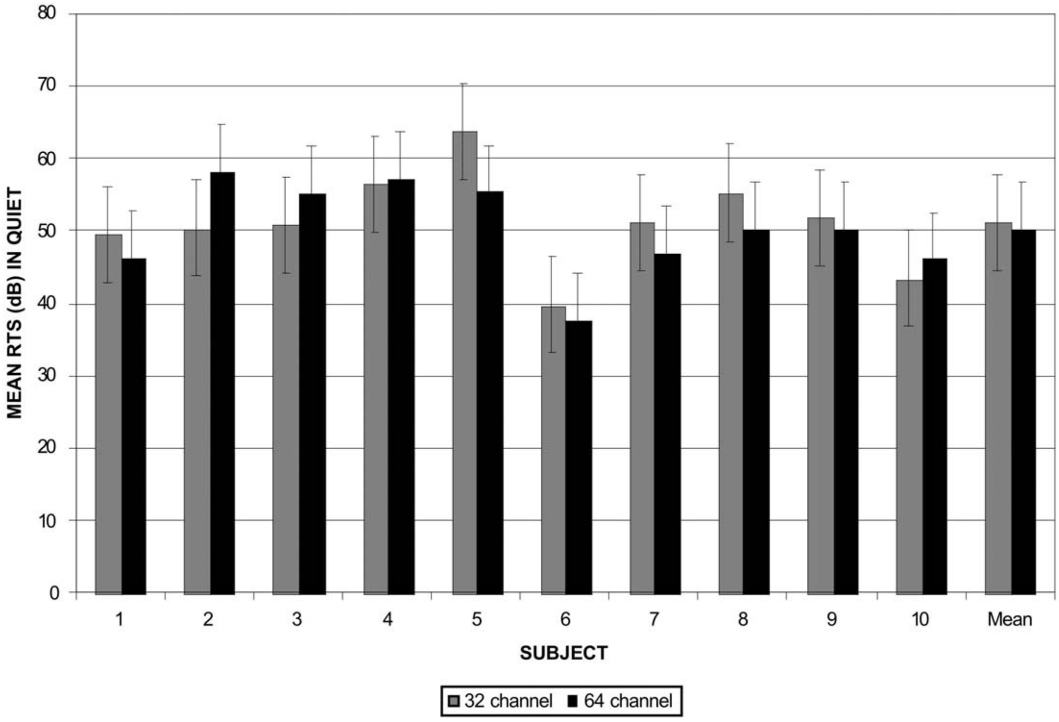

Figure 5. Mean reception threshold for sentences (RTS in dB) in quiet for 32- and 64-channel signal processing strategies in two listening conditions. Error bars represent ±1 SD.

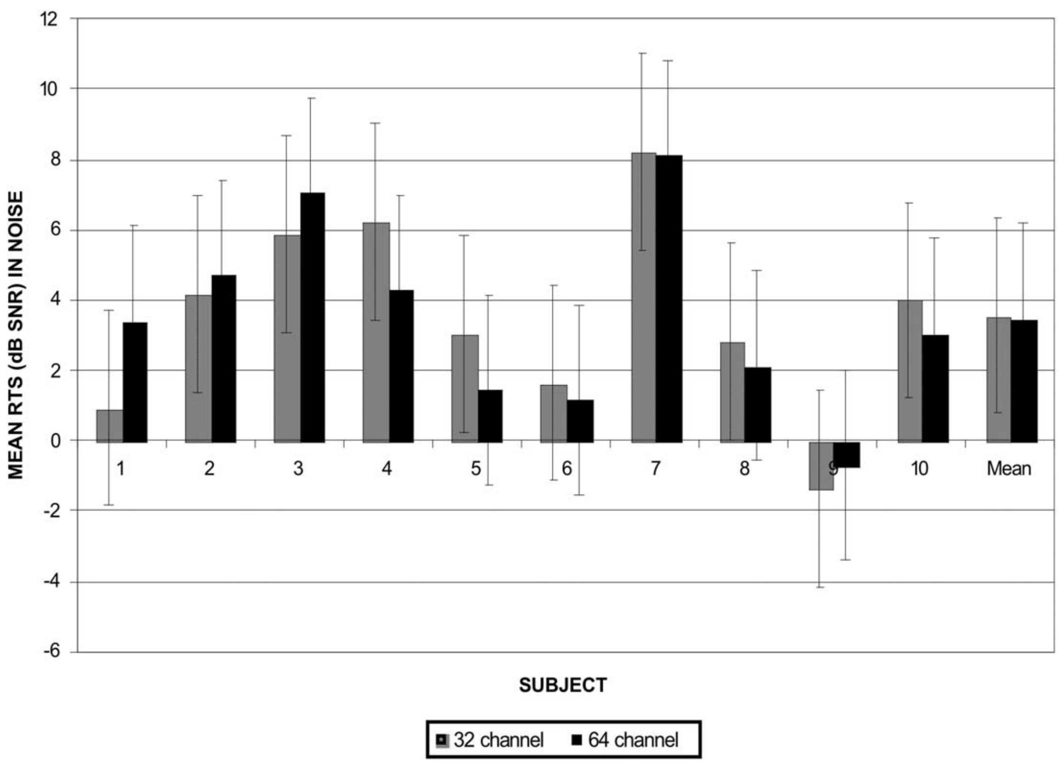

Figure 6. Mean reception threshold for sentences (RTS in dB) in noise for 32- and 64-channel signal processing strategies in two listening conditions. Error bars represent ±1 SD.

in noise (F = .051; d,f = 1,9; $p < 0.83$) or quiet (F = .288; d,f = 1,9; $p < 0.60$).

Figure 6 illustrates that the mean performance with 32-channel signal processing (3.6 dB) is not significantly different than the mean performance for 64-channel signal processing (3.5 dB) when listening in diffuse noise. Figure 5 illustrates that the mean performance with 32-channel signal processing (51.2 dBA) is also not significantly different than the mean performance for 64-channel signal processing (50.4 dBA) when listening in quiet.

Subjective Loudness Test

Figure 7 illustrates the mean loudness rating (±1 SD) for the 32-channel and 64-channel signal processing strategies for the 15 sound samples. Figure 8 illustrates the mean *satisfaction* rating (±1 SD) for each signal processing strategy.

Of the 15 sound samples, 14 sounds reveal no significant difference in *loudness* ratings between channel conditions. Only one sound sample demonstrates a significant difference in mean *loudness* rating based on a repeated randomized block ANOVA for 32-channel versus 64-channel signal processing. An ANOVA performed on the data appearing in Figure 7 illustrates the mean loudness rating for the party noise (#10) is significantly louder for 32-channel signal processing (5.8) than the mean loudness rating for 64-channel signal processing (4.7) (F = 5.8; d,f = 1,9; $p < .04$). Although a significant result is reported, this finding needs to be viewed with caution due to the small effect size (1.1). The computed observed power is .57 based upon a computed alpha of .05 indicating that the sample size may not be sufficient for the reported size effect.

An ANOVA performed on the data appearing in Figure 8 illustrates 14 of the 15 sound samples also show no significant difference in *satisfaction* ratings. Only the mean *satisfaction* rating for the flute (#5) sound sample for the 32-channel signal processing condition is significantly poorer than the mean satisfaction rating for the 64-channel signal processing (4.2) (F = 7.6; d,f = 1,9; $p < .02$). Again, although a significant result is reported, this finding should be viewed with caution due to the small effect

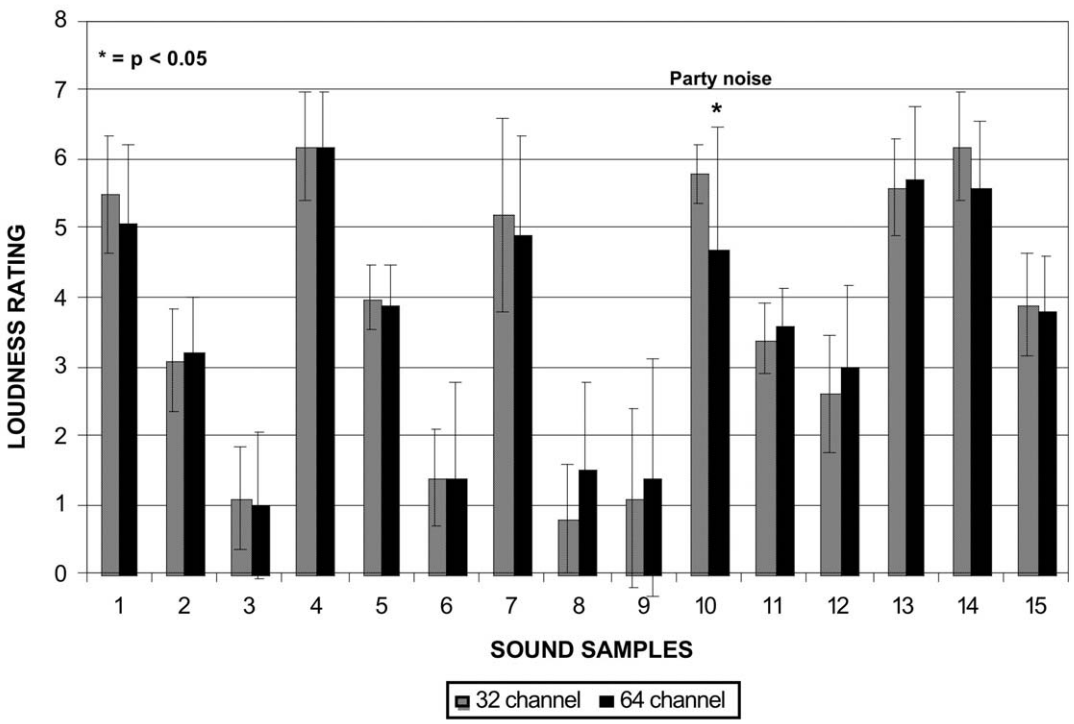

Figure 7. Mean loudness ratings for 32- and 64-channel signal processing strategies for the *Subjective Loudness Test*. Error bars represent ±1 SD.

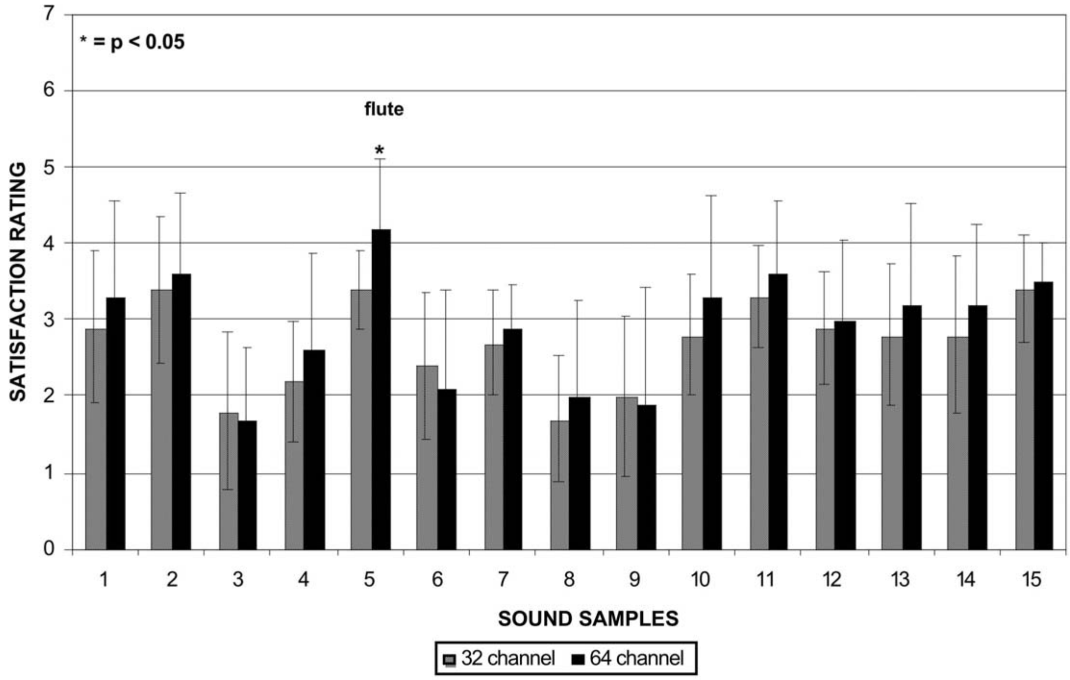

Figure 8. Mean satisfaction ratings for 32- and 64-channel signal processing strategies for the *Subjective Loudness Test*. Error bars represent ±1 SD.

size (0.8). The computed observed power is .69 based on a computed alpha of .05.

Environmental Sounds Questionnaire

Figure 9 illustrates the mean *loudness* rating (±1 SD) for the 18 environmental sounds for the unaided and the 32- and 64-channel aided conditions. Because the *Environmental Sounds Questionnaire* was completed two times, after wearing the hearing aids in each of the two processing conditions, unaided data was collected twice. The results of the two unaided conditions are averaged in Figures 9 and 10 because no significant differences were found between the two unaided conditions. Figure 10 illustrates the same results for the *satisfaction* rating. In Figures 9 and 10, if the ANOVA performed on each environmental sound between the unaided and the two aided conditions is significant, either a ‡ ($p \leq .05$) or ‡‡ ($p \leq .01$) symbol is placed where appropriate. Further, if a significant difference is found between the two aided conditions, then an * ($p \leq .05$) is placed where appropriate.

An ANOVA performed on each sound sample revealed the mean *loudness* rating for 17 of the 18 sound samples is significantly softer in the unaided condition in comparison to the aided 32-channel processing condition. Only sound sample #9, chewing soft food ($F = 4.6$; d,f = 1,7; $p < .057$), does not reveal statistical significance between aided and unaided conditions. In addition, the results of the ANOVA on the data appearing in Figure 8 illustrate that the mean loudness rating for the unaided condition is significantly softer than the aided 64-channel condition for 17 of the 18 sound samples. Only sound sample #11, water boiling ($F = 8.0$; d,f = 1,6; $p < .104$), does not show a significant difference between unaided and aided conditions.

Results of the ANOVA also show the loudness rating on 17 of the 18 sound samples is not significantly different between aided conditions. Only one sound sample, #8 (motorbike passing by), demonstrates significant differences in loudness ratings between the 32-channel and 64-channel conditions. The mean loudness rating for the 32-channel processing (6.2) is significantly louder than the mean loudness rating for the 64-channel processing (5.5) ($F = 13.5$; d,f = 1,9; $p < .025$).

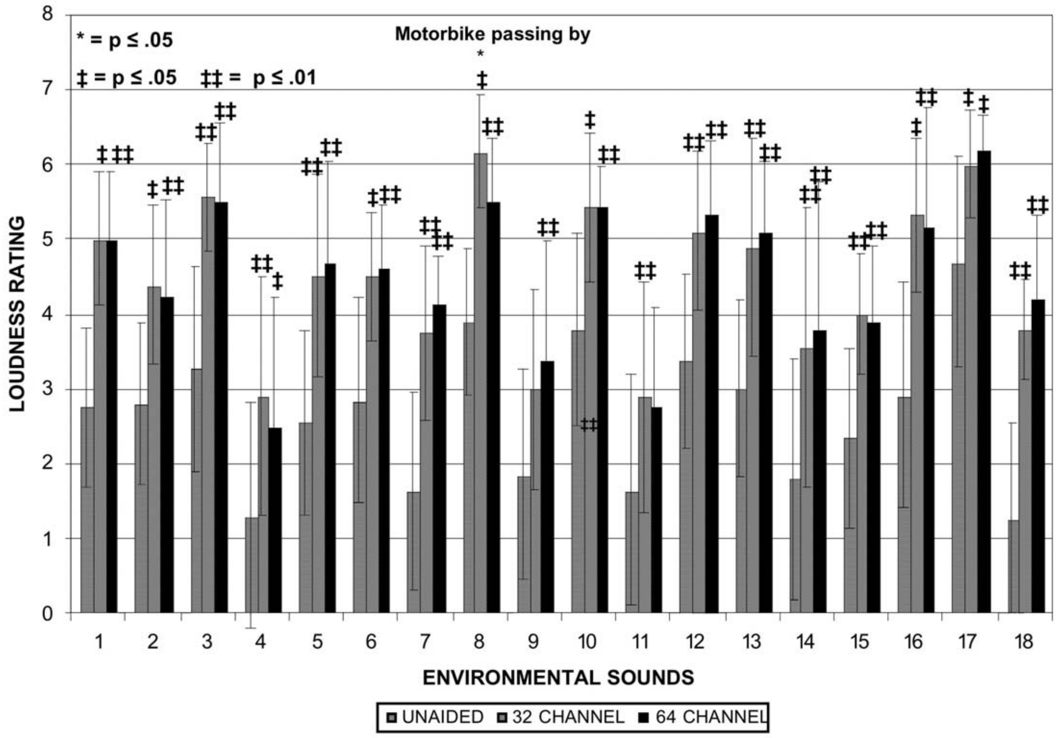

Figure 9. Mean loudness ratings for the 18 environmental sounds unaided, 32- and 64-channel signal processing strategies for the *Environmental Sounds Questionnaire*. Error bars represent ±1 SD.

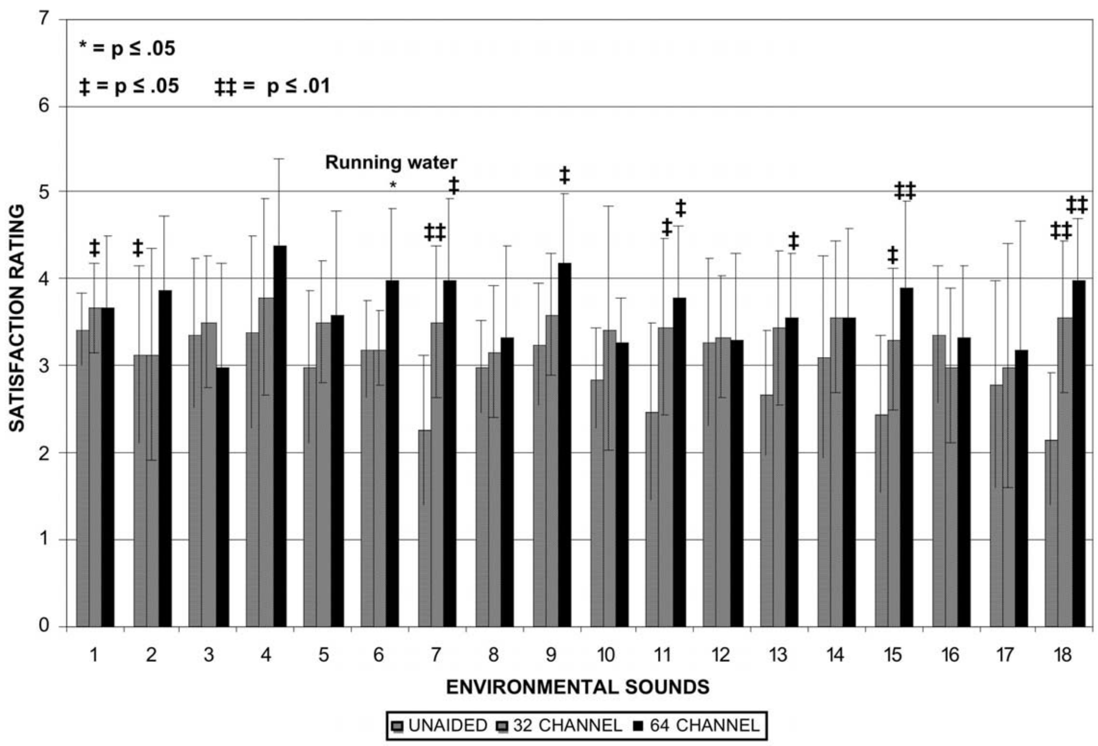

Figure 10. Mean satisfaction ratings for unaided, 32- and 64-channel signal processing strategies for the *Environmental Sounds Questionnaire*. Error bars represent ±1 SD.

An ANOVA performed on each sound sample appearing in Figure 10 reveals the mean *satisfaction* rating for the unaided condition is significantly poorer than the aided 32-channel processing aided condition for 6 sound samples: #1, dog barking close by ($F = 7.4$; d,f = 1,3; $p < .025$); #7, car turn signal ($F = 42.4$; d,f = 1,3; $p < .002$); #11, water boiling ($F = 3.9$; d,f = 1,3; $p < .035$); #15, microwave oven beeping ($F = 5.1$; d,f = 1,3; $p = .023$); and #18, birds twittering ($F = 5.4$; d,f = 1,3; $p < .007$). Results of the ANOVA also reveal the mean satisfaction ratings for the unaided condition are significantly poorer than the aided 64-channel processing condition for 6 sound samples: #7, car turn signal ($F = 42.4$; d,f = 1,3; $p < .036$); #11, water boiling ($F = 3.9$; d,f = 1,3; $p < .016$); #13, telephone ringing ($F = 3.9$; d,f = 1,3; $p < .035$); #15, microwave oven beeping ($F = 5.1$; d,f = 1,3; $p < .006$); and #18, birds twittering ($F = 5.4$; d,f = 1,3; $p < .006$).

Finally, ANOVA results on the data in Figure 10 indicate the mean *satisfaction* rating for 17 of the 18 sound samples is not significantly different between aided channel conditions. Only for sound sample #6, running water, is the mean satisfaction rating for the 32-channel processing condition (3.2) significantly poorer than the 64-channel processing condition (4.0) ($F = 3.3$; d,f = 1,3; $p < .022$).

The *Listening Tasks Questionnaire*

The *Listening Tasks Questionnaire* was used to subjectively assess the subject's perception of his or her own voice quality. For the 32-channel condition, eight of the ten subjects report his or her voice to be "comfortable" at one week postfitting. For the 64-channel condition, seven of the ten total subjects report the sound of his or her own voice to be "comfortable" one week following the fitting. Of the remaining subjects, the reports of own voice quality included the descriptors "tinny" and "hollow." Yet, increasing the number of channels does not seem to play a part in contributing to the negative sound quality of the subject's own voice because those subjects who rated his or her voice to be other than "comfortable" in one processing strategy also reported undesirable descriptors of their voice in the alternate signal processing rationale as well.

DISCUSSION

Past research has questioned whether the advantages of multichannel signal processing can be realized without users experiencing the possible negative side effects (i.e., increased loudness discomfort, decreased speech recognition, and poor sound quality of subject's own voice) of increasing the number of processing channels. The current study examined the effect of increasing the number of signal processing channels from 32 to 64 channels on loudness comfort and satisfaction, sentence recognition, and the sound quality of the subjects' own voice in ADRO signal processing.

As was previously discussed in the introduction, loudness discomfort as a result of channel summation has been reported to be a possible negative side effect when increasing the number of processing channels. The results reported from the *Subjective Loudness Test* in the current study did not demonstrate any significant differences between the 32- and 64-channel processing in the loudness rating for 14 of the 15 sound samples. The mean loudness rating for only party noise was significantly different between channel conditions. For this sound sample, the party noise in the 32-channel condition was reported on average to be significantly louder than in the 64-channel condition. This finding was contrary to what might be expected if channel summation is occurring. However, the overall mean satisfaction rating for the party noise sound sample was not significantly different between channel conditions. This indicates that even though the mean loudness of the party noise might have been perceived to be louder with 32-channel signal processing in comparison to 64-channel signal processing, subjects overall were equally satisfied with the loudness of the sound of the party noise in both conditions. Therefore, it seemed that the increase in loudness did not lead to a decrease in overall subject satisfaction or loudness discomfort for this sound sample. Results of the ANOVA on the *Environmental Sounds Questionnaire* also demonstrated that the loudness ratings for 17 of the 18 sounds were not significantly different. Only one of the 18 sound samples (motorbike passing by) showed any significant difference in loudness between the two channel conditions, and again, the satisfaction rating between the 32-channel and 64-channel conditions was not significantly different, suggesting no difference in loudness discomfort. Additionally, this significant difference in loudness and satisfaction ratings must be viewed with caution due to the small effect size between the two processing conditions as a result of the small number of subjects.

It must also be noted that the lack of reported differences in loudness ratings on the *Subjective Loudness Test* and the *Environmental Sounds Questionnaire* between the two channel conditions may be a result of the fitting method used in the current study. Recall that the study hearing aids were fit using an in-situ loudness judgment method in which loudness comfort values were gathered across discrete frequencies for each channel condition. By utilizing the subject's individual dynamic range as the target for the fitting, the probability of the subject experiencing loudness discomfort was decreased as compared to using predicted values as determined by a prescriptive fitting target. Therefore, any effects of channel summation may have been accounted for at the time of the fitting.

A second objective of this study was to determine the effect of an increased number of processing channels on speech recognition. A significant amount of research evaluating the effect of multichannel processing on speech recognition has been performed, utilizing different types of speech stimuli in a variety of testing conditions. Results of the HINT in the current study demonstrate no significant differences in sentence recognition in quiet or diffuse noise between the 32- and 64-signal processing channel conditions. This was in agreement with past research using HINT sentences measuring the effect of multichannel processing. Moore et al (1999) reported no significant differences in HINT performance when increasing the number of signal processing channels between one, two, four, and eight signal processing channels. Yund and Buckles (1995) found when the number of signal processing channels was greater than eight channels, increasing the number of processing channels had no effect on subjects' speech recognition abilities. Even though the Moore et al (1999) and Yund and Buckles (1995) studies, as well as the current study, utilized subjects with different levels of hearing aid experience and different testing environments, the results were still comparable. This may be due to the similarities in subject inclusion criteria. All of these studies included users with mild-to-moderate hearing losses. It is unknown if subjects' performance would be more affected by increasing the number of processing channels if subjects with more severe hearing losses who may rely more significantly on the temporal cues in speech were utilized in the studies.

Crain and Yund (1995) published results that demonstrated that even as the number of processing channels continues to increase to thirty-one channels, negative effects on speech recognition were still not found. Despite the fact that Crain and Yund (1995) evaluated vowel and consonant stimuli and the current study uses HINT sentences, the results were still in agreement that increasing the number of processing channels did not cause speech recognition to decrease even when the number of processing channels exceeded 30 channels.

Finally, the current study also examined the effect of longer group delay on the sound quality of a subject's own voice as a result of increasing the number of processing channels. The mean group delay for the 32-channel and 64-channel signal processing conditions was reported to be approximately 7 and 13 msec, respectively, in the current study. These values did not exceed the 15 msec upper limit for group delay for hearing-impaired listeners as previously reported by Stone and Moore (2005). Additionally, the results of this study indicate that increasing the number of ADRO processing channels did not change the subjects' subjective reports of own voice quality. Those subjects who reported the sound quality of their own voice to be undesirable (i.e., tinny, hollow, etc.) with one processing strategy did with the other processing strategy as well.

These reports of a change in the sound quality of a subject's own voice also agreed with data published by Stone and Moore (1999), who state that even delays shorter than 10 msec could result in the subtle change of the timbre of the subject's own voice as was seen with some of the subjects in this study. Additionally, it was reported that subjects who had a low-frequency pure-tone (PTAlf) average between 30–39 dB HL reported significantly higher disturbance than the other four groups with lesser or greater degrees of hearing loss. In the current study, the mean PTAlf of the subjects was 38.6, which would have included the subjects in the significantly more disturbed group in the Stone and Moore (1999) study.

Consideration must also be given to the finding that increasing the number of processing channels from 32- to 64-signal processing channels did not seem to yield any significant advantages in the current study as well. As was reported, previous data (Blamey et al, 2004) has shown that increasing the number of processing channels could be beneficial. These advantages were not realized in terms of improved loudness comfort and satisfaction, improved sentence recognition in quiet and noise, or improved sound quality of subjects' own voice in this study. Therefore, it seems subjects' performance with the 32-channel signal processing strategy was equivalent to performance with the 64-channel signal processing in this study, and the additional channels did not provide significant improvement.

It is worth noting that the results found in this study are exclusive to the ADRO™ signal processing. If another form of processing (i.e., wide dynamic range compression, linear with output limiting, etc.) would have been utilized, the results of increasing the number of processing channels is unknown and may not have been similar to what was published here. Additionally, if an alternate fitting method were utilized instead of the manufacturer's in-situ fitting procedure, the results might have varied as well. Additional research needs to be completed to determine if alternative forms of signal processing or fitting methods would have yielded different results between the 32- and 64-channel processing conditions. It could also be beneficial if an experimental aid with fewer channels (i.e., one, four, or eight channels) could be included in the future studies for a baseline comparison.

In conclusion, this study evaluated the effects of an increased number of signal processing channels on loudness comfort and satisfaction, sentence recognition, and sound quality of subject's own voice utilizing the ADRO processing in 32-channel and 64-channel signal processing strategies in ten subjects. The results of the study revealed:

1. No significant differences in loudness comfort were present between 32- and 64-channel processing strategies for 14 of the 15 sound samples as measured by the *Subjective Loudness Test*. Additionally, mean loudness ratings for 17 of the 18 sounds on the *Environmental Sounds Questionnaire* revealed no significant differences between 32-channel and 64-channel conditions.
2. No significant differences in loudness satisfaction were present between 32- and 64-channel processing strategies for 14 of the 15 sound samples as measured by the *Subjective Loudness Test*. Additionally, mean satisfaction ratings for 17 of the 18 sounds on the *Environmental Sounds Questionnaire* revealed no significant differences between 32-channel and 64-channel conditions.
3. No significant differences were present between 32- and 64-channel processing strategies in an adaptive directional microphone mode for the reception threshold for sentences (RTS in dB) on the *Hearing In Noise Test* (HINT) sentences presented at 0° and diffuse R-Space™ noise (eight loudspeaker array) fixed at 65 dBA.
4. No significant differences were present between 32- and 64-channel processing strategies in the adaptive directional microphone mode for reception threshold for sentences (RTS in dB) on the *Hearing In Noise Test* (HINT) sentences presented at 0° in quiet.
5. Subjective differences in sound quality of subject's own voice quality are present between the 32- and 64-channel processing strategies as measured by the *Listening Tasks Questionnaire*.

Acknowledgments. The authors would like to thank Dynamic Hearing Pty for providing a grant to cover the direct costs involved for completing the project, and Interton for providing the hearing aids used in the study. Additionally, the authors would especially like to thank Todd Fortune at Interton and Peter Blamey at Dynamic Hearing for their support and assistance throughout the project. Finally, the authors would like to thank Michael Strube, Ph.D., Professor of Psychology at Washington University, for his assistance in completing the statistical analysis and

appropriate interpretation of the data reported in this manuscript.

REFERENCES

Agnew J, Thornton J. (2000) Just noticeable and objectionable group delays in digital hearing aids. *J Am Acad Audiol* 11:330–336.

American National Standards Institute. (1996) *American National Standard for Specification of Audiometers.* (ANSI S3.6-1996). New York: Acoustical Society of America.

Blamey P, Martin LFA, Fiket HJ. (2004) A digital processing strategy to optimize hearing aid outputs directly. *J Am Acad Audiol* 15:716–728.

Blamey P. (2005) Adaptive dynamic range optimization: a digital amplification strategy for hearing aids and cochlear implants. *Trends Amp* 9(2):T1–T22.

Compton-Conley C, Neuman A, Killion M, Levitt H. (2004) Performance of directional microphones for hearing aids: real world versus simulation. *J Am Acad Audiol* 15:440–455.

Crain TR, Yund EW. (1995) The effect of multichannel compression on vowel and stop-consonant discrimination in normal-hearing and hearing-impaired subjects. *Ear Hear* 16(5):529–543.

Dillon H. (2001). *Hearing Aids.* New York: Thieme Medical Publishers.

Frye G. (2001) Testing digital and analog hearing instruments: processing time delays and phase measurements. *Hear Rev* 10(8):36–42.

Kiessling J, Steffens T. (1991) Clinical evaluation of a programmable three-channel automatic gain control amplification system. *Audiology* 30(2):70–81.

Kirk RE. (1982) *Experimental Design.* 2nd edition. Pacific Grove, CA: Brooks/Cole Publishing Company.

Kuk F. (2002) Considerations in modern multichannel nonlinear hearing aids. Valente M, ed. *Hearing Aids: Standards, Options, and Limitations*. New York: Thieme Medical Publishers, 178–213.

Kuk F, Ludvigsen C. (2003) Changing with the times: choice of stimuli for hearing aid verification: puretones, speech or composite signals? Here's what to use and why. *Hear Rev* 10(9):24–28, 56–57.

Martin LFA, Blamey PJ, James CJ, Galvin KL, Macfarlane D. (2001) Adaptive dynamic range optimisation for hearing aids. *Acoust Aust* 29(1):21–24.

Moore B, Glasberg B. (1986) A comparison of two-channel and single-channel compression hearing aids. *Audiology* 25(4):210–226.

Moore B, Peters RW, Stone MA. (1999) Benefits of linear amplification and multichannel compression for speech comprehension in backgrounds with spectral and temporal dips. *J Acoust Soc Am* 105(1):400–411.

Nilsson M, Soli SD, Sullivan J. (1994) Development of the Hearing in Noise Test for the measurement of speech reception thresholds in quiet and in noise. *J Acoust Soc Am* 95:1085–1099.

Stone MA, Moore BCJ. (1999) Tolerable hearing aid delays: I. estimation of limits imposed by the auditory path alone using simulated hearing losses. *Ear Hear* 20:182–192.

Stone MA, Moore BCJ. (2002) Tolerable hearing-aid delays: II. estimation of limits imposed during speech production. *Ear Hear* 23(4):325–238.

Stone MA, Moore BCJ. (2005). Tolerable hearing-aid delays: IV. effects on subjective disturbance during speech production by hearing-impaired subjects. *Ear Hear* 26(2):225–234.

Summerfield Q. (1992) Lipreading and audiovisual speech perception. *Philos Trans R Soc Lond B Biol Sci* 335(1273):71–78.

Van Tassell D, Solis D, Kirby VM, Widen GP. (1987) Speech waveform envelope cues for consonant recognition. *J Acoust Soc Am* 82(4):1152–1161.

Yund EW, Buckles KM. (1995) Multichannel compression hearing aids: effect of number of channels on speech in noise. *J Acoust Soc Am* 97(2):1206–1223.

Appendix 1. Listening Tasks Questionnaire

For the next week we would like you to listen to a number of different situations with your new hearing aids. Your responses will help the audiologist to make any adjustments needed to help you hear better.

Unless otherwise specified, for all situations, set the hearing aid on Program 1 and the volume at a comfortable listening level.

Please check all that apply for each of the following questions.

ENVIRONMENTAL SOUNDS

Loud sounds

- When you hear a door slam, the sound is:
  - ❑ loud but ok
  - ❑ slightly too loud
  - ❑ uncomfortably loud
  - ❑ softer than expected
  - ❑ other ______________________________
- When you hear a spoon drop into the sink, the sound is:
  - ❑ loud but ok
  - ❑ slightly too loud
  - ❑ uncomfortably loud
  - ❑ softer than expected
  - ❑ other ______________________________
- When you hear traffic in the street, the sound is:
  - ❑ loud but ok
  - ❑ slightly too loud
  - ❑ uncomfortably loud
  - ❑ softer than expected
  - ❑ other ______________________________

Average sounds

- When you hear the doorbell, the sound is:
  - ❑ comfortable
  - ❑ too loud
  - ❑ slightly too soft
  - ❑ too soft
  - ❑ other ______________________________
- When you hear the phone ring, the sound is:
  - ❑ comfortable
  - ❑ too loud
  - ❑ slightly too soft
  - ❑ too soft
  - ❑ other ______________________________

Soft sounds

- When you hear the sound of paper rustling, the sound is:
  - ❑ comfortable
  - ❑ too loud
  - ❑ too soft
  - ❑ other ______________________________
- When you hear the sound of your own breathing, it is:
  - ❑ comfortable
  - ❑ too loud
  - ❑ too soft
  - ❑ other ______________________________

Feedback/whistling

- How often does your hearing aid whistle with the volume at your usual setting?
  - ❑ never
  - ❑ only when using the phone or wearing a hat
  - ❑ only when chewing or laughing
  - ❑ constantly
  - ❑ other ______________________

- Have you noticed any feedback with the volume full on?
  - ❑ never
  - ❑ only when using the phone or wearing a hat
  - ❑ only when chewing or laughing
  - ❑ constantly
  - ❑ other ______________________

YOUR OWN VOICE

Listening to your own voice

- When you are listening to your own voice, it sounds:
  - ❑ comfortable
  - ❑ slightly too loud
  - ❑ slightly too soft
  - ❑ hollow and echoing
  - ❑ tinny
  - ❑ other ______________________

PEOPLE TALKING

Listening to conversation in a quiet room

- When you are talking to one other person, speech sounds:
  - ❑ clear and comfortable
  - ❑ muffled or distorted
  - ❑ high pitched and tinny
  - ❑ hollow and dull
  - ❑ too soft
  - ❑ too loud
  - ❑ other ______________________

- When you are listening to the TV or radio, speech sounds:
  - ❑ clear and comfortable
  - ❑ muffled or distorted
  - ❑ high pitched and tinny
  - ❑ hollow and dull
  - ❑ too soft
  - ❑ too loud
  - ❑ other ______________________

NOISY PLACES

Listening to conversation in a noisy place

Set the hearing aid on Program 1 and the volume at a comfortable level and listen to the situations below. Then set the hearing aid on Program 2 and the volume at a comfortable level and listen again in the situations below.

- When several people are talking in a group, which program do you prefer to use?
  - ❑ I prefer to use Program 1
  - ❑ I prefer to use Program 2
  - ❑ Both programs work equally well

- When someone is talking and there is background noise, which program do you prefer to use?
  - ❑ I prefer to use Program 1
  - ❑ I prefer to use Program 2
  - ❑ Both programs work equally well

PROGRAMS

Changing programs

- Do you have any problems changing the programs?
  - ❑ No, I have tried using all programs
  - ❑ Yes, I find it difficult to change programs

- Can you hear the beep when the program changes?
  - ❑ Yes, I can hear a different number of beeps for each program
  - ❑ No, sometimes it is difficult to hear the beeps

VOLUME

Volume Control

- How many times a day do you manipulate the volume control?
  - ❑ never
  - ❑ 1–2 times
  - ❑ 3–4 times
  - ❑ 5 or more times

- In what situations do you manipulate the volume control?
  - ❑ ______________________________
  - ❑ ______________________________
  - ❑ ______________________________
  - ❑ ______________________________

TELEPHONE

Telecoil

- Have you tried using Program 3, the telecoil program, when speaking on the telephone?
  - ❑ never
  - ❑ 1–2 times
  - ❑ 3–4 times
  - ❑ 5 or more times

- When you are listening on the telephone through Program 3, the speech sounds:
  - ❑ clear and comfortable
  - ❑ too soft
  - ❑ too loud
  - ❑ other ______________________________

HOURS OF USE

- On average, how many hours per day did you use the experimental hearing aids in the last week?
  - ❑ never
  - ❑ 1–2 hours
  - ❑ 3–4 hours
  - ❑ 5–6 hours
  - ❑ 7–8 hours
  - ❑ 9–10 hours
  - ❑ 11–12 hours
  - ❑ more than 12 hours

- On average, what percentage of the time did you use Program 2 in the hearing aid?
  - ❑ almost never
  - ❑ 10 to 20%
  - ❑ 20 to 40%
  - ❑ 40 to 60%
  - ❑ 60 to 80%
  - ❑ 80 to 90%
  - ❑ almost always

Appendix 2. Environmental Sounds Questionnaire

Name: ______________________________	Date: _______________

Office use only: description of each hearing aid

HA1:	HA2:

PART 1

During this week, please listen to each of the following sounds with Program 1 of your hearing aid and without your hearing aid. Please enter your responses to indicate the loudness of the sound and your satisfaction with that loudness level for Program 1 of the hearing aid and without your hearing aid.

For rating the loudness of the sound, use the following loudness scale:
7 = uncomfortably loud
6 = loud but okay
5 = comfortable but slightly loud
4 = comfortable
3 = comfortable but slightly soft
2 = soft
1 = very soft
0 = do not hear
x = don't know, e.g., did not encounter that sound

For rating your satisfaction with the loudness level, use the following satisfaction scale:
5 = just right
4 = pretty good
3 = okay
2 = not too good
1 = not good at all

For example, you might rate a particular sound as "very soft." If "very soft" is your preferred level for this sound, then you would rate your loudness satisfaction as "just right." If, on the other hand, you think the sound should be louder than "very soft," then your loudness satisfaction rating might be "not too good" or "not good at all." The loudness satisfaction rating is not related to how pleasing or easy it is to hear the sound, but, rather, how satisfied you are with the loudness level perceived.

IMPORTANT: Remember to enter a loudness and satisfaction rating for Program 1 of the hearing aids and without your hearing aids. That means FOUR ratings for each sound described.

Loudness scale:
7 = uncomfortably loud
6 = loud but okay
5 = comfortable but slightly loud
4 = comfortable
3 = comfortable but slightly soft
2 = soft
1 = very soft
0 = do not hear
x = don't know, e.g., did not encounter that sound

Satisfaction scale:
- 5 = just right
- 4 = pretty good
- 3 = okay
- 2 = not too good
- 1 = not good at all

1. Dog barking close by.
   With hearing aid: Loudness Rating _______ Satisfaction Rating _______
   Without hearing aid: Loudness Rating _______ Satisfaction Rating _______

2. Traveling in a car with the windows closed.
   With hearing aid: Loudness Rating _______ Satisfaction Rating _______
   Without hearing aid: Loudness Rating _______ Satisfaction Rating _______

3. Traffic noise when standing on the curb of a busy road.
   With hearing aid: Loudness Rating _______ Satisfaction Rating _______
   Without hearing aid: Loudness Rating _______ Satisfaction Rating _______

4. Your own breathing.
   With hearing aid: Loudness Rating _______ Satisfaction Rating _______
   Without hearing aid: Loudness Rating _______ Satisfaction Rating _______

5. Washing machine.
   With hearing aid: Loudness Rating _______ Satisfaction Rating _______
   Without hearing aid: Loudness Rating _______ Satisfaction Rating _______

6. Running water, such as a toilet or shower.
   With hearing aid: Loudness Rating _______ Satisfaction Rating _______
   Without hearing aid: Loudness Rating _______ Satisfaction Rating _______

7. Car indicator signal.
   With hearing aid: Loudness Rating _______ Satisfaction Rating _______
   Without hearing aid: Loudness Rating _______ Satisfaction Rating _______

8. A motorbike passing by.
   With hearing aid: Loudness Rating _______ Satisfaction Rating _______
   Without hearing aid: Loudness Rating _______ Satisfaction Rating _______

9. Chewing soft food.
   With hearing aid: Loudness Rating _______ Satisfaction Rating _______
   Without hearing aid: Loudness Rating _______ Satisfaction Rating _______

10. Vacuum cleaner.
    With hearing aid: Loudness Rating _______ Satisfaction Rating _______
    Without hearing aid: Loudness Rating _______ Satisfaction Rating _______

11. Water boiling on the stove.
    With hearing aid: Loudness Rating _______ Satisfaction Rating _______
    Without hearing aid: Loudness Rating _______ Satisfaction Rating _______

12. Door slamming.
    With hearing aid: Loudness Rating _______ Satisfaction Rating _______
    Without hearing aid: Loudness Rating _______ Satisfaction Rating _______

13. Telephone ringing close by.
    With hearing aid: Loudness Rating _______ Satisfaction Rating _______
    Without hearing aid: Loudness Rating _______ Satisfaction Rating _______

14. Refrigerator motor.
    With hearing aid: Loudness Rating _______ Satisfaction Rating _______
    Without hearing aid: Loudness Rating _______ Satisfaction Rating _______

15. Microwave oven beeping.
    With hearing aid: Loudness Rating _______ Satisfaction Rating _______
    Without hearing aid: Loudness Rating _______ Satisfaction Rating _______

16. Hair dryer or electric shaver.
    With hearing aid: Loudness Rating _______ Satisfaction Rating _______
    Without hearing aid: Loudness Rating _______ Satisfaction Rating _______

17. Lawn mower.
    With hearing aid: Loudness Rating _______ Satisfaction Rating _______
    Without hearing aid: Loudness Rating _______ Satisfaction Rating _______

18. Birds twittering.
    With hearing aid: Loudness Rating _______ Satisfaction Rating _______
    Without hearing aid: Loudness Rating _______ Satisfaction Rating _______

PART 2

Please answer the following questions by entering information on the line or ticking the relevant box.

19. If you provided a low satisfaction rating (1 or 2) for some sounds in Part 1 of this questionnaire, please provide reasons for your dissatisfaction.

    ____

20. What types of sounds or listening situations do you normally find loud or noisy?

    ____

21. How often do you experience these loud or noisy sounds?
    ❒ Several times per day
    ❒ Several times per week
    ❒ Only occasionally

    A Question about the Multiprogram Hearing Aid

22. Have you found sounds or listening situations that were too noisy or uncomfortably loud?
    ❒ Yes ❒ No

    If yes, which program would you prefer to use under these circumstances?
    ❒ Program 1 ❒ Program 2 ❒ Program 3

23. For any of the programs, do you have any other comments to make about how loud or soft sounds in the environment were, or how you perceived loud and soft sounds?

Appendix 3. Subjective Loudness Test

	LOUD	MEDIUM	SOFT
LOW FREQUENCY	SHIP'S HORN LOUDNESS (0–7): SATISFACTION (0–5)	DOUBLE BASS LOUDNESS (0–7): SATISFACTION (0–5)	DISTANT THUNDER LOUDNESS (0–7): SATISFACTION (0–5)
MIDFREQUENCY	TRAFFIC NOISE LOUDNESS (0–7): SATISFACTION (0–5)	FLUTE LOUDNESS (0–7): SATISFACTION (0–5)	FLOWING WATER LOUDNESS (0–7): SATISFACTION (0–5)
HIGH FREQUENCY	WHISTLES LOUDNESS (0–7): SATISFACTION (0–5)	KEYS LOUDNESS (0–7): SATISFACTION (0–5)	BIRDS SING SOFT LOUDNESS (0–7): SATISFACTION (0–5)
BROADBAND	PARTY NOISE LOUDNESS (0–7): SATISFACTION (0–5)	DIALOG IN QUIET LOUDNESS (0–7): SATISFACTION (0–5)	FEMALE SPEECH IN QUIET LOUDNESS (0–7): SATISFACTION (0–5)
ADDITIONAL LOUD SOUNDS	PNEUMATIC HAMMER LOUDNESS (0–7): SATISFACTION (0–5)	LOUD MUSIC LOUDNESS (0–7): SATISFACTION (0–5)	PAPER RUSTLING LOUDNESS (0–7): SATISFACTION (0–5)