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# Comparative Performance of an Adaptive Directional Microphone System and a Multichannel Noise Reduction System

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**Abstract**

The amplification outcomes of two hearing aid prescriptions, NAL-NL1 and Digital Perception Processing (DPP), of nine moderate to moderately severe hearing-impaired adults were compared in the same digital hearing instrument. NAL-NL1 aims at optimizing speech intelligibility while amplifying the speech signal to a normal overall loudness level (Dillon, 1999). DPP focuses on restoring loudness based on normal and impaired cochlear excitation models (Launer and Moore, 2003). In this comparison, DPP resulted in better sentence recognition performance than the NAL-NL1 algorithm in the signal-front/noise-side condition, and the two prescriptions gave similar performance in the signal-front/noise-front condition. Subjective evaluations by the participants using the Abbreviated Profile for Hearing Aid Benefit and sound quality comparisons did not give conclusive results between the two prescriptions.

With each hearing aid prescription, the ability of the hearing aid circuitry to reduce the effects of noise was evaluated by a sentence-in-noise test in three conditions: (1) adaptive directional microphone (DAZ), (2) multichannel noise reduction system (FNC), and (3) a combination of FNC and DAZ (FNC + DAZ). In the signal-front/noise-side condition, DAZ and FNC + DAZ gave better performance than FNC in nearly all participants, whereas in the signal-front and noise-front evaluation, the conditions revealed no significant differences.

**Key Words:** Adaptive directional microphone, digital perception processing, hearing aid prescription, multichannel noise reduction, NAL-NL1

**Abbreviations:** APHAB = Abbreviated Profile of Hearing Aid Benefit; CHINT = Cantonese Hearing-in-Noise Test; DAZ = Digital Audio Zoom; DPP = Digital Perception Processing; FNC = Fine Noise Canceller; NAL-NL1 = National Acoustic Laboratories Non-linear 1 hearing aid prescription formula; RTS = reception threshold of speech; SNR = signal-to-noise ratio

**Sumario**

Los resultados de dos sistemas de prescripción de auxiliares auditivos, el NAL-NL1 y el Procesamiento Digital de Percepción (DPP), se compararon en el mismo dispositivo auditivo digital, en nueve adultos con pérdidas auditivas de grado moderado a moderadamente severo. El NAL-NL1 busca optimizar la inteligibilidad del lenguaje amplificando la señal de lenguaje a un nivel global normal de intensidad subjetiva (Dillon, 1999). El DPP se concentra en la restauración de la intensidad subjetiva con base en modelos de excitación en cócleas normales o alteradas (Launer y Moore, 2003). En esta comparación,

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el DPP produjo un mejor desempeño en el reconocimiento de frases que el algoritmo NAL-NL1, en la condición de señal al frente y ruido al lado, y las dos fórmulas de prescripción rindieron similarmente en la condición de señal al frente y ruido al frente. Las evaluaciones subjetivas de los participantes utilizando el Perfil Abreviado de Beneficio del Auxiliar Auditivo y las comparaciones de calidad de sonido no produjeron resultados concluyentes entre las dos prescripciones.

Con cada fórmula de prescripción para auxiliares auditivos, la capacidad de los circuitos del dispositivo de reducir los efectos del ruido fue evaluada por una prueba de frases en ruido bajo tres condiciones: (1) micrófono adaptativo direccional (DAZ); (2) sistema multicanal de reducción de ruido (FNC), y (3) una combinación de FNC y DAZ (FNC + DAZ). En la condición señal al frente-ruido al lado, DAZ y FNC + DAZ mostraron mejor desempeño que el FNC en casi todos los participantes, mientras que en la evaluación señal al frente y ruido al frente, las condiciones no revelaron diferencias significativas.

**Palabras Clave:** Micrófono direccional adaptativo, procesamiento digital de la percepción, prescripción para auxiliares auditivos, reducción multicanal del ruido, NAL-NL1

**Abreviaturas:** APHAB = Perfil Abreviado de Beneficio del Auxiliar Auditivo; CHINT = Prueba Cantonesa de Audición en Ruido; DAZ = Zoom Digital de Audio; DPP = Procesamiento Digital de la Percepción; FNC = Cancelador de Ruido Fino; NAL-NL1 = Fórmula de prescripción de auxiliares auditivos No-lineal 1 de los Laboratorios Nacionales de Acústica; RTS = umbral de recepción del lenguaje; SNR = tasa señal/ruido

The most difficult amplification challenge for the hearing-impaired population, even for the milder hearing losses, is the reduction of speech recognition ability when listening in background noise or competing speech. With appropriate prescription and fitting, a hearing aid can significantly improve a hearing-impaired individual's speech recognition in a quiet and in nonreverberant listening environments. This benefit, however, is greatly reduced in the presence of noise, especially for individuals with higher degrees of hearing loss (Killion and Niquette, 2000). The most effective way to improve speech recognition in noise is to improve the signal-to-noise ratio (SNR). Hearing aid manufacturers have implemented technologies, such as noise reduction systems and directional microphone systems, in their products that aim to improve the SNR of the incoming signal.

## Multichannel Noise Reduction Systems

There are reviews in the literature on the strategies used in various noise reduction methods in hearing aids (Dillon and Lovegrove, 1993; Levitt, 1993; Weiss and Neumann, 1993). Some of these strategies involved the use of complex signal processing algorithms such as spectral subtraction (Boll, 1979), adaptive noise cancellation (Weiss, 1987), adaptive beamforming (Peterson et al, 1987; Kompis and Dillier, 1994), and spectral enhancement (Baer et al, 1993). Research has shown, however, that these strategies are either unable to improve speech recognition in noise or in some cases worsen capability as these speech processing algorithms tend to remove or degrade the speech signal along with portions of the noise spectrum.

The most commonly used noise reduction system in current commercial multichannel digital hearing aids is based on the work by van Dijkhuizen et al (1991). The incoming signal is analyzed into different frequency channels with their respective SNRs estimated. The speech signal has amplitude modulation rates in the range of 2–30 Hz, whereas background noise has modulation rates higher than this range. The noise reduction system assumes a high SNR if the modulation pattern for a channel resembles speech, whereas a low SNR is assigned if the channel has shallow modulation depth or high modulation rate. The overall gain level is reduced in those channels with an SNR lower than a predetermined level.

For a hearing aid with only a few

frequency channels, its effectiveness to improve SNR is limited because the gain is reduced over a larger frequency range of the broadband speech signal even if the noise has a narrow bandwidth. Conversely, if there are more frequency channels in the hearing aid each channel is assigned a narrower bandwidth, and gain reduction will therefore be applied only to the channels with a poor SNR. The overall SNR may improve with gain reduction in channels with the strongest background noise levels. The benefits may be particularly prominent in situations where noise energy has a narrower bandwidth than the speech signal (Lurquin et al, 2001). Lurquin et al reported that speech recognition in noise was better when such multichannel noise reduction systems were activated. In their study, the participants achieved an average 3–4 dB lower SNR for 50% correct score in the "moderate" multichannel cancellation setting than in the "off" setting.

Alcantara et al (2003) evaluated the same multichannel noise reduction system used in the Lurquin et al (2001) study, and they hypothesized that the noise reduction system would function well for noise that is steady, nonmodulated, and had reduced energy in some frequency regions. They further speculated that overall gain would be reduced in channels dominated by noise, and speech gain would be maintained in channels dominated by speech with low noise levels. Therefore, the noise would have less masking effect on the speech signal, which might give better speech recognition. They felt that a multichannel noise reduction system would indeed work less effectively in the presence of modulated wideband noise without energy reduction in any frequency regions because estimating the SNR would be difficult due to the presence of modulation. Therefore, gain would be reduced across a larger frequency region due to the wideband noise. Surprisingly, their results did not support their hypotheses. The speech reception thresholds in noise revealed no differences between conditions with or without the multichannel noise reduction system activated, regardless of the type of background noise used. Because of the limitations of multichannel noise reduction systems on speech recognition in noise, other strategies are being evaluated on their abilities to combat noise, particularly directional microphones.

### Directional Microphone Systems

Another technology used to improve the SNR is through the use of directional microphone technology. Directional microphones have been proven effective in manipulating the SNR since the early 1980s (Madison and Hawkins, 1983; Mueller et al, 1983; Hawkins and Yacullo, 1984). A single directional microphone design consists of one microphone with two sound inlets (front and rear) that lead to separate cavities divided by a diaphragm. Sound is manipulated to ensure that it reaches the microphone diaphragm at the same time from both inlets, such as through the use of an acoustical time delay, with the effect of cancellation of the signal (Ricketts and Dittberner, 2002). Single directional microphones usually have a cardioid directivity pattern which can provide up to 3–4 dB enhancement of the SNR in a nonreverberant test environment (Nielsen and Ludvigsen, 1978; Hawkins and Yacullo, 1984). However, a single directional microphone fails to offer an omnidirectional option, which is more favorable in situations such as listening to speech in a diffuse noise source or listening to music (Kuehnel and Checkley, 2000). This problem was solved with the introduction of a dual-microphone directional design. This design uses two separate, matched omnidirectional microphones that allow the user to switch manually or electronically between the omnidirectional and directional modes. In the directional mode, an electronic time delay is introduced to the back microphone (Ricketts and Dittberner, 2002). The effectiveness of dual-microphone systems has been well documented by many research studies (Agnew and Black, 1997; Voss, 1997; Gravel et al, 1999; Pumford et al, 2000). Even with the demonstrated capability of the dual-microphone design in controlled studies, the use of directional microphones has limitations in everyday listening environments. In conditions where there are multiple and diffuse noise sources, a fixed dual-microphone design will not be as effective in attenuating the noise because of the fixed directivity pattern, which may not always match the direction of the noise source (Kuehnel and Checkley, 2000). Dual-microphone systems have been introduced, which provide an adaptive directivity pattern in response to the environment such that the null is directed toward the principal noise source. In adaptive directional microphones, the directivity pattern is continually adjusted to keep the output level of the system at a minimum in the direction of the noise source. Once the adaptive directional option in the dual-microphone system is activated, the hearing aid user does not have control over the actual selection of a specific directivity pattern. Studies have been undertaken to evaluate the effectiveness of adaptive dual-microphone systems. Optimally, if these systems can in fact maximize the SNR adaptively, that is, adjust to the changing noise conditions in everyday listening environments, it could obviate the need for manual switching.

Kuehnel and Checkley (2000) compared the performance in speech recognition in noise between an adaptive directional option and a fixed directional option (with cardioid directivity pattern), in the same dual-microphone system. There was no difference in performance between the two options when the noise was positioned at 180 degrees, that is, directly behind the participants. This was expected as the noise from the back receives maximal attenuation in the fixed directional option (cardioid) as well as in the adaptive directional option, which also chooses a cardioid pattern for noise in this condition. However, there was significantly better performance in the adaptive directional mode when the noise was presented from the sides of the participants (90 and 270 degrees, with regard to speech signal at 0 degrees azimuth). The average SNR for 50% correct score was 2 dB better than the fixed directional option. These results seem to indicate that the system was actively changing its directivity pattern to match the direction of the noise. Similar results were obtained in a later study by Ricketts and Henry (2002) in that they found the adaptive directional option resulted in better performance than the fixed directional option when noise was directed from the sides. However, there was no difference in performance between the two options when the noise was presented in a diffuse field, from the back (cardioid pattern), nor in a panning fashion from one speaker to another in clockwise direction. Ricketts and Henry calculated the audibility index-weighted polar patterns, which resulted in the adaptive directional mode providing up to 9.5 dB more attenuation than the fixed directional option when noise was positioned from 30 to 130 degrees (with regard to signal at 0 degrees azimuth).

A recent study by Bentler et al (2004) investigated whether or not the adaptive advantage of the dual-microphone system could be preserved by creating a listening environment with multiple noise sources in which a dominant noise source moved randomly around the listener. They found no difference between the adaptive directional option and the fixed directional option in speech recognition tests both in an anechoic and a moderately reverberant listening environment. These laboratory measures were also consistent with the self-report measures from the participants after a three-week trial with each of the two directional settings.

The above studies indicate that in a dual-microphone system, the adaptive directional option can add improved directivity over that of a fixed directional pattern (null typically at 180 degrees) when there is a single noise source from the side of the listener. With the presence of multiple noise sources, the adaptive advantage would be lost as the microphone then defaults to a fixed directivity pattern.

### *Nonlinear Hearing Aid Prescriptions*

The National Acoustic Laboratories Nonlinear 1 (NAL-NL1) hearing aid prescription formula's goal (Dillon, 1999) is to provide a gain and frequency response that maximizes speech intelligibility for any input level of speech. The overall loudness is normalized to that of a normal hearing listener, or when possible, loudness is lowered if it offers higher speech intelligibility particularly for higher level inputs.

The Digital Perception Processing (DPP) prescription is a proprietary algorithm provided by the Phonak Fitting Guideline software. This prescription incorporates a psychoacoustic model of normal and impaired cochlear function as derived from the work of Launer and Moore (2003). The goal of the DPP algorithm is to restore normal loudness perception. With different processing goals (that is, the NAL-NL1 focuses on maximizing speech intelligibility and DPP focuses on normalizing loudness), there may be differences in amplification outcomes between the two prescriptions.

In current digital hearing aid technology, both multichannel noise reduction systems and adaptive directional dual-microphone systems can be incorporated into the same hearing aid. Relatively few studies had been conducted to investigate the comparative performance between a multichannel noise reduction system and an adaptive directional dual-microphone system in speech recognition, especially in the same hearing aid. The purpose of this investigation was to compare the speech recognition performance from the same digital hearing aid implementing the NAL-NL1 and DPP prescriptions under the following three settings: (1) multichannel noise reduction only, (2) adaptive directional dual microphone only, and (3) multichannel noise reduction plus adaptive directional dual microphone.

## METHODS

### Participants

The participants were nine adults (six men and three women) who ranged in age from 39 to 79 years (M = 57.8; SD = 13.7). They had documented moderate to moderately severe bilaterally symmetrical sensorineural hearing loss. All participants were experienced hearing aid users and had at least six months of hearing aid use (range of hearing aid use = 6–15 months; M = 9.1; SD = 2.7). All participants had worn only one hearing aid prior to participation in the study, and the fitted ear was selected to use the trial hearing aid in the study with the subject's custom earmold. Five out of nine participants' (S4, S6-S9) own hearing aids were with wide dynamic range compression circuits while the others had output limiting compression circuits. Only two hearing aids (of S6 and S9) had directional microphones, and three hearing aids (of S4, S6, and S9) had multichannel noise reduction systems.

### Hearing Aid

Depending on the degree of hearing loss of individual participants, either the Phonak Perseo 211 or Perseo 311 behind the ear (BTE) hearing aid was prescribed for use in this study. These hearing aids offer both a multichannel noise reduction system (Fine-scale Noise Canceller [FNC]) and an adaptive directional dual-microphone system (Digital Audio Zoom [DAZ]). FNC and DAZ can be activated together or alone in the Perseo hearing aid.

The FNC system provides frequency dependent and gain adjustments after the SNR is estimated in each of the 20 frequency channels. The gain is not reduced for channels with frequencies centered between 900 to 2500 Hz, which are assumed to be the most important frequencies for speech recognition. SNR-dependent gain adjustments are made in the frequency channels below 900 Hz and above 2500 Hz when the SNR is estimated to be poor. The aids used in this study allow the selection of the "amount" of FNC from "off" (no noise cancellation) to "strong" (highest noise cancellation algorithm). The level of "moderate" was used in this study. The amount of frequency dependent and SNR dependent gain reductions are displayed in Figure 1.

In the DAZ system of Perseo, the front-bank ratio detector estimates the location of the dominant input signal. Then there is a modulation detector that analyzes the modulation pattern and the spectral distribution of the incoming signal and determines the presence or absence of speech and noise. The internal delay of the posterior microphone was adjusted until the system determines the polar pattern that yields the minimum power output. Any polar pattern with nulls between 90 to 270 degrees can be chosen. The adaptation speed for changing

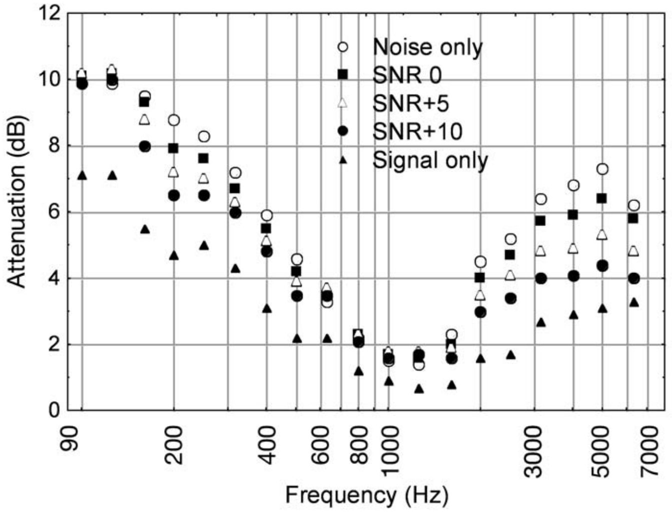



**Figure 1.** The amount of gain reductions in different frequency regions at different SNR estimations when using the "moderate" option of the Fine-scale Noise Canceller (FNC; a multichannel noise reduction system) in the Perseo hearing aid. (Data courtesy of Juergen Tchorz, Phonak AG.)

between polar patterns for maintaining the minimum power output is 100 msec. When the system detects multiple noise sources, the directionality will be reverted to a fixed cardioid pattern. For a detailed description of the system, please refer to Chung (2005).

## Procedures

The study was described to prospective participants, and written consent was obtained. Pure-tone air-conduction thresholds were obtained for the frequencies of 250, 500, 1000, 2000, 4000, and 8000 Hz, using a GSI-61 audiometer (re: American National Standards Institute [ANSI] S3.6, 1996). The average pure-tone threshold values are shown in Figure 2.

The participants wore the study hearing aid for two weeks. The NAL-NL1 and the DPP prescriptions were each used for one week, in randomized order across participants. During the trial, one program in the hearing aid was given that had both the FNC and DAZ functions activated simultaneously all the time. Another program was given with both functions disabled, therefore operating under omnidirectional mode without multichannel noise reduction. The participants were advised to use the FNC plus DAZ program as much as possible. Speech recognition and subjective benefit evaluations were conducted at the end of each week. A paired comparison judgment on sound quality between the two prescriptions was conducted at the end of the second week.

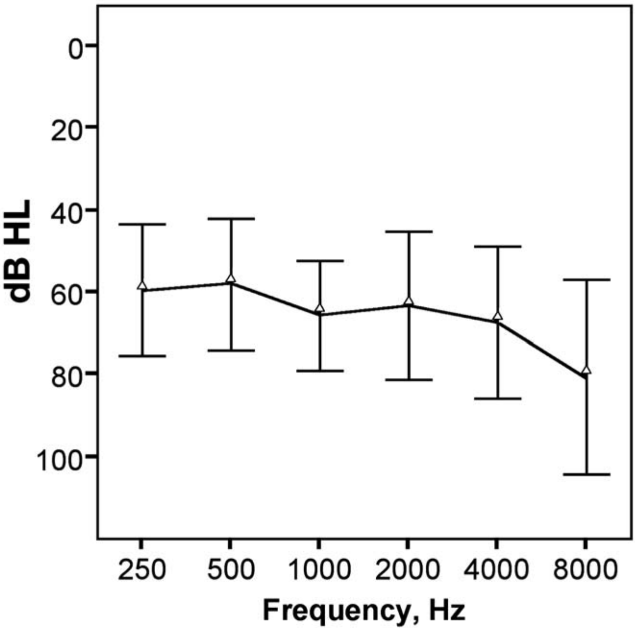



**Figure 2.** The average pure-tone threshold values for the subjects with +/-1 standard deviation indicated.

Individual subject's measurements of hearing thresholds, real ear unaided response, and real-ear-to-coupler difference were entered into the Phonak Fitting Guideline 8.3 software to generate individual NAL-NL1 and DPP prescriptions.

Speech recognition was assessed using the Cantonese Hearing-in-Noise Test (CHINT; Wong and Soli, 2005) with two test conditions: (1) signal-front/noise-front and (2) signal-front/noise-side. For each test condition, three settings in the hearing aid were separately evaluated with one sentence list (20 sentences) of CHINT: (1) FNC only, (2) DAZ only, and (3) FNC plus DAZ (FNC + DAZ). In the FNC-only condition, the microphone was in omnidirectional mode. The test order for the two test conditions and the three hearing aid settings were randomized. Each CHINT sentence list consists of 20 sentences, each containing ten syllables. The participants were instructed to repeat each sentence spoken by a male speaker in the presence of speech-spectrum weighted noise presented at a fixed level at 65 dBA. The presentation level of the sentence was adjusted adaptively, dependent on the correctness of the response of the previous sentence. The SNR for 50% correct score performance (hereafter called "reception threshold of sentences," or "RTS") was calculated by averaging the presentation level of sentences numbered 5 through 20. At least one practice list was presented to familiarize the participants with the test procedure and the speaker's voice. Out of the nine participants, RTS were obtained for only two participants (S1 and S2) from the practice lists. The participants then continued with formal testing with the same adaptive procedures. For the other seven participants (S3 to S9), the RTS exceeded the response variability limit from the practice lists, suggesting that the participants' responses were not consistent and the test results could not be considered reliable. Therefore, fixed SNR testing procedure was used rather than the adaptive procedure. The fixed SNR was determined for each subject using practice lists that gave scores that fell in the range of 30–70%, and was maintained throughout the study for all test conditions.

In order to investigate if similar frequency response and gain were provided across the three hearing aid conditions, FNC only, DAZ only, and FNC + DAZ 2cc coupler measurements were obtained by placing the hearing aid with the front microphone port at zero degrees azimuth facing the sound source in the sound chamber. Figure 3 shows the frequency and gain of the hearing aid with a 65 dBSPL input of Digital Speech ANSI signal (interrupted composite signal with ANSI speech spectra from ANSI S3.42 [1992] standard) for the three hearing aid conditions from subject S1. This indicates that the programming among these conditions did not significantly change the frequency response and gain level. The microphone was therefore equalized between the adaptive directional options (in DAZ only or FNC + DAZ) and the omnidirectional option (in FNC only).

The Chinese translation of the Abbreviated Profile of Hearing Aid Benefit (APHAB; Cox and Alexander, 1995) was used to evaluate the subjective benefits of the two hearing aid prescriptions. For the sound quality paired comparison judgment task, 19 prerecorded soundtracks comprising environmental sounds, speech monologue, music and singing voices were presented in quiet. The NAL-NL1 and DPP prescriptions were presented in randomized order while the soundtracks were played. The participants were blinded on which prescription was presented. FNC and DAZ were both disabled during this evaluation.

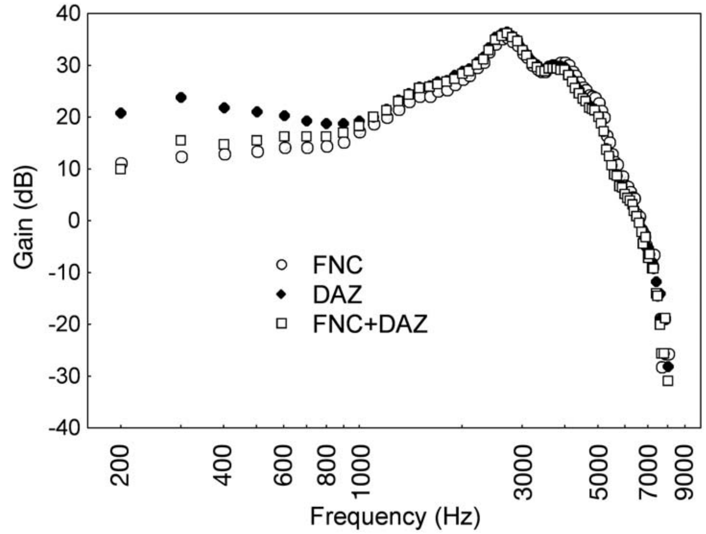



**Figure 3.** 2cc coupler gain response of the three hearing aid settings: FNC only, DAZ only, FNC + DAZ of subject S1.

## RESULTS

### Speech Recognition Performance

Individually for the signal-front/noise-front condition and the signal-front/noise-side condition, speech recognition performances from CHINT were compared between hearing aid prescriptions (NAL-NL1 and DPP) and among hearing aid settings (FNC only, DAZ only, and FNC + DAZ).

For the data analyses of participants S1 and S2, the 95% confidence intervals (CI) of the normative RTS across-list differences from Wong and Soli (2005) were used to delineate

**Table 1. The Difference in RTS between Test Conditions Is Compared with the 95% CI of the Normative RTS across-List Differences from Wong and Soli (2005) for Participants S1 and S2**

| | | Difference in RTS between test conditions | | | |
|---|---|---|---|---|---|
| | | NAL-NL1 vs. DPP | DAZ vs. FNC | FNC + DAZ vs. FNC | FNC + DAZ vs. DAZ |
| Subject | 95% CI of RTS across-list difference | | | | |
| | | Signal-front/Noise-front | | | |
| S1 | 2.2 | ND (1.00) | ND (0.80) | ND (2.15) | ND (1.35) |
| S2 | 2.2 | ND (0.07) | ND (0.35) | ND (0.75) | ND (0.90) |
| | | Signal-front/Noise-side | | | |
| S1 | 2.4* | ND (0.73) | DAZ (4.55) | ND (0.05) | DAZ (4.6) |
| S2 | 1.7* | ND (0.17) | DAZ (9.05) | FNC + DAZ (9.75) | ND (0.7) |

*Note:* ND = No difference; DAZ = adaptive directional microphone system; FNC = multichannel noise reduction system. The upper panel shows the signal-front/noise-front test conditions, and the lower panel shows the signal-front/noise-side test conditions. If any hearing aid prescription and hearing aid setting gave significantly better performance (exceeding the 95% CI), this is indicated for each subject; otherwise no difference is noted.

*The 95% CI of normative RTS across-list differences were 2.4 and 1.7 for noise from left (in S1) and noise from right (in S2), respectively.

if there was any statistical difference between test conditions. In other words, difference between conditions exceeding the 95% CI was considered to be statistically significant. The comparisons are shown in Table 1.

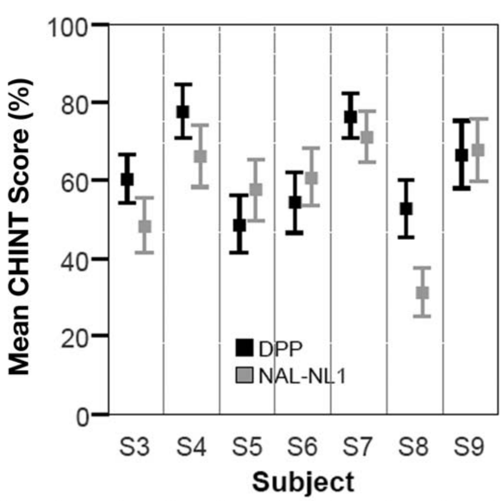
Since RTS could not be obtained from participants S3-S9 due to response inconsistencies, percentage correct scores at fixed SNR were obtained. For the data analyses of participants S3 to S9, a two-way repeated measures analysis of variance (ANOVA) was employed to investigate the main effects of (1) hearing aid prescriptions and (2) hearing aid settings of individual subject's data, using the scores of individual sentences as separate observations. Post hoc Tukey Honest Significant Difference test was employed to check for any significant difference at $p < .05$ between test conditions.

### *Signal-Front/Noise-Front Test Condition*

Figure 4 shows the individual CHINT performance between the two hearing aid prescription formulae with data collapsed from the three hearing aid settings. Figure 5 shows the individual CHINT performance among the three hearing aid settings, with data collapsed from the two hearing aid prescriptions. In Figure 4, performance with the DPP prescription was significantly better than from the NAL-NL1 prescription in only one subject (S8). In Figure 5, performance across the three hearing aid settings of FNC only, DAZ only, and FNC + DAZ was not significantly different.

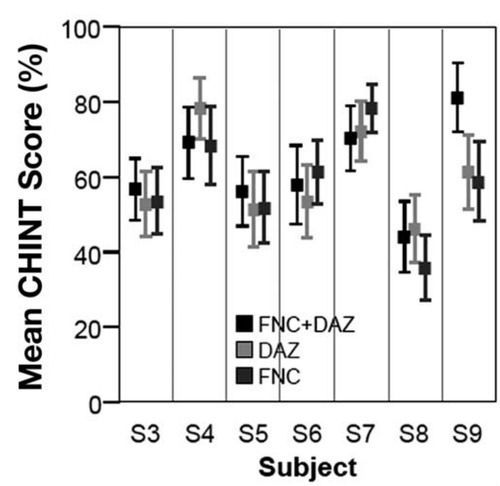
### *Signal-Front/Noise-Side Test Condition*

Figure 6 shows the individual CHINT performance between the two hearing aid prescriptions with collapsed data from all three hearing aid settings. Figure 7 shows the individual CHINT performance among the three hearing aid settings with collapsed data from the two hearing aid prescriptions. In Figure 6, DPP prescription was significantly better compared to the NAL-NL1 prescription in four participants (S5, S7, S8, and S9). In Figure 7, performance with DAZ was significantly better than with FNC for all participants except S6; performance with FNC + DAZ was significantly better than with FNC



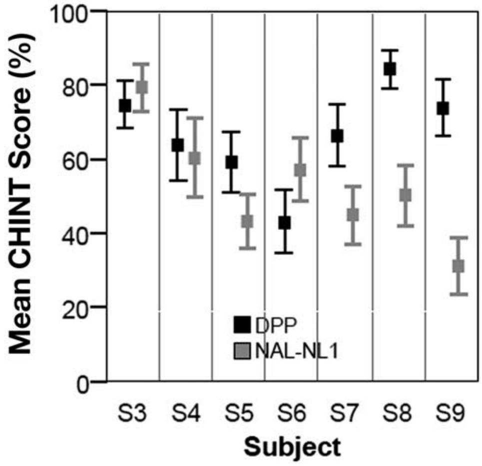

| | S1 | S2 | S3 | S4 | S5 | S6 | S7 | S8 | S9 |
|---|---|---|---|---|---|---|---|---|---|
| SNR | - | - | 0 | 0 | +5 | +10 | +5 | -5 | +5 |
| DPP vs NAL | | | | | | | | DPP | |

**Figure 4.** Individual subject CHINT performance of DPP and NAL-NL1 hearing aid prescriptions in the signal-front/noise-front condition.



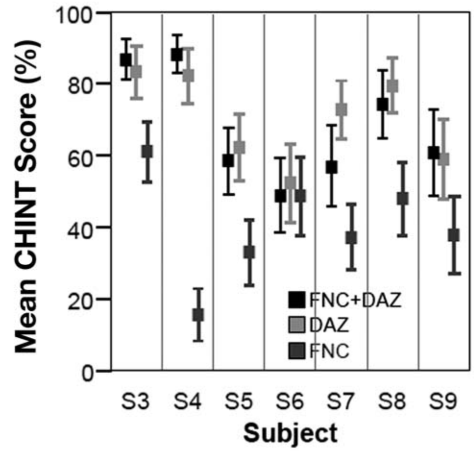

| | S1 | S2 | S3 | S4 | S5 | S6 | S7 | S8 | S9 |
|---|---|---|---|---|---|---|---|---|---|
| SNR | - | - | 0 | 0 | +5 | +10 | +5 | -5 | +5 |
| FNC+DAZ vs DAZ | | | | | | | | | |
| FNC+DAZ vs FNC | | | | | | | | | |
| DAZ vs FNC | | | | | | | | | |

**Figure 5.** Individual subject CHINT performance of the three hearing aid settings in the signal-front/noise-front condition.




| | S1 | S2 | S3 | S4 | S5 | S6 | S7 | S8 | S9 |
|---|---|---|---|---|---|---|---|---|---|
| SNR | - | - | 0 | 0 | +5 | +10 | +5 | -5 | +5 |
| DPP vs NAL | | | | | DPP | | DPP | DPP | DPP |

**Figure 6.** Individual subject CHINT performance of DPP and NAL-NL1 hearing aid prescriptions in the signal-front/noise-side condition.




| | S1 | S2 | S3 | S4 | S5 | S6 | S7 | S8 | S9 |
|---|---|---|---|---|---|---|---|---|---|
| SNR | - | - | 0 | 0 | +5 | +10 | +5 | -5 | +5 |
| FNC+DAZ vs DAZ | DAZ | | | | | | | | |
| FNC+DAZ vs FNC | | FNC+ DAZ | FNC+ DAZ | FNC+ DAZ | FNC+ DAZ | | | FNC+ DAZ | FNC+ DAZ |
| DAZ vs FNC | DAZ | DAZ | DAZ | DAZ | DAZ | | DAZ | DAZ | DAZ |

**Figure 7.** Individual subject CHINT performance of the three hearing aid settings in the signal-front/noise-side condition.

in six out of nine participants, whereas performance from DAZ was significantly better than FNC + DAZ in one subject.

In Figures 4 to 7, for any significant difference ($p < .05$, Tukey post hoc HSD test) between any two conditions for a subject, there was an entry of the better condition in the table under the column of that subject. The fixed SNRs used for each subject are listed in the table, except for S1 and S2, for which RTS (rather than fixed SNR) was used as the outcome measure. Error bars of the figures show 95% confidence intervals of mean percentage correct score for S3 to S9.

### *Subjective Benefits*

The Wilcoxon Signed-Ranks Test was used to analyze the individual and group performance between NAL-NL1 and DPP prescriptions on the four subscales of APHAB: ease of communication (EC), reverberation (RV), background noise (BN), and adversiveness (AV). From group data, Table 2 shows the sum of ranks of the two prescriptions. The larger the sum of ranks, the lower the occurrence of problems in the listening situations. DPP gave lower occurrence of listening problems than NAL-NL1 in the RV subscale ($p < .05$), whereas NAL-NL1 gave

**Table 2. The Sum of Ranks of the Group Performance of the Four APHAB Subscales from the Wilcoxon Signed-Ranks Test**

| APHAB subscale | Sums of negative ranks (NAL-NL1 better) | Sums of positive ranks (DPP better) | *p* value |
|---|---|---|---|
| Ease of communication (EC) | 323.0 | 343.0 | 0.87 |
| Reverberation (RV) | 70.5 | 205.5 | 0.04 |
| Background noise (BN) | 1129.5 | 950.5 | 0.55 |
| Adversiveness (AV) | 474.5 | 155.5 | 0.01 |

lower occurrence of listening problems than DPP in the AV subscale ($p < .05$). No difference was found between the prescriptions for the BN and EC subscales. Nevertheless, from individual subject data, no difference was found between the two prescriptions in any of the subscales.

### *Paired Comparison Judgment on Sound Quality*

Table 3 shows the number of soundtracks each subject preferred for each hearing aid prescription. Wilcoxon Signed-Ranks Test showed no significant difference between the preference for the two hearing aid prescriptions for group data ($p = .92$); and Chi-square Test showed no significant difference between the prescriptions for individual subject data ($p > .05$). Recall that the test was conducted with both FNC and DAZ deactivated. Therefore the hearing aid operated under omnidirectional mode without multichannel noise reduction.

**Table 3. The Number of Soundtracks Preferred for Each Hearing Aid Prescription in the Paired Comparison Judgment on Sound Quality**

| | No. of soundtracks preferred for each hearing aid prescription | |
|---|---|---|
| | NAL-NL1 | DPP |
| S1 | 6 | 12 |
| S2 | 7 | 12 |
| S3 | 7 | 11 |
| S4 | 6 | 12 |
| S5 | 6 | 10 |
| S6 | 9 | 9 |
| S7 | 10 | 7 |
| S8 | 10 | 8 |
| S9 | 12 | 6 |

## Discussion

The speech recognition results indicated that when the signal and noise were both from the front, there was no difference in performance between NAL-NL1 and DPP prescriptions, except that only one subject performed better with DPP. When the noise was moved to the side, four participants (S5, S7-S9) performed better with DPP, while no participants performed better with NAL-NL1. Difference in real ear SPL output (DPP minus NAL-NL1) for Digital Speech ANSI signal inputs at 50, 65 and 80 dBSPL is shown in Figure 8 with the FNC and DAZ settings both disabled. DPP prescribed up to 20 dB less output than NAL-NL1 in the high frequencies (around the 5000 Hz region), and a little more output in the mid- to low frequencies for S5 and S7. In contrast, DPP prescribed up to 10–15 dB more output than NAL-NL1 across the frequencies for S8 and S9.

The above results suggested that absolute audibility could not explain the reason why those four participants performed better with the DPP prescriptions. NAL-NL1 aimed at maximizing speech intelligibility in which band importance function was used in the calculation of the speech intelligibility index (SII) for generating the prescriptive targets (Ching et al, 2001). NAL-NL1 prescriptive targets were generated based on the band importance function of English, which can be quite different from the band importance function of Cantonese. For example, in sentence material, the band importance function for

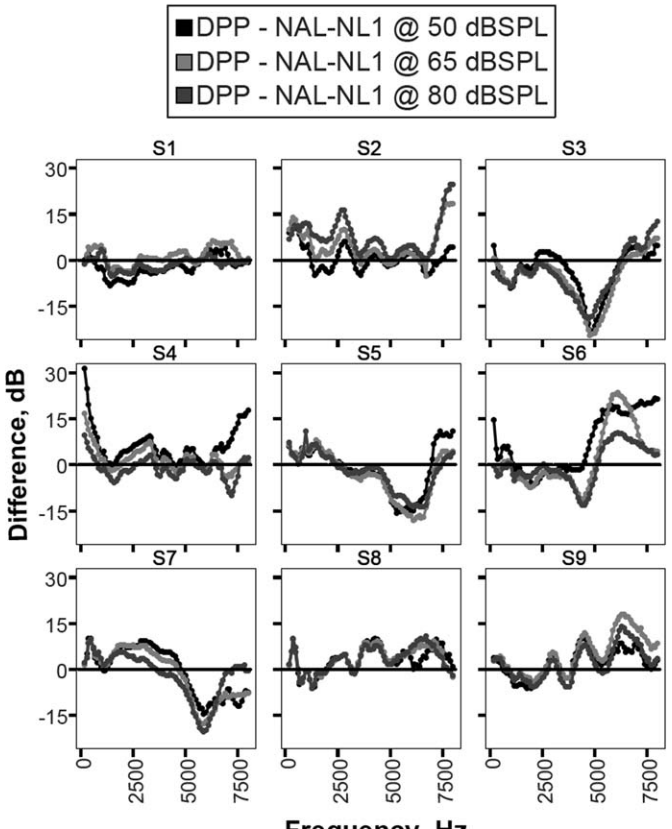



**Figure 8.** DPP minus NAL-NL1 real ear outputs for 50, 65, and 80 dBSPL Digital Speech ANSI signal inputs for each subject.

English (Eisenberg et al, 1998) peaked at around 2000 Hz, whereas in Cantonese the function shifted toward the low-frequency region and peaked at 800 Hz to 1600 Hz (Chua, 2004). Compared with English, the fundamental frequency in Cantonese may carry a heavier weight in the overall speech recognition of the language than in English since F0 level and contour is primarily responsible for providing the semantic information for differentiating different lexical tones. The prescriptive targets therefore might not have maximized the speech intelligibility for Cantonese equally well as for English. Unlike NAL-NL1, DPP was based on a psychoacoustic cochlear excitation model, which does not involve any language-specific issues in the calculation of prescriptive targets. Comparatively, DPP might be better than NAL-NL1 in achieving the goal of its prescription philosophy for the four participants who performed better with DPP, when the noise and signal were spatially separated.

No difference in performance was found among the three hearing aid settings: FNC only, DAZ only, and FNC + DAZ, when both signal and noise were from the front. This means that speech recognition in noise performance was dependent neither on the multichannel noise reduction option nor the adaptive directional microphone option when speech and noise are not spatially separated and presented from the front. The situation completely changed when the signal came from the front and the noise from the side. With the adaptive directional microphone option enabled, either alone or with the multichannel noise reduction option, the performance was better than the condition with the multichannel noise reduction option enabled alone for most of the participants. The multichannel noise reduction option enabled alone did not give better performance than any of the other two settings in any of the participants. This concludes that the adaptive directional microphone system was more effective than the multichannel noise reduction system in providing a better speech recognition in noise performance when the noise is spatially separated (from the side) from the signal (from the front), possibly by providing a better SNR listening environment.

An explanation for the superior performance of the adaptive directional microphone option over the multichannel noise reduction option was that the performance of the adaptive directional microphone option was optimized with noise directed from the sides. Recall from Ricketts and Henry's (2002) study that the adaptive directional microphone option was found to give the best speech recognition scores and the best measured AI-weighted directional pattern when the noise source was directed from the sides, in comparison with the fixed directional (cardioid) and the omnidirectional conditions.

The results of this study coincide with those obtained by Alcantara et al (2003), which indicate that the benefits of the multichannel noise reduction were limited in terms of speech recognition in noise, regardless of the direction of the noise. This study was conducted with a fixed noise source. In everyday listening situations, noise sources are variable and may come from different directions. As Bentler et al (2004) indicated, the directional advantage of the adaptive directional microphone can be significantly reduced in the presence of multiple noise sources. It can be anticipated that, in multiple noise source conditions or in everyday listening environments, the advantage of the adaptive directional microphone option over the multichannel noise reduction option would be reduced or even diminished.

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## REFERENCES

Agnew J, Black M. (1997) HINT thresholds for a dual microphone BTE. *Hear Rev* 26:29–30.

Alcantara JI, Moore BCJ, Kuhnel V, Launer S. (2003) Evaluation of the noise reduction system in a commercial digital hearing aid. *Int J Audiol* 42:34–42.

American National Standards Institute (ANSI). (1992) S3.42 *American National Standard: Testing Hearing Aids with a Broad-Band Noise Signal/Accredited Standards Committee S3, Bioacoustics*. New York: American National Standards Institute.

American National Standards Institute (ANSI). (1996) S3.6 *Specifications for Audiometers*. New York: American National Standards Institute.

Baer T, Moore BCJ, Gatehouse S. (1993) Spectral contrast enhancement of speech in noise for listeners with sensorineural hearing impairment: effects on intelligibility, quality and response times. *J Rehabil Res Dev* 30:49–72.

Bentler RA, Tubbs JL, Egge JLM, Flamme GA, Dittberner AB. (2004) Evaluation of an adaptive directional system in a DSP hearing aid. *Am J Audiol* 13:73–79.

Boll SF. (1979) Suppression of acoustic noise in speech using spectral subtraction. *IEEE Trans Acoust* 27:113–120.

Ching TYC, Dillon H, Katsch R, Byrne D. (2001) Maximizing effective audibility in hearing aid fitting. *Ear Hear* 22:212–224.

Chua EWW. (2004) Speech recognition predictability of a Cantonese speech intelligibility index. MSc thesis, The University of Hong Kong, Hong Kong.

Chung K. (2005) Challenges and recent developments in hearing aids, Part I. Speech understanding in noise, microphone technologies and noise reduction algorithms. *Trends Amplif* 8(3):83–124.

Cox RM, Alexander GC. (1995) The Abbreviated Profile of Hearing Aid Benefit (APHAB). *Ear Hear* 16:176–186.

Dillon H. (1999). NAL-NL1: a new procedure for fitting non-linear hearing aids. *Hear J* 52:10, 12, 14, 16.

Dillon H, Lovegrove R. (1993) Single-microphone noise reduction systems for hearing aids: a review and an evaluation. In: Studebaker GA, Hochberg I, eds. *Acoustical Factors Affecting Hearing Aid Performance.* Boston: Allyn and Bacon, 353–372.

Eisenberg LS, Dirks DD, Takayanagi S, Martinez AS. (1998) Subjective judgments of clarity and intelligibility for filtered stimuli with equivalent speech intelligibility index predictions. *J Speech Lang Hear Res* 41:327–339.

Gravel JS, Fausel N, Liskow C, Chobat, J. (1999) Children's speech recognition in noise using omni-directional and dual-microphone hearing aid technology. *Ear Hear* 20:1–11.

Hawkins DB, Yacullo WS. (1984) Signal-to-noise ratio advantage of binaural hearing aids and directional microphones under different levels of reverberation. *J Speech Hear Disord* 49:278–286.

Killion M, Niquette P. (2000) What can the pure-tone audiogram tell us about a patient's SNR loss? *Hear J* 53(3):46–53.

Kompis M, Dillier N. (1994) Noise reduction for hearing aids: combining directional microphones with an adaptive beamformer. *J Acoust Soc Am* 96:1910–1913.

Kuehnel V, Checkley PC. (2000) Advantages of an adaptive multi-microphone system. *Hear Rev* 7(5):58–60, 74.

Launer S, Moore BCJ. (2003) Use of a loudness model for hearing aid fitting. V. On-line gain control in a digital hearing aid. *Int J Audiol* 42:262–273.

Levitt H. (1993) Digital hearing aids. In: Studebaker GA, Hochberg I, eds. *Acoustical Factors Affecting Hearing Aid Performance*. Boston: Allyn and Bacon, 317–335.

Lurquin P, Delacressonniere C, May A. (2001) Examination of a multi-band noise cancellation system. *Hear Rev* 8(1):48–49, 52, 54, 60.

Madison TK, Hawkins DB. (1983) The signal-to-noise ratio advantage of directional microphones. *Hear Instrum* 34(2):18, 49.

Mueller GH, Grimes AM, Erdman SA. (1983) Subjective ratings of directional amplification. *Hear Instrum* 34:14–16, 47.

Nielsen HB, Ludvigsen C. (1978) Effect of hearing aids with directional microphones in different acoustic environments. *Scand Audiol* 7:217–224.

Peterson PM, Durlach NI, Rabinowitz WM, Zurek PM. (1987) Multimicrophone adaptive beamforming for interference reduction in hearing aids. *J Rehabil Res Dev* 24:103–110.

Pumford JM, Scollie SD, Jenstad LM. (2000) Speech recognition with in-the-ear and behind-the-ear dual microphone hearing instruments. *J Am Acad Audiol* 11:23–35.

Ricketts T, Dittberner AB. (2002) Directional amplification for improved signal-to-noise ratio: strategies, measurement, and limitations. In: Valente M, ed. *Strategies for Selecting and Verifying Hearing Aid Fittings*. 2nd ed. New York: Thieme Medical, 274–346.

Ricketts T, Henry P. (2002) Evaluation of an adaptive directional-microphone hearing aid. *Int J Audiol* 41:100–112.

van Dijkhuizen JN, Festen JM, Plomp R. (1991) The effect of frequency-selective attenuation on the speech reception threshold of sentences in conditions of low-frequency noise. *J Acoust Soc Am* 90:885–894.

Voss T. (1997) Clinical evaluation of users' performance with multi-microphone hearing instruments. *Hear Rev* 36:45–46, 74.

Weiss M. (1987) Use of an adaptive noise canceller as an input preprocessor for a hearing aid. *J Rehabil Res Dev* 24:93–103.

Weiss M, Neumann AC. (1993) Noise reduction in hearing aid. In: Studebaker GA, Hochberg I, eds. *Acoustical Factors Affecting Hearing Aid Performance.* Boston: Allyn and Bacon, 337–352.

Wong LLN, Soli SD. (2005) Development of the Cantonese Hearing in Noise Test (CHINT). *Ear Hear* 26(3):276–289.