The Effect of Digital Noise Reduction Time Constants on Speech Recognition in Noise

VANDERBILT UNIVERSITY

JASON GALSTER AND TODD RICKETTS ▪ VANDERBILT BILL WILKERSON CENTER

ABSTRACT

Two Digital Noise Reduction (DNR) algorithms used in two different commercially available hearing aids were evaluated. All subjects were fit bilaterally with production model hearing aids, which were matched for parameters allowed by each manufacturer's software. The focal point distinguishing the two hearing aid models was the time constants used in the processing strategies of each hearing aids DNR system. The first model (HA-S) used comparatively slow time constants in its DNR algorithm, while the second model (HA-F) used fast time constants. Speech recognition in noise thresholds were gathered using the Hearing in Noise Test (HINT) for conditions of active and inactive DNR processing. Implementation of the DNR algorithm resulted in significantly decreased performance for subjects when fit with HA-S. In contrast, significantly improved performance was associated with activating the DNR algorithm in HA-F. It was proposed that this benefit may originate from the extremely fast attack and release time constants used in the DNR algorithm.

INTRODUCTION

Various Digital Signal Processing (DSP) techniques have been applied to hearing aids in an attempt to improve speech recognition in noise ability. One scheme, generally referred to as Digital Noise Reduction (DNR), has shown isolated possibility for improving speech recognition in noise[1,2,3]. DNR strategies are of significant interest in part because, unlike multiple microphone based approaches, they are not limited by the spatial relationship between primary and competing sounds. That is, benefit may be expected regardless of source location.

Noise reduction systems utilize a processing algorithm to identify and classify dynamic aspects of the incoming signal. In order for the noise to be properly managed, the DNR algorithm must accurately identify the presence or absence of a speech signal. In order to accomplish this DNR systems implement an analysis time window, during which characteristics of the input signal are monitored to determine the presence or absence of a noise-like signal. If a signal which is noise-like is not detected during this analysis window gain is left unaltered. When noise has been identified within a channel gain in that channel is then adjusted based on characteristics specific to the DNR algorithm.

It is hypothesized that DNR systems which have the ability to improve performance in noise will require extremely fast time constants within the active DNR system. That is, the ability for an algorithm to react to changes within very short periods of time would be crucial for a release from masking to occur both at the onset and within any speech-like segments.

Hypotheses

1) On average, hearing aid wearers will show similar speech recognition in a background of steady-state noise when fitted with two commercial digital hearing aid models with DNR deactivated and matched for output.
2) The implementation of a DNR system with fast time constants will result in improved speech recognition in a background of steady-state noise in comparison to the same hearing aid with DNR deactivated.
3) The implementation of a DNR system with slow time constants will not result in improved speech recognition in a background of steady-state noise in comparison to the same hearing aid with DNR deactivated.

METHODS

Mean Audiometric Threshold

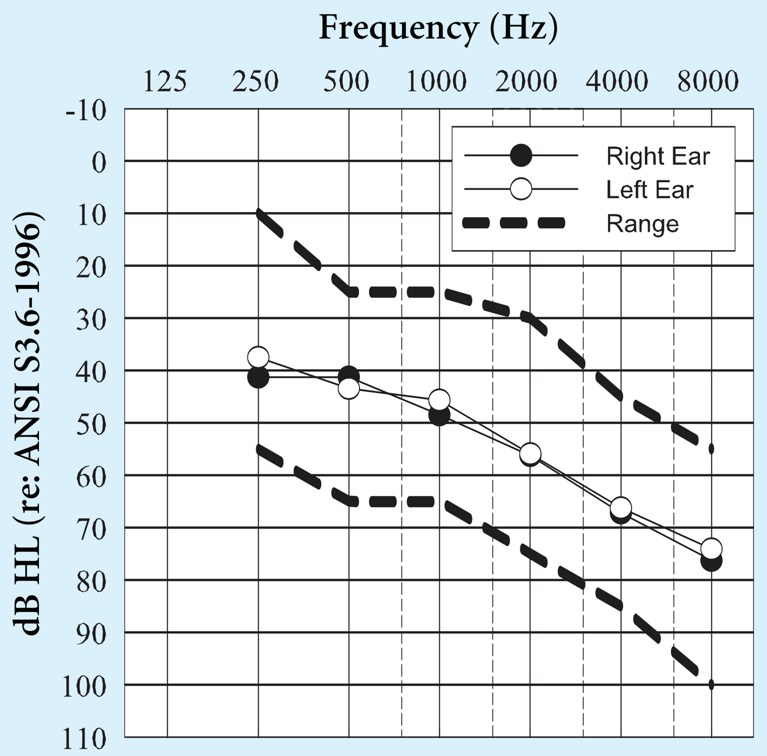

Figure 1. Average audiometric threshold for the left and right ears of all subjects tested.

In-Situ Output Levels

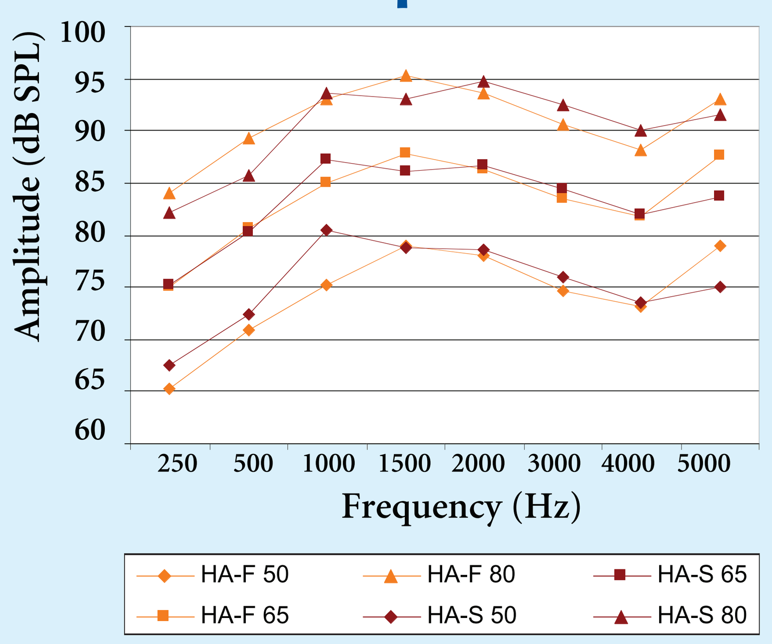

Figure 2. Displays the SPL output levels at the eardrum for all subjects and each hearing aid at input levels of 50, 65, and 80.

	Compression Threshold Knee Point	Average Low Frequency Compression Ratio	Average High Frequency Compression Ratio	Compression Attack Time Constants	Compression Release Time Constants
HA-S	38	2.0:1	2.0:1	5 msec	80 msec
HA-F	40	1.5:1	1.5:1	5 msec	6 msec

Table 1. Average values of adjusted compression parameters for HA-S and HA-F

- Subjects- 15 male and female adults ages 18-70 with sloping, bilaterally symmetrical, mild to moderately/severe, sensorineural hearing impairment (Figure 1).
- The two hearing aids used 8 and 9 channels; each variable parameter is listed in Table 1.
- Verification of gain and output was completed via interrupted composite noise (Fonix 6500; ANSI S3.22-1996).
- All fittings were matched within an RMS difference of 2.57 dB. The largest level discrepancy between HA-S and HA-F was 5.2 dB and was observed at 1000Hz for an input level of 50 dB (Figure 2).
- The Hearing in Noise Test (HINT) was used to assess speech recognition in noise under four listening conditions (two hearing aid types with the noise reduction system active and inactive).
- Each of the test conditions was evaluated using four, ten-sentence blocks randomly selected without replacement. Presentation order was counter balanced across conditions.
- A period of 8-10 seconds of steady-state noise preceded the onset of speech stimulus to insure that the noise reduction algorithms were active.
- A single source loudspeaker (Tannoy System 600™) placed at 0˚ azimuth relative to the listener was used for all testing. All testing was performed in an anechoic chamber (3m x 3m x 3m).
- Several techniques were used to ensure equal audibility of the speech signal across test conditions.

Data Analysis

Performance on the HINT across test conditions was analyzed using a two factor repeated measures analysis of variance (ANOVA). The within-subject factors (independent variables) were hearing aid model and the presence or absence of noise reduction systems. Follow up analysis was completed using one-way ANOVAs for the relationships observed both between and within hearing aid types. Statistical significance was defined at the .05 level.

RESULTS

Subject Performance on the HINT in Steady State Noise

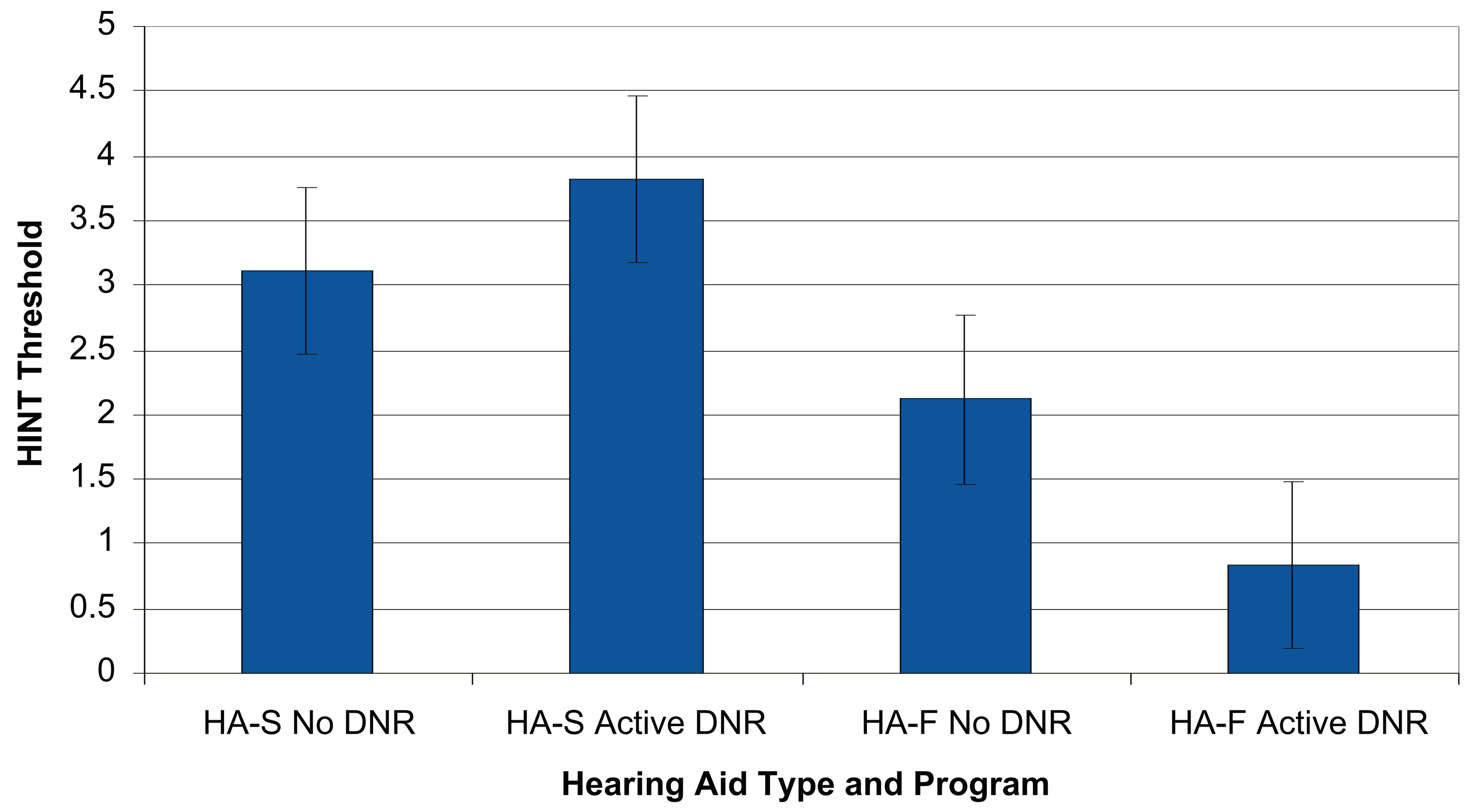

Figure 3. Subject performance across all four hearing aid conditions on the HINT test in a background of speech weighted steady-state noise.

- A significant main effect of hearing aid type was revealed $F(1,14)=40.534$, $p< .001$. A significant interaction between hearing aid type and DNR condition was also observed $F(1,14)=28.89$, $p<.001$.
- A significant increase in performance of 1.28 dB was observed within HA-F between active and inactive DNR conditions, $F(1,14)=15.845$, $p< .001$.
- A significant decrease in performance of 0.7 dB was observed within HA-S between active and inactive DNR conditions, $F(1,14)=16.561$, $p< .001$.
- Performance between HA-S and HA-F when each device was not using a DNR strategy revealed a significant increase in performance of .99 dB, $F(1,14)=11.317$, $p< .001$.

DISCUSSION

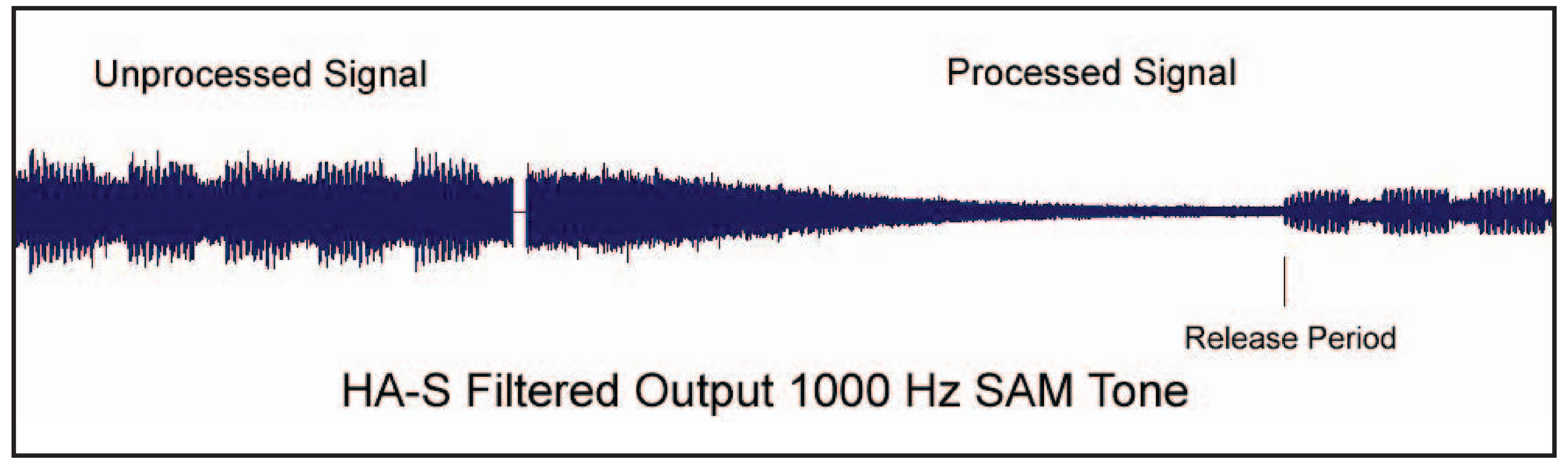

Figure 4. Time wave form of output from HA-S with an input of a 1000 Hz SAM Tone in a background of steady state noise and filtered around 1000Hz. Processed signals for both inactive and active DNR are displayed.

DISCUSSION

Fast DNR and Speech Recognition

The results of this study support a significant increase in speech recognition performance in a background of steady-state noise when subjects were fitted with commercial digital hearing aids using fast DNR time constants. The improvement in subject performance related to fast acting DNR may be attributed to a number of possible factors. Because DNR systems are not expected to impact within band instantaneous SNR, it is speculated (although not directly evaluated) that at least two alternative mechanisms for improved speech recognition in noise may be involved. One potential mechanism is based on a release from temporal masking of the speech stimulus. A second potential source of DNR benefit is a release from upward spread of masking across hearing aid channels. If one treats masking energy within a hearing aid channel as a regional masking effect which spreads basally beyond the initial basilar place of excitation. It could be hypothesized that the masking energy within a channel could compromise the audibility of a speech signal within an adjacent channel. By reducing overall level within a channel dominated by a noise-like signal, audibility may be improved for modulated or speech-like signals in the adjacent higher frequency channel(s) as the spread of masking is reduced by the DNR processing.

Slow DNR and Speech Recognition

The implementation of DNR processing with slow time constants resulted in a significant decrease in performance. While no improvement in speech recognition performance was expected for HA-S fittings, a decrease in performance was unexpected. Electroacoustic evaluation suggested that gain did not reach target levels for speech until 500 msec after the onset of the speech signal (Figure 4). This release period was long enough that it may have compromised individual speech recognition due to a lack of audibility during the period of gain restoration.

This research was funded in part by the Dan Maddox Hearing Aid Research Laboratory.

REFERENCES

[1]Bray, V., Sandridge, S., Newman, C., Kornhass, S. Clinical research findings confirm benefit of advanced signal processing. *Sonic Innovations 2002.*

[2]Bray, V. and Valente, M. Can omni-directional hearing aids improve speech understanding in noise? *Sonic Innovations 2002.*

[3]Nilsson, M., Xiaoling, F., Ghent, R., Murphy, P., Bray, V. Improved Speech Intelligibility in Noise with a Single Microphone Noise Reduction Technique. *Sonic Innovations 2000.*

VANDERBILT KENNEDY CENTER
FOR RESEARCH ON HUMAN DEVELOPMENT