J Am Acad Audiol 28:46–57 (2017)

Evaluation of a Wind Noise Attenuation Algorithm on Subjective Annoyance and Speech-in-Wind Performance

DOI: 10.3766/jaaa.15135

Petri Korhonen*
Francis Kuk*
Eric Seper*
Martin Mørkebjerg†
Majken Roikjer†

Abstract

Background: Wind noise is a common problem reported by hearing aid wearers. The MarkeTrak VIII reported that 42% of hearing aid wearers are not satisfied with the performance of their hearing aids in situations where wind is present.

Purpose: The current study investigated the effect of a new wind noise attenuation (WNA) algorithm on subjective annoyance and speech recognition in the presence of wind.

Research Design: A single-blinded, repeated measures design was used.

Study Sample: Fifteen experienced hearing aid wearers with bilaterally symmetrical (≤10 dB) mild-to-moderate sensorineural hearing loss participated in the study.

Data Collection and Analysis: Subjective rating for wind noise annoyance was measured for wind presented alone from 0° and 290° at wind speeds of 4, 5, 6, 7, and 10 m/sec. Phoneme identification performance was measured using Widex Office of Clinical Amplification Nonsense Syllable Test presented at 60, 65, 70, and 75 dB SPL from 270° in the presence of wind originating from 0° at a speed of 5 m/sec.

Results: The subjective annoyance from wind noise was reduced for wind originating from 0° at wind speeds from 4 to 7 m/sec. The largest improvement in phoneme identification with the WNA algorithm was 48.2% when speech was presented from 270° at 65 dB SPL and the wind originated from 0° azimuth at 5 m/sec.

Conclusion: The WNA algorithm used in this study reduced subjective annoyance for wind speeds ranging from 4 to 7 m/sec. The algorithm was effective in improving speech identification in the presence of wind originating from 0° at 5 m/sec. These results suggest that the WNA algorithm used in the current study could expand the range of real-life situations where a hearing-impaired person can use the hearing aid optimally.

Key Words: assistive listening devices, hearing aids, noise reduction, wind noise

Abbreviations: BTE = behind-the-ear; KEMAR = Knowles Electronic Manikin for Acoustic Research; LMS = least mean squares; RIC = receiver-in-the-canal; SD = standard deviation; SNR = signal-to-noise ratio; SPL = sound pressure level; WNA = wind noise attenuation

INTRODUCTION

Wind noise is encountered by all hearing aid wearers but is especially a problem to those who spend a significant amount of time outdoors. Wind noise is created at a hearing aid microphone when the flow of the air is obstructed as it moves past the hearing aid. The obstructions cause turbulence that results in air pressure fluctuations at the microphone membrane. These changes in air

*Widex Office of Research in Clinical Amplification (ORCA-US), Lisle, IL; †Widex A/S, Lynge, Denmark

Corresponding author: Petri Korhonen, Widex Office of Research in Clinical Amplification (ORCA-US), Lisle, IL 60532; E-mail: ptko@widex.com

pressure are perceived as wind noise. Wind noise can overload the microphone preamplifier resulting in audible distortion and drowning out the acoustic signals that are transduced by the hearing aid microphone. A wind speed of 6 m/sec can result in wind noise levels that exceed the long-term average speech spectrum across most frequencies, and at 12 m/sec may completely saturate the microphone output (Zakis, 2011). The wind noise discussed here differs from the environmental sounds created by wind. When wind passes through other objects in the listening environment (trees, shrubs, car window, etc.) it can produce acoustical waves, which are also transduced by the hearing aid microphone. These sounds are not considered as wind noise here. Such sounds are part of the natural environment and not artifacts caused by wind at the hearing aid. Wind noise affects a substantial number of hearing aid wearers. About 42% of hearing aid wearers are dissatisfied with the performance of their hearing aids in situations where wind is present (Kochkin, 2010). Wind noise can limit the optimal use of the hearing aids outdoors and may negatively impact participation in normal activities. There are two main approaches to combating the problem of wind noise in hearing aids: acoustic modifications and signal processing techniques.

When airflow is obstructed by a physical obstacle on its direct path, the air will attempt to go around the object. If the air velocity is low, the flow around the object is laminar. In laminar flow, the air moves in straight-line layers, and all the particles within a layer have the same velocity. If the air velocity is high, the airflow around the object will generate eddies (swirling of air and reverse current). There is a large amount of mixing of the air particles between the layers, which results in large spatial pressure differentials. These pressure differentials are picked up by the microphone as wind noise.

Mechanical solutions to lower the wind noise aim to reduce the amount of turbulent wind flow that causes acoustical wind noise at the microphone by laminating, redirecting, or diffusing the wind flow. One approach is to add a cover or a hood on top of the microphone to laminate the airflow and to shield the sensing surface of the microphone from the direct airflow (Kates, 2008). The cover laminates the wind by providing a smooth surface for the wind to flow along with fewer eddies. This solution can provide broadband reduction of wind noise up to 18 dB (Widex SUPER brochure; Lynge, Denmark). Another approach is to add a thin piece of foam on top of the microphone preventing the full velocity of the wind from reaching the transducer (Kates, 2008). One disadvantage of a foam windscreen is that the microphone's high-frequency response is attenuated above 10 kHz, depending on the density of the protective layer. While the utility of high frequencies can be insignificant for speech understanding, the broader bandwidth has been associated with better sound quality for hearing-impaired individuals with less than a moderate degree of hearing loss (Ricketts et al, 2008). Dillon et al (1999) measured the wind flow patterns at the ear using laser Doppler velocimeter. They showed that pinna and tragus can act as sources of wind turbulence. Simultaneously, head, pinna, and tragus can also act as wind guards. They showed that the amount of wind turbulence generally reduces as a function of distance away from the head. Thus, the hearing aid form factor can affect how much wind turbulence may be experienced at the microphone opening. In general, custom products such as completely-in-the-canal and in-the-canal devices experience less wind noise than behind-the-ear (BTE) devices because of its microphone placement (Dillon et al, 1999; Zakis, 2011). Some commercial entities take advantage of the position effect by placing the microphone in the small indentation between the crura of the helix and the antihelix of the pinna to minimize turbulence (Kates, 2008). These mechanical approaches have achieved varying degrees of success in combating wind noise.

Another approach to reduce the negative effects of wind noise is to use digital signal processing algorithms. These algorithms use the knowledge of the differences in acoustic characteristics between the desired acoustic signals and wind noise signals to separate these two signals. Wind noise exhibits several important characteristics. It is typically loud, for example, at 12 m/sec wind speed (26.8 mph or 43.2 km/h), typical of fast cycling, the sound pressure level (SPL) can reach as high as 116 dB for some BTE hearing aids (Zakis, 2011). The wind noise level is proportional to the square of wind speed, so doubling the wind speed would increase the turbulence by a factor of four and the wind noise by 12 dB on average (Strasberg, 1988; Kates, 2008). However, the increase in the SPLs observed in the real world is even faster (Morgan and Raspet, 1992). Wind noise level is also dependent on the direction, with the wind originating from the front having the strongest wind noise level (Dillon et al, 1999).

The spectra of wind noise vary widely under different wind situations. Generally speaking, energy from wind noise is concentrated in the low frequencies with a relatively flat spectrum <300 Hz and sloping at a rate of −6 dB per octave >300 Hz (Wuttke, 1991; Dillon et al, 1999; Raspet et al, 2006). The spectrum is also dependent on the wind speed; lower wind speeds generally produce wind noise with energy in the lower frequencies, whereas higher wind speeds produce wind noise with energy in higher frequencies (Brown and Mongeau, 1995; Beard and Nepomuceno, 2001; Chung et al, 2009).

Acoustic signals measured at the two microphones of a dual-microphone hearing aid are typically highly correlated for far-field signals. The signals are similar except for the delay introduced by the finite speed of sound and the spacing between the microphones. Unlike the acoustic signals, a special characteristic of wind noise is that its correlation with itself as measured at two points decreases rapidly with distance (Corcos, 1963; 1964). This is because the turbulences created by wind are unique to each measuring point in space. A consequence is that wind noise created at the two microphones of a dual-microphone hearing aid has independent fluctuations (i.e., the two wind noise signals are largely uncorrelated).

The knowledge on the wind noise spectra, level, and correlation makes it possible to create signal processing solutions that aim to mitigate the annoyance caused by wind noise. The signal processing approaches to wind noise reduction typically consist of two stages: detection of the wind noise, followed by the reduction or attenuation of the wind noise. During the first stage, the algorithms determine if wind noise is present by measuring the degree of correlation of the low- and midfrequency (e.g., <3 kHz) input between the front and the back microphones. Unlike wind noise, external sounds arriving at both microphones of a dual-microphone hearing aid are correlated because the wavelengths of these sounds <10 kHz ($\lambda = 0.343$ m, $c = 343$ m/sec) exceed the physical distance of the two microphones. This difference in correlation between wind noise and the acoustic signals can be used to identify the presence/absence of wind noise. If the two signals at the hearing aid microphones (front and back) are correlated, the signals likely originate from an external sound source. On the other hand, if the signals are uncorrelated, the origin of the signal could be wind noise. In addition, the noise needs to exceed a predetermined level to meet the requirement of wind noise. This predetermined level is unique to each wind noise algorithm implementation.

Various techniques have been attempted to reduce the amount of wind noise after it has been detected. Because wind noise is concentrated primarily in the lower frequencies, gain reduction in the low frequencies can reduce the intensity of wind noise encountered by hearing aids (Bentler and Chiou, 2006). In one implementation, the levels of environmental sounds and wind noise are monitored, and gain at each frequency channel is independently reduced based on the estimated level of the environmental sounds and the level of the wind noise. The goal is to bring the level of the wind noise to the long-term average SPL of the user's environment (Stender and Hielscher, 2011). Unfortunately, no additional details were reported because of its proprietary nature.

Wireless technologies have also been used in wind noise reduction in bilateral hearing aid fittings. In one such implementation (Latzel and Appleton, 2013a), the two hearing aids of the bilateral pair monitor the presence of wind and share such information between the two instruments. When wind noise is detected, the low-frequency part (<1.5 kHz) of the microphone signal from the ear where more wind noise is present is substituted with the low-frequency part of the microphone signal from the other ear where less wind noise is present. The rationale of replacing only the lower frequencies is that wind noise is typically in the lower frequencies at a mild-to-moderate speed. Latzel and Appleton (2013b) reported 27% improvement in speech scores (Oldenburg Sentence Test, OLSA) using this feature on listeners with a moderate hearing loss. In their investigation, the wind was generated using a fan located at the right side (60°) at a speed of 3.5–4 m/sec, and speech was presented from the opposite side (270°) at 65 dB SPL. Details on the type of fan or of the wind flow patterns used in their study were not specified.

Wind noise has an additional impact on hearing aids with directional microphones. Directionality in hearing aids is achieved by subtracting the signal at the rear microphone from the signal at the front microphone. For environmental acoustic signals, the correlation at the two microphone locations can be high, especially in the low frequencies. Consequently, the subtraction method used to achieve directionality partially cancels the low frequencies of the desired signal. In many directional microphone designs, this low-frequency loss is compensated by boosting the low-frequency output of the directional microphone by 6 dB per octave (Kates, 2008). This compensates for the potential loss of loudness for sounds presented from the front. Unfortunately, when combining uncorrelated signals (such as in wind noise), the signal power always increases irrespective of the relative phases of the signals (i.e., summation and subtraction have the same effect). As a result, the low-frequency boost used in the directional microphones further increases the level of the unwanted wind noise when compared to an omnidirectional microphone where no compensation is needed. One way to combat the negative effects of wind noise in a directional hearing aid is to use a fully adaptive directional microphone that automatically switches to an omnidirectional microphone when wind is detected (Kuk et al, 2005). Alternatively, since the wind noise is located primarily in the lower frequencies, the use of an omnidirectional microphone mode in the lower frequencies and a directional microphone mode in the higher frequencies may be used as a strategy to alleviate the wind noise problem (Stender and Hielscher, 2011). Such a microphone configuration is possible in a multichannel adaptive directional microphone system that can adjust the directional pattern of the microphone independently in each frequency channel based on the acoustic characteristics of the input environment (Kuk et al, 2005).

Recently, Widex introduced a patented wind noise attenuation (WNA) algorithm that is based on least mean squares (LMS) filtering. LMS algorithm is a simple and effective algorithm for adaptive filter design used to mimic a desired filter by finding the filter coefficients that produce the LMS of the difference between the desired and the actual signal. The LMS algorithm attempts to find the optimum filter by updating each filter coefficient iteratively in the direction of the instantaneous gradient of the squared error signal with respect to the coefficient in question. When the desired and the actual signals correlate, the algorithm is capable of finding a set of filter coefficients that minimize the squared error. Conversely, the algorithm is not capable of finding filter coefficients that would minimize the difference between the two uncorrelated signals. Therefore, LMS filter is known to reduce uncorrelated noise (Haykin, 2014).

LMS filters are applicable in situations when some parameters of the desired processing operation are correlative but are not known in advance or are changing over time. Such may be the case in wind noise management. Speech may be the correlative desirable signals and wind is the uncorrelative, undesirable signals. This WNA algorithm uses correlation between the signals at the two hearing aid microphones to filter out the noise.

Once the input is determined to be wind noise, the wind noise–contaminated signals from both microphones are used to estimate the desirable wind-free signal. Because only the desired signal is correlated at the two microphones, the adaptive filter is capable of estimating only the desired portion of the signal. Thus, the estimation of the wind noise is zero, and the output of the adaptive filter consists of only the desired signal. The output of the adaptive filter (i.e., estimate of the desired signal) is used as the input to the hearing aid amplifier.

The current study evaluated the effect of the new WNA algorithm using a single-blinded repeated measures design. First, listeners' subjective impressions on wind noise annoyance were measured with and without the algorithm. The listeners' speech-in-wind performance at a wind speed of 5 m/sec was measured at several speech levels in a controlled laboratory environment to examine how the attenuation of wind noise affects speech intelligibility.

METHODS

Participants

A pilot study was conducted on six individuals using the same test procedures as the actual test. On the basis of their results on the speech test, we estimated that the sample size required for a significant improvement with a power >0.8 ranged from 3 to 15 across all test conditions. Thus, a sample size of 15 was selected in the actual study.

Fifteen adults (eight females and seven males) with bilaterally symmetrical (±10 dB) sensorineural hearing loss participated. The averaged four-frequency (0.5, 1, 2, and 4 kHz) pure-tone averages were 42.3 dB HL (standard deviation [SD] = 10.0 dB) for the right ear and 42.5 dB HL (SD = 7.3 dB) for the left ear (see Figure 1). All participants were native English speakers. Their ages ranged from 31 to 82 yr with a mean age of 70.1 yr (SD = 13.8 yr). On average, the participants had worn hearing aids for 9.5 yr (SD = 7.8 yr). Nine of the participants wore receiver-in-the-canal (RIC), and three in-the-ear hearing aids as their own hearing aids. Three participants did not own or wear hearing aids regularly. Participants were informed of the purpose of the study, benefits, and risks before their participation. All participants signed informed consent and were financially compensated for their participation.

Hearing Aids

Hearing aids used in the study were Widex UNIQUE Fusion 440 RIC BTE hearing aids (P-receiver). The hearing aids were programmed using the Compass GPS fitting software (version 2.0.411.0). This hearing aid is a 15-channel-wide dynamic range compression hearing aid with a compression threshold as low as 0 dB HL. The sampling frequency of the analog-to-digital stage is 33 kHz with an input resolution of 18 bits. The input limit before saturation is 113 dB SPL. The frequency response of this instrument ranges from 100 to 6400 Hz (ANSI, 2009). The maximum power output of this instrument is 121 dB SPL. This instrument

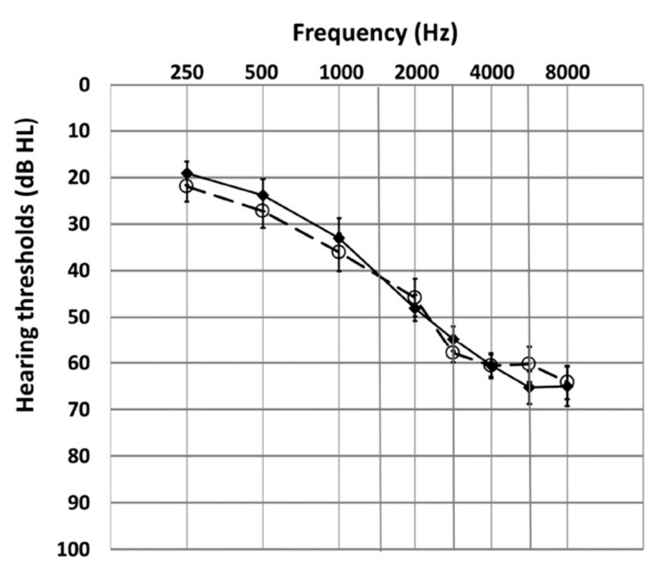

Figure 1. Averaged audiometric thresholds for left (solid line) and right (dashed line) ears. Error bars indicate ±1SD.

includes an active feedback cancellation algorithm, two methods of digital noise reduction, and an environmental classifier system that adaptively optimizes the processing parameters based on the listening environment. The noise reduction algorithms and classifier were disabled during this study. An omnidirectional microphone was used to minimize any potential processing changes as a result of hearing aid orientation. Hearing aids were programmed using the average hearing loss from all participants and the default frequency-gain setting.

The WNA algorithm is designed to reduce the acoustic consequence of wind turbulence while maintaining a good sound quality for other sounds. This algorithm operates in two stages. In the detection stage, the algorithm uses correlation of signals at the two microphones of the dual-microphone system, the frequency spectrum of the input signal, and the energy level of the input signal to make a decision on the presence (or absence) of wind noise. For the input to be classified as wind noise, inputs at the two microphones must be uncorrelated. In addition, its spectrum must be primarily in the low frequencies, and its intensity level has to be >40 dB SPL. The algorithm proceeds to the attenuation stage only if all three criteria are met.

Adaptive filtering using LMS is used to reduce wind noise levels. To illustrate the steps, let us denote the desired acoustic sounds at the two microphones with s_1 and s_2, and the unwanted wind noise with w_1 and w_2. The two signals y_1 and y_2 entering the hearing aids are the sum of the acoustic signal and the wind noise, that is, $y_1 = s_1 + w_1$ and $y_2 = s_2 + w_2$. Because the distance between the two microphones is much smaller (~16 mm) than the distance between the sound sources in the environment and the hearing aid microphones, we can assume a far-field model for the acoustic sounds. In a far field, the SPL does not vary significantly with the small changes in position. Consequently, the desired acoustic signals s_1 and s_2 picked up by the two microphones are highly correlated within the bandwidth of interest (<16 kHz). This is in contrast to the wind noise signals w_1 and w_2, which are highly uncorrelated. The algorithm uses an adaptive filter $H(z)$ to alter one of the microphone inputs (Figure 2). The parameters of this adaptive filter are determined in an iterative manner in an attempt to minimize the difference between the signals y_1 and y_2 (see u in Figure 2). Adaptive filter can only predict the part of the signal y_2 that is correlated with y_1 (i.e., the desired signal). Because the two wind signals w_1 and w_2 are uncorrelated, they are left out of the predictor output $\hat{s}$. Coefficients of $H(z)$ are recursively calculated such that the mean-squared difference u is minimized. The wind noise reduction algorithm works for frequency bands up to and including the band at 3.2 kHz. The study aid also included a microphone cover that shields the microphone from the direct wind.

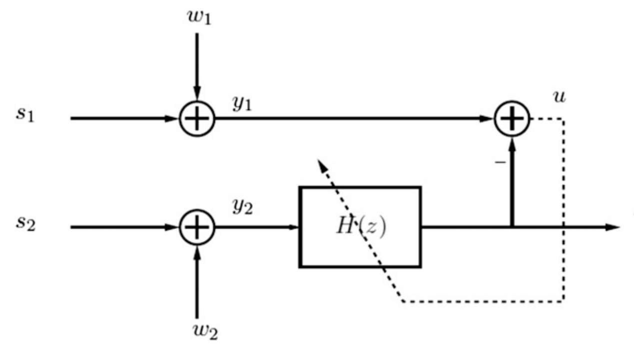

Figure 2. Block diagram displaying the adaptive filtering used to reduce wind noise in the study hearing aid.

This microphone cover provides broadband reduction of wind noise by up to 18 dB (Widex SUPER brochure).

Stimuli

All the stimuli used in the current study were prerecorded in a wind tunnel and presented via insert earphones during the data collection. Use of prerecorded stimuli allowed us to control for the wind characteristics across different hearing aid processing conditions. The prerecording was carried out in a wind tunnel at G.R.A.S. Sound & Vibration A/S in Holte, Denmark. This wind tunnel is an open-return design consisting of a duct with a 0.63-m diameter and a 0.315-m-wide exhaust. This tunnel uses a 4.4-kW fan rotating at 2,044 rpm (maximum). It is capable of producing wind speeds up to 10 m/sec.

The recordings were carried out using a Knowles Electronic Manikin for Acoustic Research (KEMAR; Holte, Denmark) head and torso simulator placed at the exhaust of the wind tunnel. The KEMAR ears were at a distance of 0.38 m from the exhaust outlet and 0.655 m from the floor. A loudspeaker presenting the speech stimuli (G.R.A.S. 44AA Mouth Simulator) was also placed at 270° azimuth 0.655 m height from the floor at a distance of 1 m from the center of the KEMAR head. In the aided condition, the study hearing aid was coupled to KEMAR's left ear using a fully occluding earmold. Output from KEMAR [RA0045 Ear Simulator IEC 60318-4 (60711)] was recorded using Tascam DR-680 portable multitrack recorder (Tascam, Montebello, CA) with 44.1-kHz sampling frequency. The wind speed and speech level was calibrated at the location of the KEMAR head in the absence of the head using an air velocity meter (TSI VelociCheck 8330-M-GB; Shoreview, MN) and integrating sound level meter (Brüel & Kjær 2238 Mediator; Nærum, Denmark). A 1-kHz calibration tone was recorded that allowed us to present the stimuli at the same level as was recorded in the wind tunnel.

Subjective annoyance rating for wind noise was obtained for wind presented alone without speech. Wind was presented at wind speeds of 4, 5, 6, 7, and 10 m/sec with wind originating from 0° or 290°. The angle of 290° was selected over the 270° (directly to the left) because the 290° angle was rated to result in more wind noise based on an informal listening test carried out during the recordings. Each wind noise sample was 20 sec in duration.

Objective phoneme identification in the presence of wind was measured using the Widex Office of Research in Clinical Amplification Nonsense Syllable Test (Kuk et al, 2010). This is an open-set consonant-vowel-consonant-vowel-consonant test containing 25 English consonants each appearing at least once in the initial, medial, and final word positions unless prohibited by phonotactic constraints. A shortened 32-item female version of the test was used in the current study. Participants verbally repeated what they thought they heard. The test administrator listened to the responses and scored the participants' responses phonemically using a custom software. The stimulus conditions included speech presented at 60, 65, 70, and 75 dB SPL from 270° with the wind originating from 0° at a wind speed of 5 m/sec. This wind speed was selected because it occurs frequently in real life during outdoors and leisure activities.

Procedure

During data collection the participants were seated in a 3 × 3 × 2-m ($W \times L \times H$) audiometric test booth (IAC Acoustics, Aurora, IL). The stimuli were presented to each participant's left ear using an insert earphone with a fully occluding foam insert (Eartone 3E; Aearo Company, Indianapolis, IN). The stimuli were generated at 44.1-kHz sampling frequency using Echo Audio Gina 24/96 sound card (Santa Barbara, CA). The presentation level was controlled by routing the signal through an audiometer (GSI 61; Grason-Stadler, Eden Prairie, MN). All the stimuli were presented at the same level that was measured during the recording of the stimuli in the wind tunnel. The correct calibration levels were verified using a sound level meter (Quest 1800; Quest Technologies, Oconomowoc, WI).

The subjective rating of annoyance was measured using a rating scale from 1 to 7 with a smaller number indicating less annoyance. Specifically the intervals were 1 = not noticeable (and thus not annoying); 2 = slightly noticeable, but not annoying; 3 = somewhat noticeable, but not annoying; 4 = slightly annoying; 5 = somewhat annoying; 6 = very annoying; and 7 = extremely annoying. The participants indicated their rating for each sample on a touch screen. Order of presentation across wind speeds and processing conditions were randomized.

Participants indicated their responses for the annoyance rating on the touch screen computer. For the speech test, the participants indicated their responses verbally. The test administrator used a custom software to score the responses. All test conditions were presented in a counterbalanced order using a single-blinded design.

RESULTS

Acoustic Analysis

The acoustic effect of the WNA was evaluated by analyzing the output of the hearing aid between the WNA-ON and WNA-OFF conditions. The spectrum of the wind was obtained with fast Fourier transform using a window length of 4,096 points and Blackman windowing and averaging across the duration of the whole wind noise recording for each condition. The ⅓-octave band levels were obtained by averaging over the fast Fourier transform values at each frequency region. The level of the spectrum was normalized to known SPL of the calibration tone. Figure 3A–E displays the output in dB SPL for wind alone from 0° at 4, 5, 6, 7, and 10 m/sec for the WNA-OFF and WNA-ON conditions. Figure 3F shows the difference in spectra between the WNA-OFF and WNA-ON conditions. The overall dB SPL for wind from 0° and 290° azimuths were tabulated in Table 1. The level of the output (dB SPL; Table 1) ranged from 81 to 109 dB SPL with greater levels of wind noise for wind from 0° than from 290°. Levels were higher with WNA-OFF than WNA-ON, except at 4 and 5 m/sec wind speeds when wind originated from 290°. The greatest effect of WNA was measured at frequencies between 315 and 2500 Hz (Figure 3). The greatest attenuation (14.6 dB) was measured at 500 Hz when the wind speed was 4 m/sec. The maximum attenuation was measured at 1 kHz at wind speeds 5, 6 and 7 m/sec. The amount of attenuation was 17.2, 16.3, and 16.5 dB at wind speeds of 5, 6, and 7 m/sec, respectively. At 10 m/sec wind speed, the maximum attenuation (18.3 dB) was measured at 2000 Hz. The amount of attenuation provided by the WNA algorithm was the greatest when wind speed was 10 m/sec.

Subjective Rating of Annoyance

Figure 4 displays the median annoyance ratings for wind presented alone from an azimuth of 0°. The annoyance ratings were consistently higher with WNA-OFF than with WNA-ON (i.e., more annoying with WNA-OFF). The difference in median ratings between the WNA-ON and WNA-OFF conditions was the greatest at a wind speed of 6 m/sec where the rating changed from 7 (extremely annoying) with WNA-OFF to 5 (somewhat annoying) with WNA-ON. In addition, the median rating of 7 (extremely annoying) was reached at a wind speed of 6 m/sec with WNA-OFF. The same rating was not reported until at a wind speed of 10 m/sec when WNA was activated (WNR-ON). The annoyance ratings were significantly higher with WNA-OFF than with WNA-ON for wind speeds 4, 5, 6, and 7 m/sec (Z = 3.06, 2.23, 3.14, and 2.65, respectively; $p < 0.05$) using a Wilcoxon signed-rank test. There was no significant

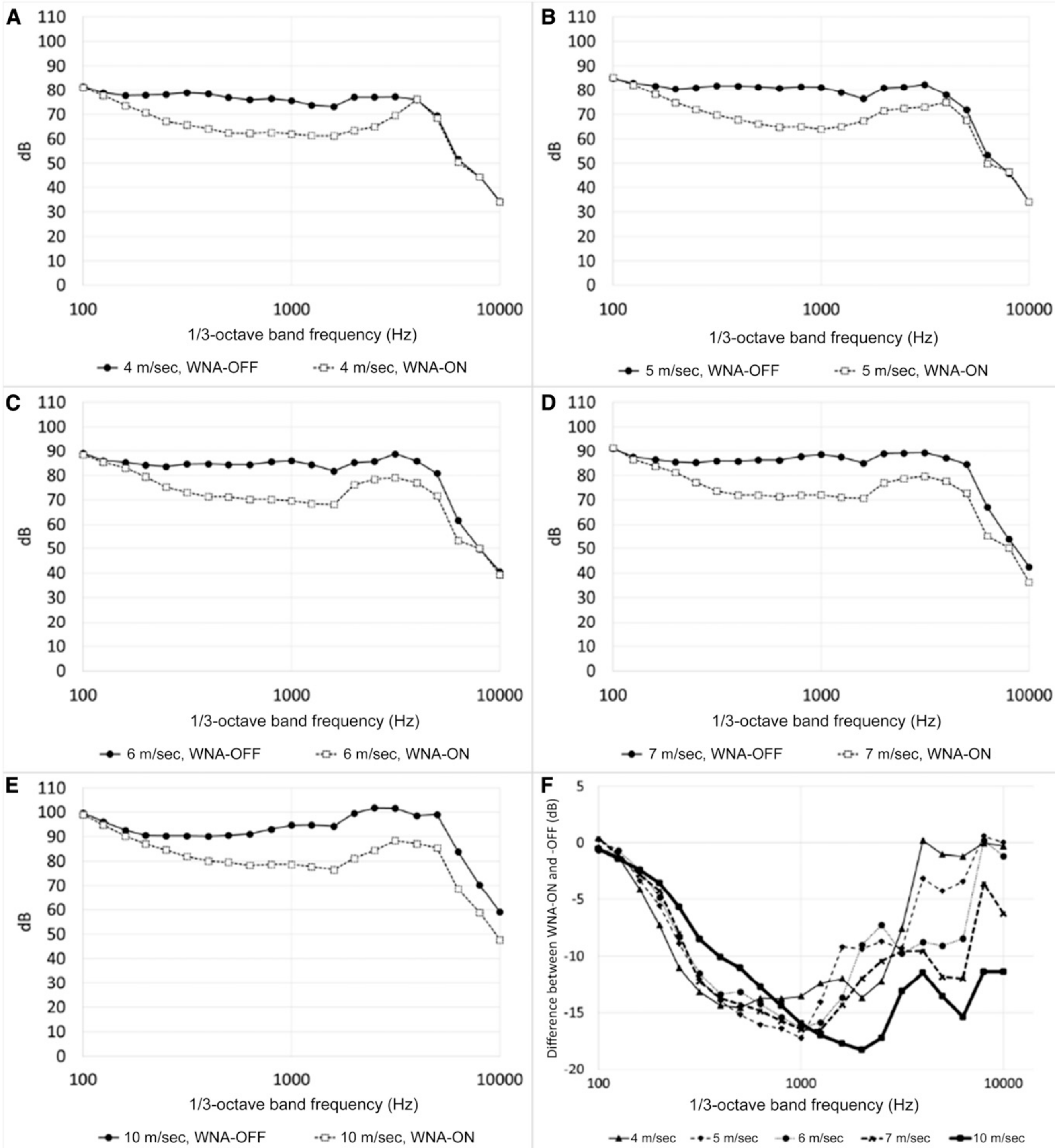

Figure 3. Output of the hearing aid in dB SPL at ⅓-octave bands for wind coming from an azimuth of 0° with WNA-OFF (solid) and WNA-ON (dashed) at (A) 4 m/sec, (B) 5 m/sec, (C) 6 m/sec, (D) 7 m/sec, and (E) 10 m/sec. Levels are normalized to a known SPL calibration tone. (F) Difference between the WNA-OFF and WNA-ON conditions at each wind speed.

difference in ratings between WNA-ON and WNA-OFF ($p > 0.05$) at the wind speed of 10 m/sec.

Figure 5 displays the participants' median annoyance ratings when wind was presented from 290°. Again, the annoyance rating was consistently higher with WNA-OFF than with WNA-ON (i.e., more annoying with WNA-OFF). The difference between the WNA-ON and WNA-OFF when the wind speed was between 6 and 7 m/sec (difference of 1 point) was statistically significant ($Z = 2.06$ and 3.19, respectively; $p < 0.05$). There was no significant difference in annoyance ratings between WNA-ON and WNA-OFF for wind speeds

Table 1. Wind Noise Levels (dB SPL, C, Slow) Measured at the Output of the Hearing Aid with WNA-ON and WNA-OFF for Wind Azimuths 0° and 290°

Wind Speed (m/sec)	Wind Noise Levels (dB SPL)			
	Wind Azimuth 0°		Wind Azimuth 290°	
	WNA-OFF	WNA-ON	WNA-OFF	WNA-ON
4	89	83	81	81
5	93	86	82	82
6	99	90	93	87
7	101	92	99	89
10	109	98	108	98

Note: C = C-weighting.

at 4, 5, and 10 m/sec ($p > 0.05$). Unlike wind from 0°, subjective rating did not reach ceiling with WNA-OFF at a wind speed of 6 m/sec, suggesting that wind from the side was less annoying than wind from the front.

Speech-in-Wind Performance

Figure 6 displayed a scatterplot showing phoneme identification scores (in %) of each participant with WNA-ON on the abscissa and WNA-OFF on the ordinate. At a speech level of 60 and 65 dB SPL, the performance with WNA-OFF was close to zero. Only at higher speech levels (70 and 75 dB SPL) were the participants able to identify some speech with WNA-OFF. With WNA-ON all participants were able to perform >0% level even when speech was presented at the softest level of 60 dB SPL. The average phoneme identification scores with WNA-ON were 34.4%, 49.6%, 64.4%, and 68.1% for 60, 65, 70, and 75 dB SPL speech levels, respectively, and with WNA-OFF were 0%, 1.4%, 31.4%, and 57.8% for 60, 65, 70, and 75 dB SPL speech levels, respectively (Figure 7). The difference in performance between WNA-ON and WNA-OFF was statistically significant at all speech levels ($p < 0.01$).

Because the phoneme identification scores were measured at multiple speech levels, we could estimate the signal-to-noise ratio (SNR) benefit that the WNA algorithm provides at a given performance level. Figure 7 illustrates how the SNR benefit estimate was obtained. First, the performance-intensity functions measured with WNA-OFF and WNA-ON were generated by connecting the measured phoneme scores at 60, 65, 70, and 75 dB SPL presentation levels. The horizontal distance between these two functions at a particular performance level (e.g., say 50%) reflected the signal level difference that resulted from the WNA algorithm. For phonemes, the SNR benefit when using the WNA feature was 8.39 dB at the 50% speech performance level.

DISCUSSION

The current study showed that the WNA algorithm reduced the amount of wind noise at the hearing aid output by 13 to 18 dB at 1 kHz when wind originated from an azimuth of 0° at speeds ranging from 4 to 7 m/sec, and by as much as 19 dB at 2200 Hz at a wind speed of 10 m/sec. This attenuation resulted in reduced subjective annoyance for wind and improved speech identification performance at a speed of 5 m/sec. The benefits reported were based on adaptive filtering using inputs from the two microphones to derive an estimate of the correlated input signal. Because of the uniqueness of this feature, the results of the current study may not be generalized to other wind noise reduction algorithms that use other strategies.

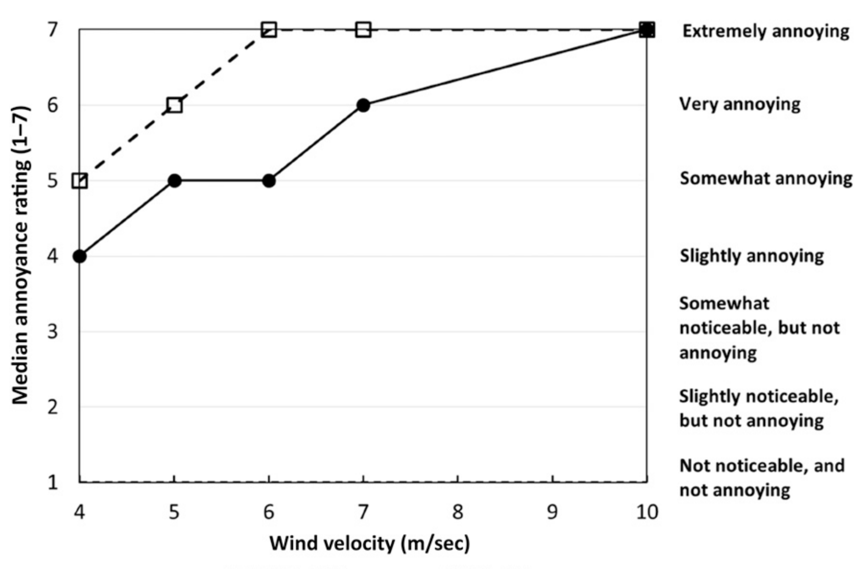

Figure 4. Median listener rating of annoyance (1–7) when the wind was presented alone from 0° at speeds 4, 5, 6, 7, and 10 m/sec.

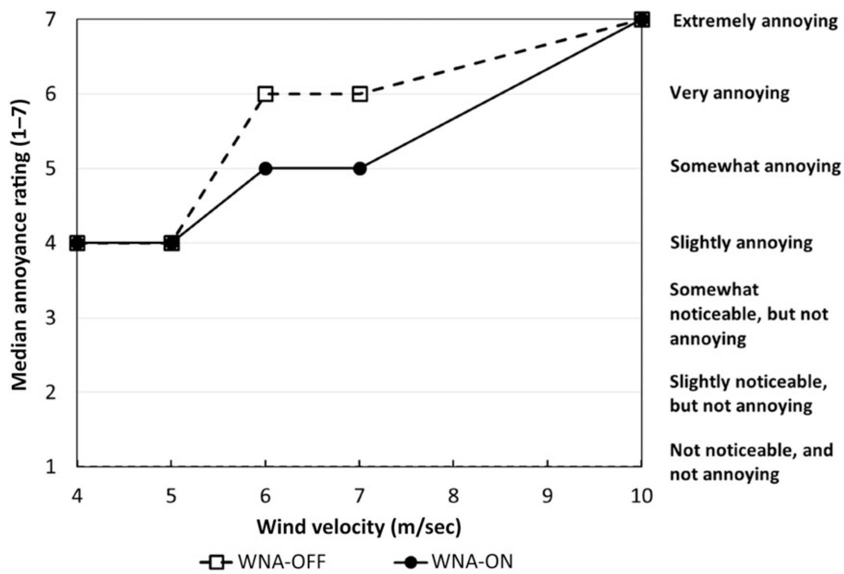

Figure 5. Median listener ratings of annoyance (1–7) when the wind was presented alone from 290° at speeds 4, 5, 6, 7, and 10 m/sec.

Wind noise algorithms based on bilateral level differences between the two ears would likely not provide benefit when wind originates from 0°. This is because when wind originates from the front, there is no significant difference in the wind noise levels between the two ears. Thus, the results seen in the Latzel and Appleton (2013b) study that evaluated a binaural speech in wind feature may not be evident under the current test conditions. Unlike the current study, participants in the Latzel and Appleton (2013b) study were able to identify speech at 65 dB SPL speech with >50% accuracy even without the use of wind noise reduction algorithm. We can identify at least two reasons for the difference in reported performance between the two studies. First, the lower wind speed used in the Latzel and Appleton (2013b) study likely resulted in lower wind noise level than in the current study. Wind speed has been shown to be a significant factor in the overall level of wind noise (Zakis, 2011). Lower wind noise level would result in more favorable SNR when using a fixed speech level. Second, because in the Latzel and Appleton (2013b) study the wind originated from 60°, the amount of wind at the opposite ear may have been lessened by the head shadow, allowing the listener to use the ear with better SNR for speech identification. The acoustic analysis showed that the current algorithm lowered the wind noise level and reduced the subjective annoyance when wind originated from the side (290°). Furthermore, we showed that the WNA algorithm improved speech identification performance. Therefore, we expect that the WNA algorithm used in the current study would also be beneficial for the conditions used in the Latzel and Appleton (2013b) study but not vice versa.

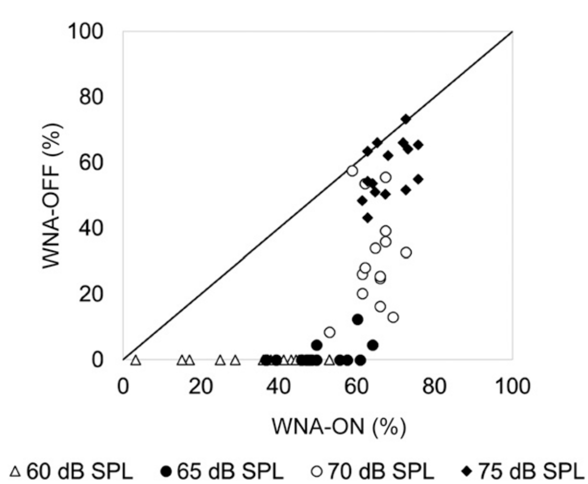

Figure 6. Phoneme identification performance for each of the 15 participants with and without the WNA algorithm. Wind originated from 0° at 5 m/sec speed. Speech was presented at 60, 65, 70, and 75 dB SPL from 270°.

One may wonder if traditional noise reduction approaches such as Wiener filtering and spectral subtraction methods, which have been used for acoustic noise reduction (Kuk et al, 2002; Dillon, 2012), may be suitable for wind noise reduction. Acoustic noise reduction algorithms take advantage of the spectral differences of speech and noise to suppress unwanted noise. Wiener filtering algorithm attempts to minimize the mean-squared error between the desired input and the filtered output. The spectral subtraction algorithm estimates the noise spectrum, and then subtracts it from the noisy speech spectrum to get an improved estimate of the original speech. Both of these methods obtain an estimate of the noise during the pauses in the desired signal using a voice activity detector. The primary problem is that they assume stationary signals. However,

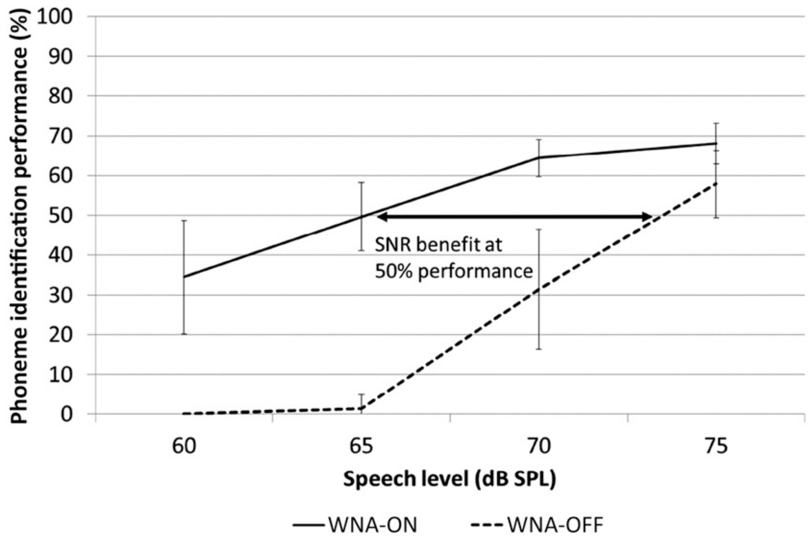

Figure 7. Phoneme identification performance averaged across all participants for WNA-ON and WNA-OFF conditions when speech was presented from 270° at 60, 65, 70, and 75 dB SPL and wind originated from 0° at 5 m/sec speed. The error bars represent ±1SD. The arrow highlighting the horizontal distance between the performance functions represents the SNR benefit at 50% speech performance level.

wind noise is typically nonstationary and constantly changing in spectrum. Thus, the noise estimates obtained during the speech pauses would not be valid if and when the wind characteristics change during the speech segment. In the adaptive filter–based algorithm used in the current study, the noise detection and the noise estimation are inexplicitly built into one algorithm. Therefore, there are no separate circuits dedicated for noise estimation. This means it can operate even in nonstationary wind noise, because it constantly adapts to the wind noise, even during speech. Second, the current algorithm attenuates only the uncorrelated wind noise and spares the acoustic signals from gain reduction. In spectral-shaping-based methods, all sounds (correlated and uncorrelated) in a given frequency region are reduced in level by the same extent based on the gain reduction applied to that frequency region. Consequently, while the overall output level of the hearing aid may be reduced, the unfavorable SNR at each frequency channel resulting from wind noise remains. These approaches may be effective in improving listening comfort in windy situations, but they may have limited effectiveness in improving speech-in-wind performance. The current algorithm does not reduce the level of desired speech sounds (correlated) while reducing the level of the unwanted wind noise (uncorrelated).

Participants consistently rated the annoyance from wind noise higher with WNA-OFF than with WNA-ON. The greatest change in annoyance was measured when the wind originated from 0° at a speed of 6 m/sec. In this condition, the use of the WNA algorithm changed the median subjective rating from “extremely annoying” to “somewhat annoying.” For the highest tested wind speed (10 m/sec), there was no difference in the perceived annoyance between WNA-ON and WNA-OFF despite the results of the acoustic analysis showing that the WNA algorithm reduced the wind noise by 18.3 dB at 2200 Hz at a wind speed of 10 m/sec. This suggests that there is an upper limit at which the WNA feature is “perceived” to be beneficial. This upper limit may be set by the absolute level of wind noise measured at the output of the hearing aid, for example, the level of wind noise at 10 m/sec was 98 dB SPL in the WNA-ON condition. This is comparable to the WNA-OFF condition at 6 m/sec, which was 99 dB SPL. In both cases, participants rated the wind noise as “extremely annoying.” Unless the WNA algorithm can attenuate even more noise levels at higher wind speeds, its use will be limited to moderately windy situations only.

The current data set also allowed us to estimate the subjective annoyance associated with the level of in situ wind noise by combining the subjective ratings (Figures 4 and 5) with the measured dB SPL levels of noise (Table 1). The wind noise levels at the hearing aid output, the wind speeds required to generate these wind level with WNA-OFF and WNA-ON, and the associated subjective annoyance ratings are shown in Table 2. For example, when the output wind noise level was between 81 and 83 dB SPL, the annoyance rating was “slightly annoying.” When the output wind noise level was between 86 and 90 dB SPL, the annoyance rating was “somewhat annoying.” This confirms that the annoyance rating was driven by the absolute SPL of wind noise at the hearing aid output. The absolute SPL of the wind noise at the hearing aid output was influenced by the processing condition, wind speed, and wind direction with generally lower wind noise level with

Table 2. Measured Wind Noise Levels at Hearing Aid Output, Associated Subjective Annoyance Ratings, and Wind Conditions that Generated These Levels with WNA-OFF and WNA-ON

		Wind Speed (m/sec)			
		Wind from 0°		Wind from 290°	
Wind Noise Level, Output (dB SPL)	Subjective Annoyance Rating	WNA-OFF	WNA-ON	WNA-OFF	WNA-ON
81–83	Slightly annoying	—	4	4–5	4–5
86–90	Somewhat annoying	4	5–6	—	6–7
92–93	Very annoying	5	7	6	—
≥98	Very/extremely annoying	6–10	10	7–10	10

WNA-ON than with WNA-OFF for same wind conditions. It should be noted that, since the study aid included a protective microphone cover, the wind speeds required to produce the output levels and subjective annoyance ratings listed in Table 2 would likely to be lower with a hearing aid that does not have a microphone cover. Or, if the wind speed is fixed, the hearing aid without a microphone cover would result in worse subjective annoyance rating than listed in Table 2.

While WNA strategies may reduce the annoyance caused by wind noise through gain reduction, they could negatively affect the perception of desirable sounds, such as speech if the gain for both desirable sounds and undesirable wind noise is reduced. A merit of the current LMS filtering–based approach to WNA is the potential that the uncorrelated wind noise could be reduced without affecting the integrity of the correlated speech sounds. Thus, we measured phoneme and word identification performance in the presence of wind noise. The results demonstrated that the use of the WNA algorithm improved speech identification performance for a range of wind speeds. When wind originated from an azimuth of 0° at 5 m/sec the wind turbulence was loud enough to mask the 60 and 65 dB SPL speech almost completely in the WNA-OFF condition. When using the WNA algorithm, the phoneme identification accuracy improved to 34.4% for 60 dB SPL speech and 49.6% for 65 dB SPL speech. At this wind speed, the hearing aid wearers provided a median annoyance rating of "very annoying" when not using the WNA algorithm. This could lead some hearing aid wearers to turn off or mute their hearing aids completely to ensure a comfortable listening experience. Some may not even want to use their hearing aids in situations where wind may be expected. In so doing, their speech intelligibility could be compromised greatly. Considering that a majority of hearing aids dispensed today are BTE and RIC styles, (Strom, 2013) which are most prone to wind turbulence (Dillon et al, 1999), the use of an effective WNA algorithm would seem to be most appropriate.

A reduction of the wind noise level increased the speech-to-wind ratio because the wind noise level at the hearing aid input was fixed. This enabled the participants to hear some of the speech sounds at lower speech levels with the WNA algorithm. One would expect the difference between WNA-ON and WNA-OFF to reduce as the speech input levels increased. This was indeed the case when the benefit decreased from ~50% at a speech level of 65 dB SPL to only 10% at the 75 dB SPL speech level. This does not mean that a higher speech level does not yield speech benefits. On the contrary, a higher speech input may show comparable increase in speech intelligibility at a wind speed >5 m/sec used in this study. The acoustic analysis showed that the WNA algorithm was successful in reducing the amount of wind noise even at higher wind velocities (6, 7, and 10 m/sec). And at such higher wind speeds, it is also normal for the talkers to raise their voices to result in a higher speech level for the listeners. This suggests that the WNA algorithm may potentially be effective in providing speech-in-wind benefit even at higher wind speeds than that used in the current study.

In the current study, the speech performance was measured with wind originating from the front (0° azimuth) and speech presented from the side (270° azimuth). Some may question if the same results would be obtained if both speech and wind originated from the same (or similar) direction. There are both theoretical and practical reasons for the choice of the current test conditions. Theoretically, wind noise is a result of the turbulence at the hearing aid microphone. As such, it is not an external sound, and thus it does not have a direction per se, even though the wind flow that causes the turbulence has a direction. In fact, this property of wind noise being spatially uncorrelated is exploited in the detection and attenuation of the wind noise in the present WNA algorithm. Thus, we would expect the WNA algorithm to improve speech-in-wind performance even if speech originated from the same direction as the wind that generated the wind noise. The current test condition was chosen for two reasons. First, this is likely the most common wind scenario hearing aid wearers experience in the real world. This scenario corresponds to walking with someone outdoors side by side and talking to that person. A second practical reason was that positioning the loudspeaker in the front of the wind tunnel outlet would have restricted the wind flow directed toward the KEMAR manikin. For these reasons, we choose the wind front, speech side test condition used in this study.

It should be noted that the wind speed observed in real life is the vector sum of the true wind and the headwind experienced in still air due to physical movement. This phenomenon is called "apparent wind." For example, a hearing aid wearer walking at 1 m/sec against a 3 m/sec wind would result in 4 m/sec apparent wind at the hearing aid. Alternatively, a hearing aid wearer jogging at 3 m/sec with 3 m/sec downwind would not experience any wind assuming the running direction being identical to wind direction. Therefore, the range of wind conditions that the WNA algorithm would be beneficial may differ in real-life situations based on the movement of the hearing aid wearer. Notably, the hearing aid wearer in movement could be experiencing wind noise even in calm environmental wind conditions. A special example of such situation is exercising indoors (running, playing tennis, or basketball).

Wind noise is encountered by all hearing aid wearers especially those who spend time outdoors. Wind can therefore limit environments in which the hearing aid can perform satisfactorily. The LMS-based wind noise algorithm in the current study was demonstrated to reduce annoyance for a range of wind speeds while simultaneously improving speech intelligibility. The current WNA algorithm could therefore seamlessly expand the range of real-life situations where a hearing aid wearer can listen comfortably. This can promote consistent use of the hearing aid and promote effortless hearing. Because of the potential benefits that this feature may offer, its inclusion in hearing aids as a necessary feature should be considered.

REFERENCES

American National Standards Institute (ANSI). (2009) *Specification of Hearing Aid Characteristics*. ANSI S3.22-2009. New York, NY: ANSI.

Beard J, Nepomuceno H. (2001) *Wind Noise Levels for an ITE Hearing Aid*. Knowles Engineering Report No. 128, Revision A. Itasca, IL: Knowles Electronics.

Bentler R, Chiou L-K. (2006) Digital noise reduction: an overview. *Trends Amplif* 10(2):67–82.

Brown DV, Mongeau LG. (1995) *The Design, Construction, and Validation of a Small, Low-Speed, Quiet Wind Tunnel with Application to Noise from the Flow over a Cavity*. Herrick Laboratories Internal Report No. 204. West Lafayette, IN: Purdue University, HL95–HL99.

Chung K, Mongeau L, McKibben N. (2009) Wind noise in hearing aids with directional and omnidirectional microphones: polar characteristics of behind-the-ear hearing aids. *J Acoust Soc Am* 125(4):2243–2259.

Corcos GM. (1963) Resolution of pressure in turbulence. *J Acoust Soc Am* 35:192–199.

Corcos GM. (1964) The structure of the turbulence field in boundary-layer flows. *J Fluid Mech* 18:353–378.

Dillon H. (2012) *Hearing Aids*. 2nd ed. Turramurra, Australia: Thieme, Boomerang Press.

Dillon H, Roe I, Katsch R. (1999) Wind Noise in Hearing Aids: Mechanisms and Measurements. *Report*. Chatswood, Australia: National Acoustics Laboratories.

Haykin S. (2014) *Adaptive Filter Theory*. Pearson, Upper Saddle River, NJ.

Kates J. (2008) *Digital Hearing Aids*. San Diego, CA: Plural Publishing.

Kochkin S. (2010) MarkeTrak VIII: consumer satisfaction with hearing aids is slowly increasing. *Hear J* 63(1):19–27.

Kuk F, Keenan D, Lau C-C, Ludvigsen C. (2005) Performance of a fully adaptive directional microphone to signals presented from various azimuths. *J Am Acad Audiol* 16(6):333–347.

Kuk F, Lau CC, Korhonen P, Crose B, Peeters H, Keenan D. (2010) Development of the ORCA nonsense syllable test. *Ear Hear* 31(6): 779–795.

Kuk F, Ludvigsen C, Paludan-Muller C. (2002) Improving hearing aid performance in noise: challenges and strategies. *Hear J* 55(4): 34–46.

Latzel M, Appleton J. (2013a) Evaluation of a binaural speech in wind feature, Part 1: verification in the laboratory. *Hear Rev* 8: 32–34.

Latzel M, Appleton J. (2013b) Evaluation of a binaural speech in wind feature, Part 2: validation and real-life benefit. *Hear Rev* 9: 32–35.

Morgan S, Raspet R. (1992) Investigation of the mechanisms of low-frequency wind noise generation outdoors. *J Acoust Soc Am* 92(2):1180–1183.

Raspet R, Webster J, Dillion K. (2006) Framework for wind noise studies. *J Acoust Soc Am* 119(2):834–843.

Ricketts TA, Dittberner AB, Johnson EE. (2008) High-frequency amplification and sound quality in listeners with normal through moderate hearing loss. *J Speech Lang Hear Res* 51(1):160–172.

Stender T, Hielscher M. (2011) Windguard: bringing listening comfort to windy conditions. White paper. ReSound, Glenview, IL.

Strasberg M. (1988) Dimensional analysis of windscreen in noise. *J Acoust Soc Am* 83:544–548.

Strom KE. (2013) HR 2013 hearing aid dispenser survey: dispensing in the age of Internet and big box retailers. *Hear Rev* 21(4): 22–28.

Wuttke J. (1991) Microphones and the wind. *J Audio Eng Soc* 40: 809–817.

Zakis JA. (2011) Wind noise at microphones within and across hearing aids at wind speeds below and above microphone saturation. *J Acoust Soc Am* 129(6):3897–3907.