PRESERVING INTERAURAL TIME DELAY CUES DURING NOISE REDUCTION IN HEARING AIDS

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Hearing impaired persons localize sounds better without their bilateral hearing aids than with them [8]. This is not surprising, since noise reduction algorithms currently used in hearing aids are not designed to preserve speech localization cues [2]. The inability to correctly localize sounds puts the hearing aid user at a disadvantage as well as at risk. The sooner the user can localize the speech, the sooner the user can begin to exploit visual cues. Generally, visual cues lead to large improvements in intelligibility for hearing impaired persons [3]. Moreover, in certain situations, such as traffic, incorrectly localizing sounds could endanger the user. This paper focuses specifically on interaural time delay (ITD) cues, which help the listener localize sounds horizontally [10]. ITD is the time delay in the arrival of the signal between the left and right ear. If the ITD cues of the processed signal are the same as the ITD cues of the unprocessed signal, we assume that a user will localize the processed signal and the unprocessed signal to the same source.

The goal of this paper is to develop a noise reduction algorithm that preserves speech and noise ITD cues. Unfortunately, in order to preserve the noise ITD cues some of the noise signal must arrive at the output of the algorithm unprocessed. Consequently, any algorithm that preserves noise ITD cues will sacrifice some noise reduction performance. The user and current acoustical situation determine how much noise reduction performance can be sacrificed. In [1], the authors explain that the goal of a noise reduction algorithm is to improve the signal-to-noise-ratio; so the hearing aid user can understand the sentence as a normal hearing person. Completely attenuating the noise signal is not necessary.

In [9], a binaural adaptive noise reduction algorithm is proposed. This algorithm takes a microphone signal from each ear as inputs. The inputs are filtered by a high-pass and a low-pass filter with the same cut-off frequency to create a high frequency and a low frequency portion. The high frequency portion is adaptively processed and added to the delayed low frequency portion. Since ITD cues are contained in the low-frequency regions, as the cut-off frequency increases more ITD information will arrive undistorted to the user [10]. The major draw back to this approach is that the low frequency portion containing the ITD cues also contains noise. Consequently, noise, as well as ITD cues are passed from the input to the output unprocessed. Therefore, a trade-off exists between noise reduction and the preservation of the speech and noise ITD cues. As the cut-off frequency increases the preservation of the speech and noise ITD cues improves at the cost of noise reduction.

This paper extends the binaural multi-channel Wiener filtering algorithm discussed in [5]. The controlled binaural multi-channel Wiener filtering algorithm attempts to estimate the speech component and a specified amount of the noise signal of the mth microphone pair. A Wiener filter estimates a portion, λ, of the noise signal. Subtracting this noise signal estimate from the original signal leads to the estimate of the speech component and the specified amount of the noise signal. If $\lambda = 1$, then the algorithm performs the maximum amount of noise reduction possible and is the same as the algorithm proposed in [5]. On the other hand, when $\lambda = 0$ no noise reduction is performed. As less emphasis is put on noise reduction, more noise arrives at the output of the algorithm unprocessed; accordingly more noise ITD cues will arrive undistorted to the user. Therefore, one can control the distortion of the ITD cues of the noise source. A value for $\lambda \in [0, 1]$ must be chosen that suits the user and the current acoustical situation. Although both algorithms must sacrifice noise reduction to preserve noise ITD cues, the advantage of the controlled binaural multi-channel Wiener filtering algorithm is that it preserves the speech ITD cues without sacrificing noise reduction performance.

In order to estimate the Wiener filters, we make two standard assumptions. First, the speech signal is assumed to be statistically independent of the noise signal. Second, we assume that the noise is short-term stationary.

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The recordings used for our simulations were made in an anechoic room. Two CANTA behind the ear (BTE) hearing aids were placed on a CORTEX MK2 artificial head. Mounted on each hearing aid were two omni-directional microphones. The speech and noise sources were placed one meter from the center of the artificial head. The sound level measured at the center of the dummy head was 70dB SPL. Speech and noise sources were recorded separately. All recordings were performed at a sampling frequency of 32kHz. HINT sentences and ICRA noise[1] were used for the speech and noise signals [6].

Using only the front microphone signals from each hearing aid, the algorithms estimated the speech component in this first microphone pair. The signals fed into the binaural algorithms were 10 seconds in length. The first half of the signal consisted of noise only. A short one and a half second sentence was spoken in the second half amidst the continuous background noise. The speech source varied from 0 to 345 degrees in increments of 15 degrees. The noise source remained fixed throughout the simulations at 90 degrees. These simulations tested the influence of λ, in the controlled binaural multi-channel Wiener filtering algorithm, and the cut-off frequency, in the binaural adaptive algorithm of [9], on ITD preservation and noise reduction performance. For the controlled binaural multi-channel Wiener filtering algorithm the filter length, N, was fixed at 100. The parameter λ was set at 1, 0.7 and 0.6. The filter length of the binaural adaptive algorithm of [9] was 201. Cut-off frequencies of $500Hz$ and $1200Hz$ were simulated. The ITD cues of the processed and unprocessed signal were calculated. If the ITD cues of the processed and unprocessed signal match, then ITD is preserved. The intelligibility weighted signal-to-noise-ratio (SNR_{INT}), defined in [4], is used to quantify the noise reduction performance.

The controlled binaural multi-channel Wiener filtering algorithm, with $\lambda = 1$, 0.7 and 0.6, always preserves the speech ITD cues. Since the parameter, λ, controls the amount of noise reduction performed, as λ decreases from 1 to 0.7 and again to 0.6, the error of the noise ITD cues also decreases. Unfortunately, this comes at a price; the noise reduction performance of the algorithm degrades as λ decreases. On the other hand, the binaural adaptive algorithm of [9] sacrifices noise reduction performance in order to preserve speech and noise ITD cues. With a cut-off frequency equal to $500Hz$, there are still large ITD errors for the speech and noise components. In order to preserve the speech ITD cues, the cut-off frequency must be increased to $1200Hz$. This is also the cut-off frequency necessary to preserve the noise ITD cues. Such a high cut-off frequency causes poor noise reduction performance.

What makes the controlled binaural multi-channel Wiener filtering algorithm so attractive is its ability to adapt to the user and current acoustical situation. If the user and current acoustical situation require little improvement in $SNR_{INT}$to bring the user to a normal hearing condition, then λ can be decreased to preserve the noise ITD cues. On the other hand, if a large improvement in SNR_{INT} is necessary, then λ can be increased. Although increasing λ will distort the noise ITD cues, the ITD cues of the speech component will be preserved regardless of λ. The binaural adaptive algorithm of [9] is incapable of preserving the speech ITD cues without sacrificing noise reduction performance. Clearly, the controlled binaural multi-channel Wiener filtering algorithm is preferable to the binaural adaptive algorithm of [9], since it does not sacrifice noise reduction in order to preserve the speech ITD cues.

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