PREVENTING ITD DISTORTION IN BINAURAL HEARING AIDS DURING NOISE REDUCTION USING MULTI-CHANNEL WIENER FILTERING

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ABSTRACT

This paper presents a subband extension of the controlled binaural multi-channel Wiener filtering algorithm discussed in [1]. The motivation for moving to a subband implementation of this algorithm is twofold. In addition to the complexity reduction and the improved noise reduction performance [2], a subband implementation allows one to vary the emphasis on noise reduction per subband. The algorithm presented in [1] always preserves the interaural time delay (ITD) cues of the speech component. On the other hand, in order to preserve the noise ITD cues, some of the noise signal is passed unprocessed to the output of the algorithm. In this paper less emphasis is placed on noise reduction only in the low frequency bands, leading to the preservation of the interaural time delay (ITD) cues of the noise component, without sacrificing as much noise reduction performance, especially in the high frequency bands.

1. INTRODUCTION

Hearing impaired persons localize sounds better without their bilateral hearing aids than with them [3]. In addition, noise reduction algorithms currently used in hearing aids are not designed to preserve localization cues [4]. 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 a speech signal, the sooner the user can begin to exploit visual cues. Generally, visual cues lead to large improvements in intelligibility for hearing impaired persons [5]. Moreover, in certain situations, such as traffic, incorrectly localizing sounds could endanger the user. Furthermore, preserving the spatial separation between the target speech and the interfering signals leads to an improvement in speech understanding [6].

This research work was carried out at the ESAT laboratory of the Katholieke Universiteit Leuven, in the frame of the Belgian Programme on Interuniversity Attraction Poles, initiated by the Belgian Federal Science Policy Office IUAP P5/22 ('Dynamical Systems and Control: Computation, Identification and Modelling'), the Concerted Research Action GOA-AMBioRICS, and the Research Project FWO nr.G.0233.01 ('Signal processing and automatic patient fitting for advanced auditory prostheses'). The scientific responsibility is assumed by its authors.

This paper focuses specifically on preserving interaural time delay (ITD) cues, which help the listener localize sounds horizontally [7]. ITD is the time delay in the arrival of the sound 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 design a noise reduction algorithm that does not introduce any processing effects that could adversely affect the hearing aids users, such as distorting ITD cues.

A subband implementation of the controlled binaural multi-channel Wiener filtering algorithm, discussed in [1], is presented in this paper. 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. This is accomplished by designing a Wiener filter that estimates a portion, λ, of the noise signal. Subtracting this partial noise signal estimate from the original signal leads to the estimate of the speech component and the specified amount of the noise signal. If $\lambda = 1$, the algorithm performs the maximum amount of noise reduction possible. 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.

The motivation for a subband implementation of the controlled binaural multi-channel Wiener filtering algorithm is twofold. In [2] it has been shown that a subband implementation of the monaural multi-channel Wiener filtering leads to an improvement in noise reduction performance as well as a reduction in complexity. Additionally, a subband implementation allows one to vary the emphasis of noise reduction in each subband. The major downfall of the controlled binaural multi-channel Wiener filtering algorithm is that putting less emphasis on noise reduction,

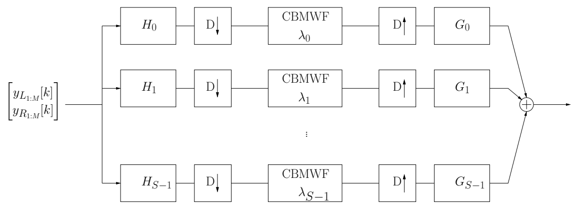

Figure 1: *Subband implementation of the controlled binaural multi-channel Wiener filtering algorithm*

by reducing λ, in order to preserve the noise ITD cues, affects the noise reduction performance over all frequencies even though ITD cues reside only in the low frequencies [7]. With a subband implementation of this algorithm less emphasis can be put on noise reduction in the frequency bands where the ITD cues reside. This leads to significant savings in noise reduction performance.

2. SUBBAND CONTROLLED BINAURAL MULTI-CHANNEL WIENER FILTERING

A typical listening scenario consists of a hearing aid user wearing a binaural hearing aid, each hearing aid has M microphones, with a desired speaker speaking intermittently amidst continuous background noise. The signals received at time instant k for the M microphones on both hearing aids are written as, $y_{L_{1:M}}[k]$ and $y_{R_{1:M}}[k]$. Figure 1 depicts the subband implementation of the controlled binaural multi-channel Wiener filtering algorithm. The signals received at each hearing aid are fed into the algorithm, where they are split up into S subbands by the analysis filters, $H_0, H_1, \ldots H_{S-1}$, and a D-fold downsampling is applied. Next, the controlled binaural multi-channel Wiener filter is applied to each subband using a λ that has specifically been chosen for that subband. After the subband processing the subbands are upsampled and filtered by the synthesis filters, $G_0, G_1, \ldots G_{S-1}$ and summed to produce the enhanced signal. This can be done for as many microphone pairs as necessary.

We use a nearly perfect reconstruction oversampled ($S > D$) uniform discrete Fourier transform modulated filterbank to split the signals up into subbands [9]. The parameters of the filterbanks must be chosen carefully. A large stop-band attenuation is required to limit inter-band aliasing; concurrently the delay of the filterbank must be constrained in order to prevent the degradation of lip reading cues [10].

It should be noted that any algorithm implementing the multi-channel Wiener filter can be used to perform the filtering. Therefore by choosing a QRD-based algorithm, discussed in [11], we can further reduce complexity [10].

3. PERFORMANCE

3.1. Experimental setup and performance measures

The recordings were made in a reverberant room, $T_{60} = 0.76$. Two GN ReSound Canta behind the ear (BTE) hearing aids were placed on a CORTEX MK2 artificial head. Each hearing aid had 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 artificial head was 70dB SPL. Speech and noise sources were recorded separately. All recordings were performed at a sampling frequency of 16kHz. Hint sentences and HINT noise were used for the speech and noise signals [12].

For the experiments the speech source was located at 30 degrees and the noise source at 90 degrees. Only the first microphone pair was used, $M = 1$. The simulations compare the fullband approach from [1], with filter lengths $N = 50, 100$ with the subband approach. Both 16 subbands with a downsampling factor of 12 and 32 subbands with a downsampling factor of 20 were considered, with subband Wiener filter lengths $N_{sub} = 2, 4, 6, 8$. In the subband case, the parameter λ was varied from 0 to 1 only for subbands containing frequencies below 1500Hz (2 subbands for $S = 16$ and 4 subbands for $S = 32$), for the rest of the subbands λ was fixed at 1. The parameter λ was varied from 0 to 1 for the fullband case. A batch implementation of the controlled binaural multi-channel Wiener filtering algorithm was used for these simulations. As mentioned earlier any algorithm implementing a binaural Wiener filter could be used, thereby further reducing the complexity.

The algorithm was evaluated using the ITD error of the processed and unprocessed signals and the improvement in speech intelligibility weighted signal-to-noise ratio. The ITD error is the absolute difference between the ITD of the processed signal and the unprocessed signal. Cross correlation is used to compute the ITD. The intelligibility weighted signal-to-noise ratio (SNR), defined in [13], is used to quantify noise reduction performance.

3.2. Discussion

The simulation results are shown in Figures 2 to 7. Figures 2, 3, 5, and 6 compare the noise reduction performance of the subband algorithm (both 16 and 32 subbands) and the fullband algorithm for the left and right ear. The ITD error is shown in Figures 4 and 7. The ITD error of the speech component, though not pictured, is zero for all simulations.

To make a fair comparison between the subband and fullband approach, we consider the equivalent fullband filter length of the subband algorithm to be the downsampling

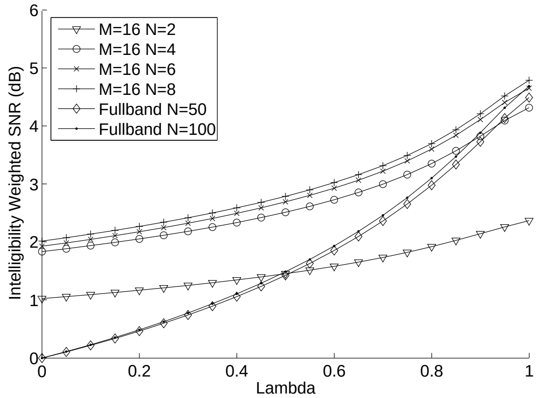

Figure 2: *Improvement in speech intelligibility weighted SNR in the left ear: Comparing 16 subbands and the fullband approach*

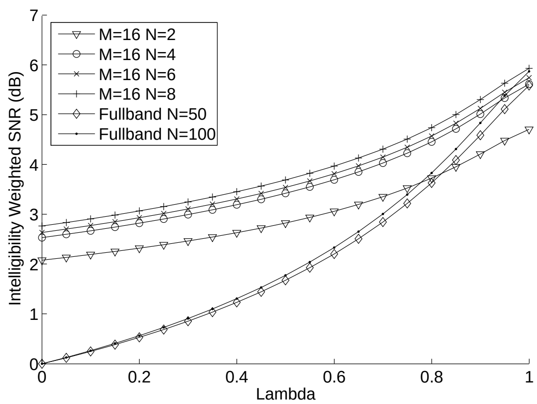

Figure 3: *Improvement in speech intelligibility weighted SNR in the right ear: Comparing 16 subbands and the fullband approach*

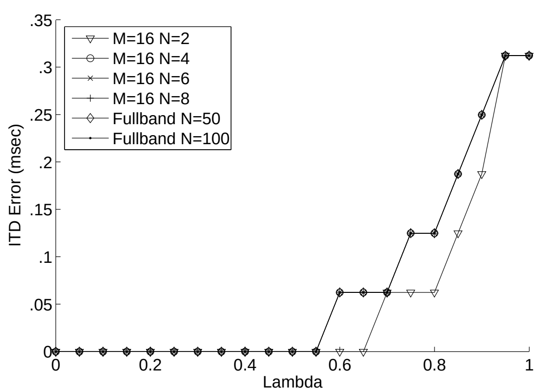

Figure 4: *Absolute ITD error of the noise component: Comparing 16 subbands and the fullband approach*

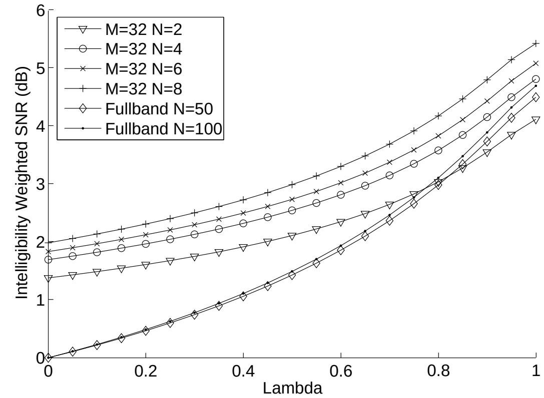

Figure 5: *Improvement in speech intelligibility weighted SNR in the left ear: Comparing 32 subbands and the fullband approach*

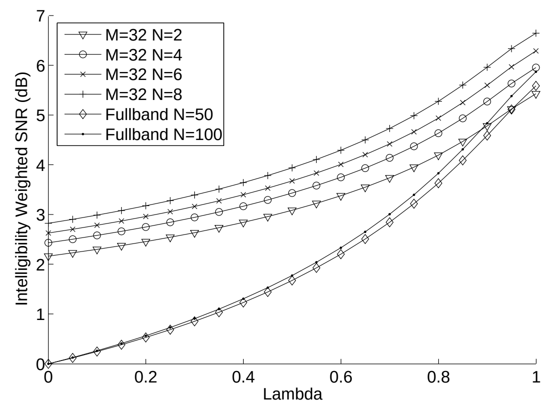

Figure 6: *Improvement in speech intelligibility weighted SNR in the right ear: Comparing 32 subbands and the fullband approach*

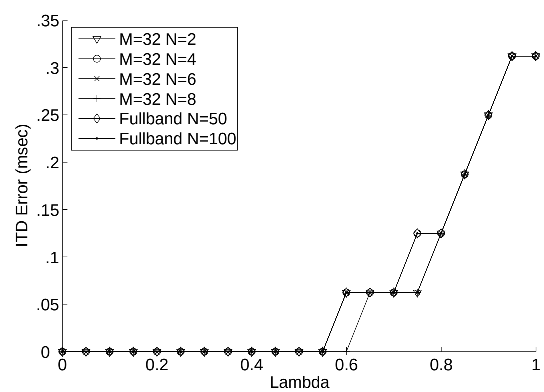

Figure 7: *Absolute ITD error of the noise component: Comparing 32 subbands and the fullband approach*

factor multiplied by the length of the subband filter. When looking at Figures 2, 3, 5, and 6, with $\lambda = 1$, we see that the subband algorithm outperforms the fullband approach. This is consistent with the results shown in [2].

Focusing on Figures 4 and 7, it is clear that in this acoustical situation λ must be reduced from 1 to 0.55 in order to preserve the ITD cues of the noise component. Looking at the improvement in speech intelligibility weighted SNR plots for $\lambda = 0.55$, the subband approach clearly outperforms the fullband approach (except for 16 subbands and a subband filter length of 2). This stems from the fact that λ is frequency selective in the subband approach. Therefore λ only needs to be decreased for the frequencies below 1500Hz in order to preserve the ITD cues of the noise component. This leads to a significant saving in speech intelligibility weighted SNR performance between the subband approach and the fullband approach.

4. CONCLUSION

In this paper a subband extension of the controlled binaural multi-channel Wiener filtering algorithm, discussed in [1], is presented. The motivation for developing a subband implementation is twofold. First, the subband approach reduces the complexity of the algorithm [2]. Second, a subband implementation allows one to vary the emphasis on noise reduction per subband. Like the fullband implementation, the subband implementation preserves the ITD cues of the speech component for all values of λ. However, since the subband implementation allows one to place less emphasis on noise reduction in the low frequency bands, where the ITD cues reside, the amount of noise reduction performance sacrificed in order to preserve the ITD cues of the noise component can be reduced.

5. REFERENCES

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