SPEECH ENHANCEMENT FOR HEARING AIDS IN NOISY ENVIRONMENT

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ABSTRACT

Hearing impaired persons lose their ability to discriminate speech in ambient noise. Usually hearing aids amplify equally speech and noise, thus they do not provide any speech enhancement. Two techniques are proposed to be used on H.A. based on real time adaptive beamforming to reduce the ambient noise thus enhancing the main speech signal. These techniques are evaluated in real time using a specially developed board plugged in a Personal Computer. The evaluation concerns the directivity patterns, the signal to noise ratio and intelligibility improvement in realistic noisy situations. The results are presented.

INTRODUCTION

Hearing impaired persons lose their ability to discriminate speech in ambient noise. The loss of this ability seems to be one third of the total hearing loss (1). Usually hearing aids amplify equally speech and noise; thus they don't provide any speech enhancement.

In this paper we propose the application of digital adaptive beamformers (2,3) in order to cancel the ambient noise in hearing aids.

In all cases we suppose that the hearing impaired person is facing the main interlocutor and that the direction of the noise source(s) is different. The purpose is to create nulls in the directivity pattern of the Hearing Aid at the direction of the noise(s).

Because of the application in Hearing Aids, the number of the microphones used should be small. Two different beamformers using two and three microphones are proposed(5,6).

These adaptive noise cancelling techniques were evaluated in real time using a special board developed in our laboratory(4). The directivity patterns of these beamformers with one and two noise sources are presented.

The efficiency of the filtering on the intelligibility in various every day noisy environments was measured. Non-filtered and filtered lists of monosyllabic french words were presented for that purpose to hearing impaired persons. The results are presented.

ADAPTIVE INTERFERENCE CANCELLING USING TWO MICROPHONES.

One noise source

We suppose that the hearing impaired is facing his interlocutor while the noise is coming from some different direction (fig.1).

Two microphones situated near each ear (for instance at the ear canal's entrance) are used to pick up the sound. Because of symmetry the speech components at each microphone are equal. On the other hand because of head diffraction the noise components are different. This difference is not only a phase difference due to the sound propagation delay time, but also an amplitude difference.

Figure 1. Principle of beamformer II

If we take the difference of the two microphone signals,

\[ D = M_l - M_r = N_l - N_r \]

\( D \) is uncorrelated with the speech but correlated with the noise(5). Therefore it can be used as the reference input of an adaptive noise canceller(3) (see fig.1). We have proved theoretically that at the convergence of the adaptive filter the filter's output is free of noise(6).

Two noise sources(6)

Things become more complicated when two (or more) uncorrelated noise sources of equal power \( \Phi_n \) are present(fig. 2).

When both sources are situated at the same side of the head(left or right) the total noise power at the output of the filter is reduced to:

\[ \Phi_{\text{out}}(z) = \Phi_{\text{in}}(z)(1 - \frac{2(1 - \Delta \phi(z^{-1}))}{k(z)} + k(z)) \]

with \( \Delta \phi(z) \) the transfer function between the two ears for a sound source situated at an angle \( \phi \), and

\[ k(z) = |1 - \Delta \phi(z)|^2 \]

The reduction thus depends on both angles. If both noise sources are at the same position, we can easily verify that the noise is completely cancelled at the output, as we would expect from the previous section.

When the two noise sources are situated at opposite sides of the head(left and right), the expression for the total output noise is quite...
complicated and of no practical interest. We simplify by considering the case of equal angles and noise powers, $\phi=0$ and $\Phi_{11}=\Phi_{22}$.

$$\Phi_{\text{noise}}(x) = \frac{\Phi_{\text{noise}}(x)}{2} + \Phi_{11}(x)\Delta\phi(x)$$

so no noise reduction is observed.

ADAPTIVE BEAMFORMER USING THREE MICROPHONES.

Intuitively we thought that the reason we did not obtain any noise reduction in the last case, is that the components of the noises are of equal power in the reference input of the filter, but not in the primary input. Furthermore they appear with opposite sign.

In order to change this situation a third microphone is introduced at the middle of the front (see fig. 3). Its output is filtered by an FIR filter with a transfer function equal to the acoustical transfer function between the middle of the front and the ear for a source situated at 0°. Taking the difference of each microphone signal and the FIR's output, two signals uncorrelated with speech but correlated with noise are created and used in a two input adaptive noise canceller.

EXPERIMENTAL RESULTS.

In the previous sections we examined the ideal cases. In practice because of small differences between microphones, A/D converters, antialiasing filters etc., the symmetry is not perfect. On the other hand the electronic noise of the above mentioned circuits degrades the performances of the beamformers predicted by the calculation.

The directivity pattern of the filter was measured in our anechoic chamber. The noise used was a pseudorandom noise low-pass filtered at 6.4 kHz. The sound was picked up by a Sennheiser 2002 double microphone system, worn by a Kemar mannikin. During measurement the filter is kept in adaptive mode.
Fig. 4 shows the directivity pattern thus obtained with the two microphone system and fig. 5 with the three microphone system using one noise source. We observe that the performances of the two beamformers are comparable except for 180° where no reduction is observed for the two microphone beamformer while with the three microphone one a 10 dB noise reduction is observed.

In fig. 6 and 7, we have the directivity patterns of the beamformers II and III, obtained with two uncorrelated noise sources (random and pseudorandom) of equal power, low-pass filtered at 6.4 kHz and situated at 45° apart. The dashed line represents the case when the sources are situated at opposite sides of the manikin's head. In all cases beamformer III performs better than beamformer II, especially when noise sources are situated at opposite sides of the head.

Speech and noise recording for intelligibility experiments.

In realistic noise situations because of multiple echoes and multiple noise sources, the noise reduction is not so important as we would expect by the above directivity patterns obtained in an anechoic chamber. Three situations were studied, apartment with one and two noise sources, cafeteria and traffic noise.

The speech material consists of lists of balanced French monosyllabic CVC words (Fourrier series). The speech signal is amplified before emission in order to create various speech to noise ratios and is emitted by the artificial voice of a B&K Head and Torso simulator. A KEMAR manikin with a Sennheiser 2002 double microphone system at the entrance of the ears and a MKE 2 clip-on microphone between the eyes simulates the listener. The three microphone signals and the direct speech signal are recorded using a TEAC A 3440 four channel tape recorder.

In all cases radio news broadcasting was used as the first noise and white noise as the second. Noise was emitted by Fostex 6301 B monitors. For the recordings with traffic noise the two monitors were disposed at 60° and 120° and stereo prerecorded highway noise was played through them (T).

Two cafeterias of our campus (coupole, satellite) were used for the cafeteria recordings. In the first one speech bubble, dishes' noise and door slam are part of the noise (C1), while in the second one we have distinct voices and music together with speech bubble (C2).

The adaptive filtering is applied on the recorded
signals. With beamformer II the localization of the sound sources is lost. On the other hand with beamformer III a configuration is possible to keep the localization but the results are not as good as in the mono configuration.

We observe for all configurations an increase of the speech to noise ratio after adaptive filtering except in case C1. This increase however is reduced when the number of the noise sources increases.

Table 1 shows the mean speech to noise ratio (A-weighted, in dB) of the lists used for each situation for the intelligibility experiences before and after adaptive filtering. This ratio was measured using the B&K 2032 analyzer, as the ratio of the coherent and non-coherent power.

We observe for all configurations an increase of the speech to noise ratio after adaptive filtering except in case C1. This increase however is reduced when the number of the noise sources increases.

Table 1: Speech to noise ratio in dB, A-weighted

<table>
<thead>
<tr>
<th>Original</th>
<th>Filter II</th>
<th>Filter III</th>
</tr>
</thead>
<tbody>
<tr>
<td>L</td>
<td>R</td>
<td>L/R</td>
</tr>
<tr>
<td>S1</td>
<td>-1.5</td>
<td>-6.8</td>
</tr>
<tr>
<td>S2c</td>
<td>-6.5</td>
<td>-3.6</td>
</tr>
<tr>
<td>S2d</td>
<td>-3.8</td>
<td>-4.5</td>
</tr>
<tr>
<td>C1</td>
<td>+0.7</td>
<td>+1.1</td>
</tr>
<tr>
<td>C2</td>
<td>+0.8</td>
<td>-0.2</td>
</tr>
<tr>
<td>T</td>
<td>-6.6</td>
<td>-2.5</td>
</tr>
<tr>
<td>Mean</td>
<td>-2.8</td>
<td>2.8</td>
</tr>
</tbody>
</table>

Intelligibility Experiences

Experiences of intelligibility are actually done at the Physiology Institute of Lausanne University.

Recordings of the filtered and the unfiltered lists are presented through headphones at the most comfortable level. The speech to noise ratio was chosen to provide poor intelligibility before filtering.

Six Hearing Impaired persons, aged 40 for one of them and 59 to 63 for the others, have already passed the test. Three of them have a unilateral hearing loss. The mean hearing thresholds were 25-68 dB SPL for those with a unilateral hearing loss and 49-54 for the others. The following table gives the mean intelligibility scores for the two categories.

Table 2: Mean Intelligibility scores (uni/bi lateral).

<table>
<thead>
<tr>
<th>Original</th>
<th>Filter II</th>
<th>Original</th>
<th>Filter III</th>
</tr>
</thead>
<tbody>
<tr>
<td>Stereo</td>
<td>Mono</td>
<td>Stereo</td>
<td>Mono</td>
</tr>
<tr>
<td>S1</td>
<td>43/17</td>
<td>50/33</td>
<td>30/33</td>
</tr>
<tr>
<td>S2c</td>
<td>27/23</td>
<td>53/30</td>
<td>30/13</td>
</tr>
<tr>
<td>S2d</td>
<td>40/17</td>
<td>57/33</td>
<td>27/20</td>
</tr>
<tr>
<td>C1</td>
<td>50/40</td>
<td>50/37</td>
<td>63/30</td>
</tr>
<tr>
<td>C2</td>
<td>27/17</td>
<td>37/27</td>
<td>17/10</td>
</tr>
<tr>
<td>T</td>
<td>23/17</td>
<td>47/27</td>
<td>27/17</td>
</tr>
<tr>
<td>mean</td>
<td>35/22</td>
<td>49/31</td>
<td>32/21</td>
</tr>
</tbody>
</table>

Table 2: Mean Intelligibility scores (uni/bi lateral).

We observe an increase of the intelligibility after filtering in all cases except C1 where there is no change. Beamformer III gives better results than beamformer II in all cases. In case C2 a relatively small increase of the speech to noise ratio gives an important increase in intelligibility. Concerning the two configurations of beamformer III the persons with bilateral hearing loss prefer the monophonic version, while for the others it depends on the situation.

DISCUSSION.

In this paper we presented two techniques based on adaptive noise cancelling, in order to suppress the ambient noise in hearing aids.

The first one using two microphones gives very good results in the case of one noise source. As long as two noise sources are situated at the same side of the head, there is also a noise suppression although less important.

In order to create notches in the directivity pattern of the filter when two uncorrelated noise sources are situated at opposite sides of the head, we introduced an adaptive noise canceller using three microphones.

In every day situations results are less spectacular than in anechoic chamber, but they provide speech enhancement and increase of intelligibility in most cases, especially with a limited number of noise sources.

In all cases the three microphone beamformer gives better results in mono or stereo configuration. The latter has the advantage to keep the perception of the sound space.

Experiences will be held in order to determine the efficiency of the beamformer in early echo cancelling in reverberant environment.

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