Improved Hybrid Microphone Array Post-filter by Integrating a Robust Speech Absence Probability Estimator for Speech Enhancement

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Abstract

To improve the performance of multi-channel speech enhancement algorithms, we previously proposed a hybrid Wiener post-filter for microphone arrays under the assumption of a diffuse noise field [4]. In this paper, considering the speech presence uncertainty, we further improve the hybrid post-filter presented before by integrating a novel robust estimator for the \textit{a priori} speech absence probability, which makes full use of the correlation characteristics of the noises on different microphone pairs and hence offers the much more accurate speech absence probability estimates. The effectiveness of this improved hybrid post-filter was finally confirmed by the experiments using multi-channel recordings in various car environments.

\textbf{Index Terms:} Speech enhancement, Microphone array, Post-filtering, Speech absence probability.

1. Introduction

In a noisy environment, speech signals impinging on the distant microphones are severely contaminated, resulting in the significant performance decrease of many applications, such as, mobile phone and hearing aid. Enhancing the desired speech signal on distant microphones is currently one of the important issues for hands-free technologies. Therefore, practically effective and computationally efficient speech enhancement algorithms are greatly called for. Multi-channel algorithms have shown great superiority in reducing noise signals and enhancing desired speech signal [1].

Among multi-channel speech enhancement algorithms, post-filter is normally used to further improve the performance of microphone arrays in practical environments [1]. A widely used multi-channel post-filter was first presented by Zelinski under the unpractical assumption of a perfectly incoherent noise field [2]. By relaxing this assumption to that of a diffuse noise field, McCowan \textit{et al.} developed a general expression of the Zelinski post-filter based on the \textit{a priori} coherence function of the noise field [3]. The main drawbacks of these post-filters are the inability to suppress spatially correlated noises and the use of the \textit{a priori} coherence function. To overcome these problems, authors have recently presented a hybrid Wiener filter under the assumption of a diffuse noise field which was proven to be an approximate model in many practical environments [4]. This hybrid Wiener filter follows the framework of the multi-channel Wiener filter, and is capable to reduce spatially correlated as well as uncorrelated noise components.

Moreover, considering the fact that no desired speech signal is present in all frequencies and all frames, many researchers proposed to improve the speech enhancement algorithms under speech presence uncertainty by combining the \textit{speech absence probability} (SAP) [5, 6, 7]. Cohen \textit{et al.} suggested an estimator for the \textit{a priori} SAP based on the energy distribution of the signals at the microphone array output [5]. However, the signal energy distribution at microphone array output has been changed due to the effect of the microphone-array (e.g., beamforming) noise reduction algorithms. Therefore, authors proposed an estimator for the \textit{a priori} SAP based on the coherence characteristics of the noise field at microphone array output because the spatial characteristics of the noise field was proven to be more stable than the statistics of the signals and the spatial characteristics were preserved at the microphone array output [6]. However, this estimator did not fully utilize the coherence characteristics on different microphone pairs in multi-microphone scenarios, resulting in some estimation errors caused by the correlated noise components existing on different microphone pairs.

In this paper, we propose an improved hybrid post-filter for microphone arrays under the assumption of a diffuse noise field by integrating the hybrid Wiener post-filter we presented before and a robust estimator of the \textit{a priori} SAP. This robust estimator fully considers and utilizes the spatial correlations between noise signals on different microphone pairs. In particular, we show that the calculation of the original hybrid post-filter and that of this newly derived the \textit{a priori} SAP estimator can be done in an integrated mechanism. Experimental results show that this improved post-filter outperforms the original post-filter in various car environments.

2. Signal Model

Let us consider a \(M\)-sensor microphone array in a noisy acoustic environment. The observed signal \(x_m(t)\) on the \(m\)-th sensor is composed of two components. The first is the desired signal \(s_m(t)\) by transforming the desired source signal \(s(t)\) with the impulse response \(a_m(t)\) between the sound source and the \(m\)-th sensor. The second is the additive noise \(n_m(t)\). Thus, applying the \textit{short-time Fourier transform} (STFT), the observed signal on the \(m\)-th microphone can be represented as

\[
X_m(k, \ell) = S_m(k, \ell) + N_m(k, \ell), \quad m = 1, 2, \ldots, M
\]

where \(X_m(k, \ell), S_m(k, \ell)\) and \(N_m(k, \ell)\) are the STFTs of the corresponding signals \(x_m(t), s_m(t)\) and \(n_m(t)\), respectively. In this paper, we further assume that the microphone array has been steered into the direction of the desired signal beforehand.

3. Analysis of Diffuse Noise Field

To characterize a noise field, a widely used measure is the \textit{magnitude-squared coherence} (MSC) function, also called coherence function, defined as

\[
\Gamma_{x_i, x_j}(k, \ell) = \frac{|\phi_{x_i, x_j}(k, \ell)|^2}{\phi_{x_i, x_i}(k, \ell)\phi_{x_j, x_j}(k, \ell)},
\]

where \(\phi_{x_i, x_j}(k, \ell)\) is the cross-spectrum of \(x_i\) and \(x_j\), and \(\phi_{x_i, x_i}(k, \ell)\) is the auto-spectrum of \(x_i\).
where $\phi_{x_i x_j}(k, \ell)$ is the cross-spectral density between two signals $x_i(t)$ and $x_j(t)$; $\phi_{x_i}(k, \ell)$ and $\phi_{x_j}(k, \ell)$ are the auto-
spectral densities of $x_i(t)$ and $x_j(t)$, respectively.

A diffuse noise field, which is one of the underlying as-
mptions of this paper, has been shown to be a reasonable
model for many practical noise environments [1, 6]. A diffuse
noise field is characterized by the following coherence function
\[
\Gamma(k) = \left| \frac{\sin(2\pi kd/c)}{2\pi kd/c} \right|^2, \tag{3}
\]
where $d$ and $c$ represent the distance between adjacent micro-
phones and the velocity of sound. The coherence function of a
perfect diffuse field against frequency is plotted in Fig. 1. From Fig. 1 and Eq. 3, some characteristics of the diffuse
noise field can be easily observed: (i) coherence function is a
frequency-dependent and time-invariant measure; (ii) noises on
different microphones are high-correlated in the low frequen-
cies and low-correlated in the high frequencies.

4. An Improved Hybrid Wiener Post-Filer

Under the speech presence uncertainty, we can derive an esti-
mator for two states “speech presence” and “speech absence” as
a weighted sum of individual estimators of the speech signal
calculated in the two states. The weights are the a posterior-
i probabilities of the two states given the noisy observation.
Since the optimal estimator of the desired speech signal given
that this signal is absent in the noisy observation is zero, the
resulting composite estimator is the product of the estimator of
the desired speech signal given that this signal is present in the
noisy observation and the a posteriori probability of signal pres-
ence given the noisy signal. Thus, the improved hybrid Wiener
post-filter can be represented as
\[
G(k, \ell) = G_s(k, \ell) P_s(k, \ell), \tag{4}
\]
where $G_s(k, \ell)$ is the gain function of the hybrid Wiener post-
filter when speech is surely present, defined in [4] and briefly
formulated in the following subsections; $P_s(k, \ell)$ is the speech
presence probability given the noisy observation, given by [4, 5]
\[
P_s(k, \ell) = \left\{ 1 + \frac{q(k, \ell)}{1 - q(k, \ell)} \right\} \left( 1 + \frac{\xi(k, \ell)}{1 + \xi(k, \ell)} \right), \tag{5}
\]
where (i) $q(k, \ell)$ is the a priori SAP; (ii) $\xi(k, \ell) = \lambda_s(k, \ell) / \lambda_n(k, \ell)$ and $\gamma(k, \ell) = |X(k, \ell)|^2 / \lambda_n(k, \ell)$ are the a
priori SNR and a posteriori SNR as named in [5, 7]; (iii)
$\lambda_s(k, \ell)$ and $\lambda_n(k, \ell)$ are the variances of speech signal and
noise signal, respectively.

The basic ideas of our estimator of the a priori SAP which
is based on coherence characteristics are: (i) desired speech sig-
als are strongly correlated on the concerned microphone pairs;
(ii) noise signals are weakly correlated on the concerned micro-
phone pairs. Therefore, this technique will fail when speech
and noise signals are strongly correlated simultaneously on all
microphones. Previously, we presented an estimator for the a
priori SAP based on coherence characteristics [6], however, it
does not sufficiently consider the correlations of the noises on
different microphone pairs. It is believed that the discrimina-
ting accuracy between speech and noise signals, i.e., the accu-
raty of the a priori SAP estimator, will be improved when suffi-
ciently considering and using all spatial correlation information
of noise signals on different microphone pairs.

Considering the spatial coherence characteristics of the
noise field, we divide the frequency band into two parts: the low
frequency region with high noise coherence and the high fre-
quency region with low noise coherence. In the two regions, the
hybrid Wiener post-filter [4] and the robust estimator of the a
priori SAP are implemented in an integrated way. The block
diagram of the proposed post-filter along with a beamformer is
plotted in Fig. 2.

4.1. Improved hybrid post-filter in the high frequencies

As Fig. 1 demonstrates, in a diffuse noise field, the spatially
low-correlated noise components on different microphones only
exist in the frequencies over the transient frequency $f_t = c/(2d)$ [4, 6]. Since the transient frequency is determined by
the distance between microphones, microphone pairs with dif-
ferent inter-element spacing are characterized by different tran-
sient frequencies. That is, for different microphone pairs with
different inter-element spacing, low-correlated noise is found
in different frequency regions. Furthermore, for a certain fre-
quency, noise is mutually low-correlated only on limited micro-
phone pairs, generally not on all pairs. This fact motivates us
to propose a robust estimator for the a priori SAP which makes
full use of the spatial correlations of noises on different micro-
phone pairs.

Figure 1: Magnitude-squared coherence function in car envi-
ronment ($d = 10$ cm).

Figure 2: Block diagram of the proposed system.
In the high frequency region, the proposed robust estimator of the \textit{a priori} SAP and the hybrid post-filter (i.e., a modified Zelinski post-filter in high frequencies) are implemented in the following steps:

1. Determine the transient frequencies according to the microphone array geometry. Considering a $M$-sensor array with the equal adjacent-element spacing $d$ and the distance $d_{ij}$ between sensors $i$ and $j$ ($i, j \leq M$), we have $M(M-1)/2$ microphone pairs which determine $M(M-1)/2$ transient frequencies, each of them can be calculated by $f_{t,ij} = c/(2d_{ij})$. Since the inter-element spacings are identical for some microphone pairs, some transient frequencies are identical as well. In principle, among $M(M-1)/2$ microphone pairs, only $M-1$ pairs have different inter-element spacings. Correspondingly, we can determine $M-1$ different transient frequencies, denoted by $f_1^t, f_2^t, \ldots, f_{M-1}^t$. Without loss of generality, we further assume the following relationship between transient frequencies $f_1^t < f_2^t < \ldots < f_{M-1}^t$.

2. Determine the microphone pairs on which noise is mutually uncorrelated for each frequency. As a matter of fact, the $M-1$ different transient frequencies, $f_1^t, f_2^t, \ldots, f_{M-1}^t$, divide the full frequency band into $M$ sub-bands, denoted by $B_0, B_1, \ldots, B_{M-1}$. In each sub-band (except $B_0$), some microphone pairs provide low-correlated noise components on microphones of the pairs. In principle, the $M(M-1)/2$ microphone pairs can be grouped into $M-1$ sets where some microphone pairs are re-used. Each of $M-1$ sets includes the microphone pairs on which noise signals are mutually low-correlated for the individual frequency of interest. Corresponding to the transient frequencies $f_1^t, f_2^t, \ldots, f_{M-1}^t$, the $M-1$ microphone pair sets are represented as: $\Omega_1, \Omega_2, \ldots, \Omega_{M-1}$.

3. Compute the magnitude squared coherence functions. For each frequency in sub-band $B_m$ ($1 \leq m \leq M-1$), the noise on the microphone pairs of set $\Omega_m$ is assumed to be weakly correlated. Therefore, MSC functions calculated in this situation provide much more accurate cues to detect the presence of the desired speech signal. To improve the robustness of the \textit{a priori} SAP estimator against estimation errors, estimates of the auto- and cross-spectral densities are averaged across the microphone pairs in the corresponding pair set $\Omega_m$ (not all microphone pairs). The MSC function is thus given by

$$
\Gamma(k, \ell) = \frac{1}{|\Omega_m(k)|} \sum_{(i,j) \in \Omega_m(k)} \left| \phi_{x_i x_j}(k, \ell) \right|^2 \sum_{(i,j) \in \Omega_m(k)} \left| \phi_{x_i x_j}(k, \ell) \phi_{x_j x_i}(k, \ell) \right|^2.
$$

4. Compute the \textit{a priori} SAP. After calculating the MSCs, we can detect the desired speech signal as follows. If a high coherence (higher than a threshold $T_{\text{max}}^h$) is observed, a speech present state is detected presumably. If a low coherence (lower than a threshold $T_{\text{min}}^h$) is observed, a speech absent state is detected presumably. Note that the \textit{a priori} SAP decreases as the MSC increases. For the MSCs in the range $[T_{\text{min}}^h, T_{\text{max}}^h]$, the \textit{a priori} SAPs are determined by linear interpolation. Thus, the \textit{a priori} SAPs in the high frequency region $q_h(k, \ell)$ is given by

$$
q_h(k, \ell) = \begin{cases} 
0, & \Gamma(k, \ell) > T_{\text{max}}^h \\
1, & \Gamma(k, \ell) < T_{\text{min}}^h \\
\frac{T_{\text{max}}^h - \Gamma(k, \ell)}{T_{\text{max}}^h - T_{\text{min}}^h}, & \text{otherwise}
\end{cases}
$$

Moreover, due to the low noise correlation on the microphone pairs of set $\Omega_m$, the original hybrid post-filter (i.e., the modified Zelinski post-filter) in the high frequency region $G_h^h(k, \ell)$ was calculated as [4]

$$
G_h^h(k, \ell) = \frac{1}{|\Omega_m(k)|} \sum_{(i,j) \in \Omega_m(k)} \Re\left\{ \phi_{x_i x_j}(k, \ell) \right\} \sum_{(i,j) \in \Omega_m(k)} \left[ \phi_{x_i x_j}(k, \ell) + \phi_{x_j x_i}(k, \ell) \right].
$$

Note that the averages for the auto- and cross-spectral densities are performed on only limited microphone pairs in the corresponding pair set $\Omega_m$ on which noises are weakly correlated. Therefore, it avoids the estimation error caused by the correlated noises and offers more accurate and robust estimates for the \textit{a priori} SAPs, further improving the performance of the improved hybrid Wiener post-filter.

### 4.2. Improved hybrid post-filter in the low frequencies

In the low frequency sub-band ($B_0$ where $k < f_1^l$), MSCs computed in this region fail to detect the speech signal because both speech and noise are strongly correlated on all microphone pairs. Here, we compute a MSC value that is averaged across the frequencies over the minimum transient frequency, given by

$$
\Gamma(k, \ell) = \frac{1}{M-1} \sum_{m=1}^{M-1} \frac{1}{|\Omega_m(k)|} \sum_{(i,j) \in \Omega_m(k)} \left| \phi_{x_i x_j}(k, \ell) \right|^2.
$$

Following the same concept used in the high frequency region, we derive an estimator for the \textit{a priori} SAP in the low frequency region $q_l(k, \ell)$, given by

$$
q_l(k, \ell) = \begin{cases} 
0, & \Gamma(k, \ell) > T_{\text{max}}^l \\
1, & T_{\text{max}}^l - \Gamma(k, \ell) \leq T_{\text{min}}^l \\
\frac{T_{\text{max}}^l - \Gamma(k, \ell)}{T_{\text{max}}^l - T_{\text{min}}^l}, & \text{otherwise}
\end{cases}
$$

Moreover, due to the high correlations of speech and noise signals on all microphone pairs, the original hybrid post-filter (i.e., the Wiener post-filter) in the low frequency region $G_l^l(k, \ell)$ was calculated by [4]

$$
G_l^l(k, \ell) = \frac{\xi(k, \ell)}{1 + \xi(k, \ell)}.
$$

Consequently, substituting the \textit{a priori} SAP estimates, given by Eqs. (7) and (10) into Eq. (5), we can obtain the speech presence probability estimates. Finally, putting Eqs. (5), (8) and (11) into Eq. (4), we can derive the improved proposed hybrid post-filter which is expected to show more effective in speech enhancement.

### 5. Experiments and Results

To validate the effectiveness of the improved hybrid post-filter in a diffuse noise field, its performance was investigated and further compared with the original hybrid Wiener post-filter that we proposed before [4] in various car noise environments. A beamformer, implemented by a superdirective beamformer [8], is first applied to the multi-channel noisy signals. Then, the beamformer output is further enhanced by the studied post-filters. The performance is evaluated in terms of objective measures: \textit{segmental SNR} (SEGSNR) and \textit{log-spectral distance} (LSD), defined in [5].
Table 1: Segmental SNR [dB] results of the noisy signal, superdirective beamformer (SDBF), the original hybrid post-filter (ORG-PF) and the improved hybrid post-filter (Imp-PF).

<table>
<thead>
<tr>
<th>Condition</th>
<th>Global SNR</th>
<th>50km/h</th>
<th>100km/h</th>
</tr>
</thead>
<tbody>
<tr>
<td>Noisy</td>
<td>−9.07</td>
<td>−8.62</td>
<td>−8.18</td>
</tr>
<tr>
<td>SDBF</td>
<td>−8.96</td>
<td>−8.48</td>
<td>−8.02</td>
</tr>
<tr>
<td>ORG-PF</td>
<td>−7.95</td>
<td>−7.51</td>
<td>−7.09</td>
</tr>
<tr>
<td>Imp-PF</td>
<td>−6.41</td>
<td>−5.00</td>
<td>−2.87</td>
</tr>
</tbody>
</table>

Table 2: Log spectrum distance [dB] results of the noisy signal, superdirective beamformer (SDBF), the original hybrid post-filter (ORG-PF) and the improved hybrid post-filter (Imp-PF).

<table>
<thead>
<tr>
<th>Condition</th>
<th>Global SNR</th>
<th>50km/h</th>
<th>100km/h</th>
</tr>
</thead>
<tbody>
<tr>
<td>Noisy</td>
<td>9.92</td>
<td>7.75</td>
<td>5.92</td>
</tr>
<tr>
<td>SDBF</td>
<td>9.20</td>
<td>7.25</td>
<td>5.58</td>
</tr>
<tr>
<td>ORG-PF</td>
<td>5.33</td>
<td>4.15</td>
<td>3.18</td>
</tr>
<tr>
<td>Imp-PF</td>
<td>2.22</td>
<td>1.71</td>
<td>1.42</td>
</tr>
</tbody>
</table>

5.1. Experimental configurations

To assess the performance of the studied post-filters, an equally-spaced linear array, consisting of three microphones with inter-element spacing of 10 cm, was mounted above the windshield in a car. The array was about 50 cm apart from and in front of the driver. The multi-channel noise recordings were performed across all channels simultaneously when the car was running at the speeds of 50km/h and 100km/h. The multi-channel speech recordings, consisting of 10 Japanese digits, were performed when the car was stationary. Both speech and noise signals were first re-sampled to 12kHz at 16 bit accuracy. We generated the multi-channel noisy signals by artificially mixing the multi-channel speech signals and the multi-channel noise signals at different global SNR levels [−5, 20] dB.

The effectiveness of the diffuse noise field was investigated by comparing the measured coherence function calculated from real noise recordings with the theoretical function, plotted in Fig. 1. Note that the measured coherence function follows the trend of the theoretical function. Therefore, it fulfills the assumption of a diffuse noise field used in the proposed post-filter.

5.2. Objective evaluation results

Experimental results of the average SEGSNR, calculated in two noise conditions at various SNR levels, are listed in Table 1; the results of LSD are presented in Table 2. The performance was evaluated at the first microphone, the beamformer output and the studied post-filter outputs.

As shown in Table 1, the beamformer alone provides only a small degree of SNGSNR improvements compared to the noisy inputs due to its low ability in reducing the noise components in the low-frequency region where car noises mainly concentrate. The original hybrid post-filter gives relatively higher average SEGSNR improvement of about 1.3 dB. The improved hybrid post-filter newly proposed demonstrates the highest average SEGSNR improvement of about 5.17 dB, corresponding to the highest speech quality.

Concerning the results of LSD, shown in Table 2, we can readily observe that the beamformer alone decreases the LSD more or less in all conditions. Much lower LSD of about 2.56 dB decrease is observed at the original hybrid post-filter output because of the reduced noise components. Further, the improved post-filter offers the lowest LSD of about 4.1 dB decrease, corresponding to the lowest speech distortion.

6. Conclusion

In this paper, we proposed an improved hybrid Wiener post-filter for microphone arrays by integrating the hybrid post-filter we previously presented and a robust speech presence probability estimator. In this improved post-filter, we fully consider the coherence characteristics of a diffuse noise field, which offers the more accurate SAP estimates and further improve the noise reduction performance of this post-filter. Experimental results demonstrate the superiority of this improved hybrid Wiener post-filter in reducing noise signals and preserving desired speech signal in various car environments.

7. References