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NOISE REDUCTION ALGORITHMS FOR BETTER SPEECH UNDERSTANDING IN COCHLEAR IMPLANTATION- A SURVEY

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Abstract

Cochlear Implant (CI) is an electronic device which provides a sense of sound to a person who is deaf by birth or severely hard of hearing. Improvement in speech understanding in noisy environment is a major challenge in cochlear implant. Various noise reduction algorithms have been designed in cochlear implant for better understanding in noisy environment. Starting from Two microphone adaptive beamformer method, SDW-MWF, SP-MWF, EVD based filter and GEVD based approximation of the filters are discussed here. Survey regarding the various algorithms for noise reduction and better speech understanding performance in cochlear implants are studied and analysed.

Keywords- NR algorithms, SDW-MWF, EVD, SP-MWF, and GEVD based NR filters

1. Introduction

Several cochlear implant devices have been designed over the years. Improving the speech understanding in a noisy environment is one of the most important challenges in cochlear implant. Various algorithms have been developed for Noise Reduction (NR) in cochlear implantation for better speech understanding. Usually multiple microphones are included in cochlear implants for multichannel adaptive NR algorithms like BEAM in the Cochlear Freedom device which improves speech understanding for cochlear implant recipients[1].

Normally cochlear implant recipients need a 10 dB to 25 dB higher Signal to Noise Ratio (SNR) compared to normal hearing aids to achieve a similar speech understanding performance. To tune Multichannel Wiener Filter (MWF) for performing a more aggressive NR by allowing more speech distortion can be done by means of the Speech Distortion Weighted Multichannel Wiener Filter (SDW-MWF)[2],[3],[4]. Later for a single speech source the performance of SDW-MWF can be improved sometimes by reformulating the filters based on the assumption that the frequency domain autocorrelation matrix of the speech signal is a rank one matrix. These type of thee filters can be so called as Spatial Prediction MWF (SP-MWF) [5].

Based on a rank one approximation with first column decomposition, the autocorrelation matrix of the signal is estimated for all these NR algorithms, along with the assumption that the speech signal and the noise are uncorrelated and stationary. The autocorrelation matrix of the voice signal may be wrongly estimated and become non positive semi-definite in low input scenario. To overcome this issue we go for next algorithm in which an alternative rank one approximation based on Eigenvalue decomposition (EVD) [6] or Generalized Eigenvalue decomposition (GEVD) [7], of the autocorrelation matrix of the speech signal. Better noise reduction performance can be delivered by these alternative NR filters especially in low input SNR scenarios. Survey on those algorithms is described in section 2 to section 6

2. Speech understanding with the 2-Microphone Adaptive Beamformer

For speech understanding in background noise, the merits of the two-microphone adaptive beamformer in the Nucleus Freedom cochlear implant (CI) system is evaluated in [1].

2.1 Method:

2-Microphone adaptive beamformer is used in Nucleus Freedom cochlear implant systems for speech understanding.

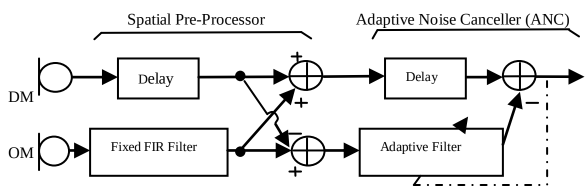

Figure 1: 2-microphone adaptive beamformer

In a main stream commercial CIs, this implementation is the first adaptive noise reduction strategy. Tests for speech understanding conducted in lab demonstrated the improvement in both the percentage correct phoneme scores and SRT in noise with BEAM compared with standard hardware microphone, especially more benefit in noise level 65dB SPL and single noise source at $90^{o.}$

3. SP SDW MWF for Noise Reduction in Hearing Aids

To improve the intelligibility for hearing impaired people, the noise reduction algorithms are crucial especially in background noise. In addition to temporal and spectral information of the desired and noise signal, Multimicrophone systems exploit spatial and so single microphone is mostly preferred.

3.1 Method:

Filter for speech processor in cochlear implant is designed by Spatially Pre-Processed Speech distortion Weighted Multi-Channel Wiener Filtering in [2].

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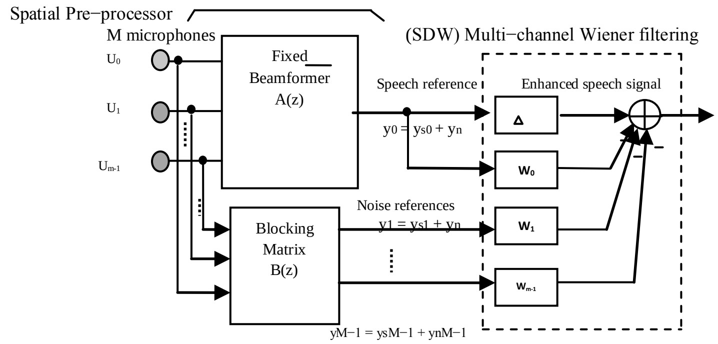

Figure 2: SDW MWF based on Spatially Pre-processed (SP-SDW MWF).

The SP-SDW-MWF mainly consist of three blocks such that spatial pre-processor (fixed beamformer) A(z) and a blocking m matrix B(z), and an adaptive SDW-MWF as shown in Figure 2.

Given M microphone signals

$$u_i[k] = u_i^s[k] + u_i^n[k], i = 1, \ldots., M \tag{1}$$

The spatial pre-processor produce a speech reference

$$y_0[k] = y_0^s[k] + y_0^n[k] \tag{2}$$

steering a beam towards the front and M − 1 noise references

$$y_i[k] = y_i^s[k] + y_i^n[k], i = 1, \ldots.., M-1 \tag{3}$$

Here a generalized scheme, called SP SDW-MWF that encapsulates the GSC and MWF as extreme cases and allows for the Speech Distortion Regularized GSC (SDR-GSC) which is an in-between solutions. By taking speech distortion explicitly into account in the design criterion of the adaptive stage, robustness is added by means of the SDR-GSC to the GSC. The SDR-GSC performs better noise reduction for small model errors than widely studied QIC-GSC, while guaranteeing robustness against large model errors. Along with that, the speech reference's extra filtering in the SP-SDW-MWF further improves the performance.

4. Frequency-Domain Criterion for SDW-MWF for robust Noise Reduction

The Spatially Pre-processed Speech Distortion Weighted Multichannel Wiener Filter (SP-SDW-MWF is a generalised noise reduction scheme. The SP-SDW-MWF achieves a better noise reduction performance for a given maximum speech distortion level. In addition to that, using diagonal correlation matrices, an adaptive frequency-domain algorithm for the SDW-MWF has been presented in [4] to reduce the memory usage and the computational complexity.

4.1 Method:

A frequency-domain criterion for the SDW-MWF is designed for noise reduction and speech distortion.

The SP-SDW-MWF defined as

$$J(w[k]) = \varepsilon_v^2[k] + \frac{1}{\mu}\varepsilon_x^2[k] = \varepsilon\left\{|v_0[k-\Delta] - w^T[k]v[k]|^2\right\} + \frac{1}{\mu}\varepsilon\{|w^T[k] \times [k]|^2\} \tag{4}$$

Frequency-domain criterion for SP-SDW-MWF can defined as

$$J_f[m] = (1-\lambda_\vartheta)\sum_{i=0}^{m}\lambda_\vartheta^{m-i}\,\underline{e}_\vartheta^H[i]\underline{e}_\vartheta[i] + \frac{1}{\mu}(1-\lambda_x)\sum_{i=0}^{m}\underline{e}_x^H[i]\underline{e}_x[i]\,) \tag{5}$$

Where $\varepsilon_v^2[k]$ represents the residual is noise energy and $\varepsilon_x^2[k]$ is the speech distortion energy. $v_0[k]$ Represents the Noise component of the speech reference and $v[k]$ is the stacked noise component vector. Where $\underline{e}_\vartheta[i]$ represents the residual noise in the frequency-domain, $\underline{e}_x[i]$ defines the speech distortion in the frequency domain. In [4], an adaptive frequency-domain algorithm for the SDW-MWF has been presented to reduce the memory usage and decrease the calculating complexity. So that the new frequency domain scheme for SDW-MWF improves the noise reduction performance and speech.

5. Multichannel Wiener Filter-Based Noise Reduction under second order statistics estimation errors

From the previous contribution it is clear that the speech distortion weighted multichannel Wiener filter (SDW-MWF) is an efficient multi-microphone noise reduction technique. It is normally assumed that there is a single target speech source for theoretical studies. The filter can then be decomposed into a spatial filter and also a single-channel post filter for a performance analysis. But the problem here is, it is not straightforward to make a robust practical implementation based on this decomposition.

5.1 Method:

In [5], a theoretical study and experimental validation is performed on both the standard SDW-MWF implementation and two recently introduced alternative filters, called the rank-1 SDW-MWF and the spatial prediction SDW-MWF. Both the cases are analyzed and studied in the presence of estimation errors in the second-order statistics. These alternative SDW-MWF implementations behave close to the theoretical optimum, which is proved theoretically through experiments.

The SDW-MWF minimizing the cost function is defined as

$$W_{MWF} = (R_x + \mu R_v)^{-1} R_x e_{ref} \tag{6}$$

Rank-One SDW-MWF defined as

$$W_{R1-MWF} = R_v^{-1} R_x e_{ref}\,\frac{1}{\mu + Tr\{R_v^{-1}R_x\}} \tag{7}$$

The speech distortion weighted SP-MWF expression is obtained as

$$W_{SP-MWF} = R_v^{-1} R_x e_{ref}\,\frac{e_{ref}^H R_x e_{ref}}{\mu e_{ref}^H R_x e_{ref} + Tr\{R_v^{-1} R_x e_{ref} e_{ref}^H R_x\}} \tag{8}$$

From the analysis in [5], we can conclude that, The R1-MWF slightly loses its benefit over the SDW-MWF when the number of target speech sources increases. But achieves high SNR improvements with more distortion with increase in number of target speakers. Whereas the SP-MWF introduces distortions in the signals with low input powers by providing large SNR improvement.

6. Low Rank-1 Approximation based MWF Algorithms for NR Algorithm

The speech signal's autocorrelation matrix can be approximated by a rank-1 matrix, in single speech scenario in [6].Filter for noise reduction can be derived from this matrix approximation providing improved signal to noise ratio.

6.1 Method:

Noise reduction filter is derived from rank-1 matrix. This rank-1 matrix is approximated from the autocorrelation matrix of the speech signal. Multichannel wiener filter approximated by rank-1 matrix is defined in (7). The NR algorithms discussed above rely on the speech autocorrelation matrix estimation. In these cases there may be chance of wrong estimation of speech autocorrelation matrix, in low input SNR and become non positive semi-definite. SDW-MWF and SP-MWF behaves unpredictably. Selecting a rank-1 approximation based on an EigenValue Decomposition (EVD) of the speech autocorrelation matrix [6] is an alternative solution for this problem. Again it is generalised to Rank-R based filter which consist of the GEVD-SDW-MWF (GEVD-1) and the SDW-MWF (GEVD-M) as extreme cases.

In advance to that approximation based on GEVD Generalized Eigen Value Decomposition of the speech autocorrelation matrix [7] is analysed. In [7] the NR filters like SDW-MWF, SP-MWF are modeled as EVD based SDW-MWF (EVD SDW-MWF) and GEVD based SDW-MWF (GEVD SDW-MWF). Similarly SP-MWF is modified as EVD SP-MWF and GEVD SP-MWF. And the performance between these filters are compared and analyzed.

SDW-MWF-EVD

$$W_{SDW-MWF} = (R_{Sr1} + \mu \left(R_n + \frac{1}{\mu} R_Z\right))^{-1} R_{Sr1} f1 \qquad (9)$$

SP-MWF-EVD

$$W_{SP-MWF} = (R_{Sr1} + \mu (R_n)^{-1} R_{Sr1} f1 \qquad (10)$$

Under the Experimental Set up in [7] performance graph for SIW-SNR is plotted for various input SNR value. These analysis is implemented for SDW-MWF,SP-MWF,EVD-SDW-MWF,EVD-SP-MWF and GEVD-SDW-MWF. Resulting values is tabulated as

Table 1: SIW-SNR performance

SNR(dB)	-15	-10	-5	0	5
SDW-MWF	0	1	2	3	2
EVD-SDW-MWF	2	3.5	4	3.5	2
EVD-SP-MWF	2	3.5	4	4	3
GEVD-SDW-MWF	2	3.7	4	4	4

From the above Table 1, EVD based NR filter and the GEVD based NR filter perform similar SIW-SNR improvement at 2dB compared with other filter like SDW-MWF and with SP-MWF SNR decreases unpredictably.

Table 2:%NPD for the left ear

SNR(dB)	-15	-10	-5	0	5
SDW-MWF	65	50	35	15	0
EVD-SDW-MWF	59	42	25	10	0
GEVD-SDW-MWF	49	35	20	5	0

From this Table 2 it is clear that for EVD based NR filter and GEVD based NR filter direct estimation leads to 70% NDP at -15dB SNR and at the same SNR direct estimation for SP-MWF leads to vary unpredictably from 50% to 60% respectively. From the above analysis, for low input SNR generalized eigen value decomposition based SDW-MWF provides high SIW-SNR improvement which leads to approximate the filter to rank-1 matrix.

7. Conclusions

From the above discussions, various noise reduction algorithms have been used for cochlear implantation for better speech understanding. Generalized Eigenvalue decomposition based filter design provides accurate rank-1 approximation to derive NR filters.

In general method, due to non-stationarity in noise at low input SNR, speech signal’s autocorrelation matrix estimation may not be positive semi-definite. EVD based Rank-1 approximation scheme for SDW-MWF and for SP-MWF has been introduced to solve this problem. By this method it is possible autocorrelation matrix of speech signal into sum of two matrixes namely rank-1 approximation and a reminder matrix.

To deliver better SIW-MWF than the corresponding SDW-MWF methods, GEVD based rank-1 approximation approach to SDW-MWF implemented with same speech distortion s proposed finally. SDW-MWF can provide same SIW-MWF as GEVD base SDW-MWF with increased speech distortion. So finally rank-R approximation based approach (GEVD-R) introduced which contain both the GEVD-SDW-MWF (GEVD-1) and the SDW-MWF (GEVD-M) as extreme cases. Discussed algorithms might limit by its benefits due to the need of a VAD at SNR ranging from -15dB to 5dB. This remark can be rectified in future.

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