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TWO-STAGE ADAPTIVE FILTERING TECHNIQUES FOR NOISE CANCELLATION IN HEARING AIDS

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ABSTRACT

Most of the existing adaptive noise cancelling techniques produce little improvement in the intelligibility of speech for the hearing impaired in reverberant environments. In this paper we propose two-stage adaptive structures to improve speech intelligibility in noise. The first stage removes the interference and produces a distorted version of the signal. The second stage then removes this distortion and the clean signal is obtained. We suggest two algorithms: an unconstrained and a constrained one for the first stage. In high Signal-to-Noise Ratios (SNR) both perform very well and more than 20dB noise reduction may be achieved. In low SNR the constrained algorithm performs better with a slight increase in the computational load.

1 INTRODUCTION

Hearing impaired listeners have difficulty in perceiving sounds when there are other noise sources or competing speakers present in the same environment (*Cocktail party effect)*. The loss of spatial cues and the reduced frequency selectivity of the ear contribute significantly to this impairment. Adaptive Noise Cancellation [1] is one of the approaches widely used to increase the spatial selectivity of the hearing aids.

Adaptive noise cancellation works extremely well when the reference input contains a signal that is highly correlated with the noise in the primary input but uncorrelated with the desired signal. When the microphones are not separated well enough the reference input contains a signal correlated with the desired signal and noise. In this case the primary input signal $X_1(z)$ is $S(z) + N(z)$ and the reference input $X_2(z)$ is $G(z)S(z) + H(z)N(z)$, where $S(z)$ is the desired signal and $N(z)$ is the noise. $G(z)$ represents the effects of room reverberations and small deviations in the desired speaker's position from the straight ahead look-direction and $H(z)$ depends on the location of competing speakers or noise sources. Here we assume that the desired speaker is straight ahead of the hearing aid wearer (*i.e.* $G(z)$ is close to unity). This situation is very common in multi microphone hearing aids. Therefore the problem now becomes one of obtaining a signal that is correlated either with the desired signal or noise by manipulating the two signals $X_1(z)$ and $X_2(z)$. Peterson [2] has proposed an approach where, by subtracting the reference signal from the primary input a noise-only signal is obtained for straight ahead targets in an anechoic field ($G(z) = 1$). A similar technique was adopted by Farassopoulos [3] to cancel interfering speakers. Both systems produce good results when the desired signal is identical in both inputs. Frost [4] has proposed a constrained LMS filter to reduce noise when the desired direction is known and the signal is identical in all the inputs. But real environments are not anechoic and as a result the signal is not identical in all the inputs ($G(z) \neq 1$). Hence performance of the above methods is degraded and signal cancellation or signal distortion occurs [1]. Strube [5] has presented a frequency domain approach to obtain a noise reference in reverberant environments. By taking the weighted sum of the inputs a reference signal correlated only with noise was derived. These optimum weights were estimated by minimising output power when the desired speaker was speaking alone. But the processed speech was observed to contain some disturbing artifacts.

In this paper we present a simple two-stage filtering approach to suppress interference in reverberant environments. The purpose of the first stage is to obtain a signal that is highly correlated with $S(z)$ but uncorrelated with $N(z)$. This is achieved by adjusting the first stage filters during quiet periods so that the interference is completely suppressed. Then the filter is locked to be used when the desired speaker is present. In subsequent processing the first stage removes the noise and produces a distorted version of the signal, $I(z)S(z)$ where $I(z)$ represents the frequency dependent distortion. The

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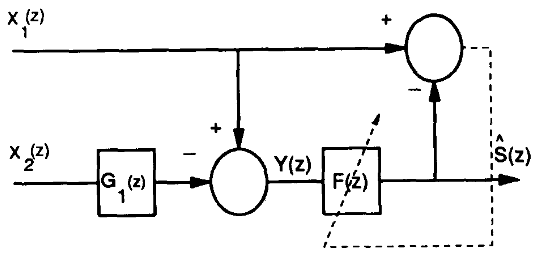

Figure 1: Unconstrained Adaptive Filtering

second stage then tries to remove this distortion by minimising the mean squared error (MSE) between the primary input and filtered reference signal.

2 UNCONSTRAINED ADAPTIVE FILTERING

The block diagram of this structure is shown in Fig. 1. The z-transform of the microphone signals are:

$$\begin{aligned} X_1(z) &= S(z) + N(z) \\ X_2(z) &= G(z)S(z) + H(z)N(z) \end{aligned}$$

In the first stage $G_1(z)$ is adjusted during quiet periods so that noise is completely removed. The optimum filter that minimises $E[|Y(z)|^2]$ is given by

$$G_1(z) = \left.\frac{\Phi_{x_1x_2}(z)}{\Phi_{x_2x_2}(z)}\right|_{\text{in quiet}} \tag{1}$$

where $\Phi_{x_ix_j}$ is the crosscorrelation power density between inputs i and j. For the given signal model

$$G_1(z) = \frac{1}{H(z)} \tag{2}$$

Therefore when signal is present

$$\begin{aligned} Y(z) &= (1 - \frac{G(z)}{H(z)})S(z) \\ &= I(z)S(z) \end{aligned}$$

where $I(z) = 1 - \frac{G(z)}{H(z)}$. Now $Y(z)$ is fed to the cascaded second stage filter and distortionless signal is produced when $F(z)$ is adjusted to the optimum Wiener solution given by

$$F(z) = \frac{\Phi_{x_1y}(z)}{\Phi_{yy}(z)} \tag{3}$$

The noise and signal being uncorrelated we obtain

$$F(z) = \frac{1}{I(z)} \tag{4}$$

In practice filters reach only a suboptimal value and this introduces errors in the system. Lets assume that $G_1(z)$ reaches a suboptimal value given by

$$G_1(z) = \frac{(1 + \Delta(z))}{H(z)} \tag{5}$$

where $\Delta(z)$ is the frequency dependent fractional misadjustment. This misadjustment depends on the characteristics of the adaptation algorithm, filter length, etc. Then when signal is present

$$\begin{aligned} Y(z) &= (1 - \frac{G(z)(1 + \Delta(z))}{H(z)})S(z) - \Delta(z)N(z) \\ &= I(z)S(z) + R(z)N(z) \end{aligned}$$

This residual noise $R(z)N(z)$ at the first stage output results in a suboptimal $F(z)$ given by

$$\begin{aligned} F(z) &= \frac{R(z^{-1})\Phi_{nn}(z) + I(z^{-1})\Phi_{ss}(z)}{|R(z)|^2\Phi_{nn}(z) + |I(z)|^2\Phi_{ss}(z)} \\ &= \frac{\beta(z)}{I(z)} \end{aligned} \tag{6}$$

where $\beta(z) = \frac{1 + \frac{R(z^{-1})}{I(z^{-1})\rho_{pri}(z)}}{1 + \frac{|R(z)|^2}{|I(z)|^2\rho_{pri}(z)}}$ and $\rho_{pri}(z) = \frac{\Phi_{ss}(z)}{\Phi_{nn}(z)}$ is the SNR at the primary input. Then

$$\hat{S}(z) = \beta(z)S(z) + \frac{R(z)}{I(z)}\beta(z)N(z) \tag{7}$$

It is clear from Eq. (6) that when the signal is severely attenuated by the first stage ($I(z) << 1$) and $R(z)$ is large, in very low SNR $F(z)$ deviates significantly from the optimum filter response (i.e. $\frac{1}{I(z)}$). This leads to signal distortion and some noise at the output $\hat{S}(z)$. Furthermore Eq. (7) characterises the trade off between noise reduction and signal distortion.

In the next section we present a constrained filtering approach that performs better even when a significant amount of residual noise is present.

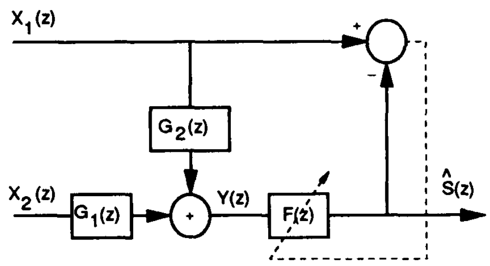

Figure 2: Constrained Adaptive filtering

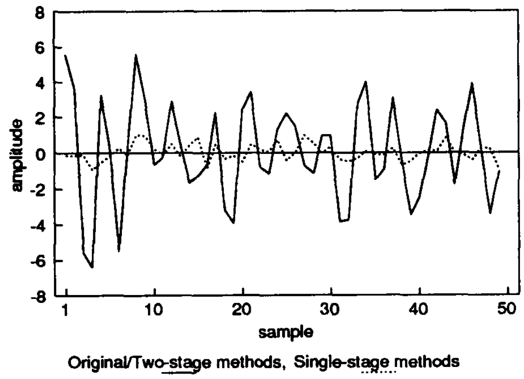

Figure 3: Reconstructed signals

3 CONSTRAINED ADAPTIVE FILTERING

Fig. 2 shows the block diagram of this second approach. Here the second stage is identical to the second stage in the first structure. But in the first stage we choose two filters $G_1(z)$ and $G_2(z)$ to minimise the output power during quiet periods subject to the constraint that they add up to unity (*i.e* $G_1(z) + G_2(z) = 1$). This avoids the filters collapsing to zero and minimises the interference without drastically attenuating the desired signal.

In the ideal case the optimum filters are given by

$$G_1(z) = \frac{1}{1 - H(z)}$$
$$G_2(z) = \frac{-H(z)}{1 - H(z)}$$

But as mentioned earlier filters only attain a suboptimal value. Given that fractional misadjustment in $G_1(z)$ is $\Delta(z)$ we have

$$G_1(z) = \frac{1 + \Delta(z)}{1 - H(z)} \quad (8)$$
$$G_2(z) = \frac{-(H(z) + \Delta(z))}{1 - H(z)} \quad (9)$$

then when signal is present

$$\begin{aligned} Y(z) &= (G_1(z)G(z) + G_2(z))\,S(z) \quad + \\ &\quad \left(H(z)\frac{1 + \Delta(z)}{1 - H(z)} - \frac{H(z) + \Delta(z)}{1 - H(z)}\right) N(z) \\ &= I(z)S(z) - \Delta(z)N(z) \end{aligned}$$

We find that $R(z) = -\Delta(z)$ as in the previous method but now $I(z)$ is constrained to be close to unity. Therefore we expect that in low SNRs $\frac{R(z)}{I(z)\rho_{pri}(z)}$ would be significantly smaller in this set up than in the unconstrained approach. Hence deviations in $F(z)$ will be smaller and as a result there will be less signal distortion and noise in the processed signal.

4 RESULTS

These algorithms were tested in simulated acoustic environments at different SNR and the results were compared to those obtained with other approaches. The proposed configuration produced more than 20dB noise reduction when the signal was identical in both inputs ($G(z) = 1$). Similar performance was obtained with Peterson's approach. But at high SNR impressive results were obtained with two-stage filtering when the inputs contained signals that are not identical. While other approaches produced significant signal distortion the proposed structure reconstructed the original signal very well with a slight increase in the number of computations and complexity (see Fig. 3). This small increase in computational load was more than compensated for by the superior noise reduction achieved when $G(z) \neq 1$.

Performance of the unconstrained (Method 1) and constrained (Method 2) approach were compared at different SNR. In high SNR performance was similar, but at very low SNR the second approach significantly out performed the first as predicted. This is well illustrated in Fig. 4. Here error $= |1 - \beta(z)|$ and represents the deviations in $F(z)$ from the optimum. For a given amount of error the operating

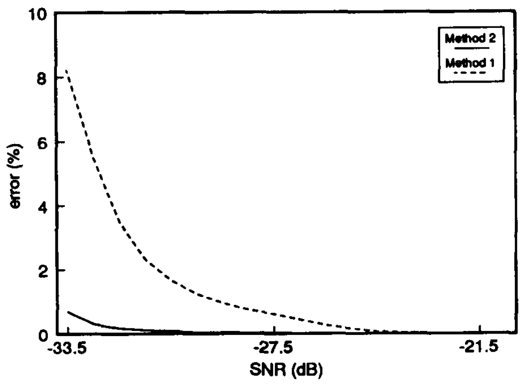

Figure 4: Deviations from the optimum filter response

region of the second method was increased by 5-6dB compared to the first method.

Furthermore, it was observed that the constrained method required twice as many taps as the first one to achieve similar noise reduction performance. This can be explained by considering the nature of the second algorithm in detail. In constrained adaptive filtering a processor having K inputs and J taps per input has KJ weights and requires J constraints to perform satisfactorily. Therefore, only the remaining $(KJ - J)$ degrees of freedom in choosing the weights may be used to minimise the noise power. In the unconstrained approach all KJ degrees of freedom are utilised in minimising noise but signal distortion is not controlled. For the two input system $(K = 2)$ the number of degrees of freedom is reduced by a factor of two, hence the need for more taps in the second structure.

5 CONCLUSIONS

In this paper we have proposed two adaptive structures that produce good noise cancellation in reverberant environments. The results obtained indicate that two-stage filtering is superior to other existing methods in reverberant acoustic fields. Furthermore, this method eliminates the need for the signal to be identical in all the inputs and hence prior knowledge about signal direction and prefilters. Performance of the first structure is slightly degraded in very low SNR when a significant amount of residual noise is present. We have shown that the second structure overcomes this problem by constraining the signal distortion at the first stage to be small.

Further research is required to assess the performance of this approach with real signals and the resulting improvement in intelligibility for the hearing impaired.

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