PERFORMANCE EVALUATION OF A MODIFIED SUBBAND NOISE CANCELLATION SYSTEM IN A NOISY ENVIRONMENT

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Abstract: This paper presents a subband noise canceller with reduced residual noise. The canceller is developed by modifying and optimizing an existing multirate filter bank that is used to improve the performance of a conventional full-band adaptive filtering. The proposed system is aimed to overcome problems of slow asymptotic convergence and high residual noise incorporating with the use of oversampled filter banks for acoustic noise cancellation applications. Analysis and synthesis filters are optimized for minimum amplitude distortion. The proposed scheme offers a simplified structure that without employing cross adaptive filters or stop band filters reduces the effect of coloured components near the band edges in the frequency response of the analysis filters. Issues of increasing convergence speed and decreasing the residual noise at the system output are addressed. Performance under white and coloured environments is evaluated in terms of mean square error MSE performance. Fast initial convergence was obtained with this modification. Also a decrease in the amount of residual noise by approximately 10dB compared to an equivalent subband model without modification was reachable under actual speech and background noise.

1 INTRODUCTION

Subband adaptive filtering using multirate filter banks has been proposed in recent years to speed up the convergence rate of the least mean square LMS adaptive filter and to reduce the computational expenses in acoustic environments (Petraglia and Batalheiro, 2008). In this approach, multirate filter banks are used to split the input signal into a number of frequency bands, each serving as an input to a separate adaptive filter. The subband decomposition greatly reduces the update rate of the adaptive filters, resulting in a much lower computational complexity. Furthermore, subband signals are often downsampled in a subband adaptive filter system, this leads to a whitening effect of the input signals and hence an improved convergence behavior.

In critically sampled filter banks, where the number of subbands equals to the downsampling factor, the presence of aliasing distortions requires the use of adaptive cross filters between subbands (Petraglia et al., 2000). However systems with cross adaptive filters generally converge slowly and have high computational cost, while gap filter banks produce spectral holes which in turn lead to significant signal distortion. Problems incorporating with subband splitting have been treated in literature regarding issues of increasing convergence rate (Lee and Gan, 2004), lowering computational complexity (Schüldt et al., 2000) and reducing input/output delay (Ohno and Sakai, 1999).

Oversampled filter banks has been proposed as the most appropriate solution to avoid aliasing distortion associated with the use of critically sampled filter banks (Cedric et al., 2006) . However this solution implies higher computational requirements than critically sampled one. In addition, it has been demonstrated in literature that oversampled filter banks themselves color the input signal, which leads to under modelling (Sheikhzaheh et al., 2003). These problems can be traced back to the fact that oversampled subband input will likely generate an ill-conditioned correlation matrix (Deleon and Etter, 1995). In this case, the small eigenvalues are generated by the roll off of the subband input power spectrum. A pre-emphasis filter for each subband is suggested by (Tam et al., 2002) as a remedy for this slow asymptotic convergence. An alternative approach to remove the band edge components might be the use of a

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bandstop filters. But this has the undesirable effect of introducing spectral gaps in the reconstructed full band signal. Furthermore it was proved by (Sheikhzaheh et al., 2003) that the introduction of pre-emphasis filters has no considerable effects on the convergence behaviour of a subband noise canceller.

In this paper an alternative procedure is adopted to improve the performance of a noise cancellation system aimed to remove background noise from speech signals. Different prototype filters are used in the analysis and synthesis filter banks. The analysis prototype filter is modified so that the coloured components near the band edges are removed by synthesis filtering, and then the analysis/synthesis filter bank is optimized in the input/output relationship to achieve minimum amplitude distortion. Compared to literature designs (Deleon and Etter, 1995) and (Cedric et al., 2006), this paper bears three differences: first, different methods used for the design of analysis and synthesis filter banks, second, optimization is performed to reduce amplitude distortion with negligible aliasing error due to the use of highly oversampled filter banks and third, the resulting design is implemented efficiently for the removal of background noise from speech signals.

The proposed modified oversampled subband noise canceller offers a simplified structure that without employing cross-filters or gap filter banks decreases the residual noise at the system output. Issues of increasing convergence rate and reducing residual noise on steady state are addressed. Performance under white and coloured environments is evaluated in terms of mean square error MSE convergence. Comparison is made with a conventional full band scheme as well as with a similar system with no modification. The paper is organized as follows: in addition to this section, section 2, formulates the subband noise cancellation problem, section 3 describes the optimum analysis/synthesis filters design, section 4 presents simulation results with discusses the main aspects of the results and section 5 warps up the paper with concluding remarks.

2 SUBBAND NOISE CANCELLER

The original noise cancellation model described in (Sayed, 2008) is extended to subband configuration by the insertion of analysis/synthesis filter banks in signal paths, as depicted by Figure 1.

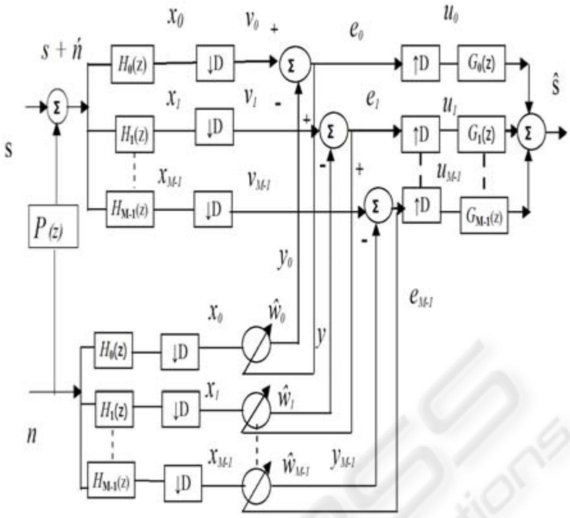

Figure 1: The subband noise canceller.

Assigning uppercase letters for z-transform representation of variables and processors, the noisy speech and the background noise are split into subbands by two sets of analysis filters. The subband analysis filters is expressed in z-domain as

$$H_k(z) = \sum_{m=0}^{L-1} h_k(m) z^{-m} \tag{1}$$

for *k=0,1,2,....M-1*.

Where k is the decomposition index, *h(m)* is the impulse response of a finite impulse response filter FIR , m is a time index, M is the number of subbands and L is the filter length. Now, consider the adaptation process in each individual branch according to Figure 1, and let us define *e(m)* as the error signal, *y(m)* is the output of the adaptive filter calculated at the downsampled rate, $\hat{\mathbf{w}}(m)$ is the filter coefficient vector at *mth* iteration, μ is the adaptation step-size factor , α is proportional to the inverse of the power input to the adaptive filter, and m is a time index, then we have

$$y_k(m) = \hat{\mathbf{w}}^T{}_k(m)\mathbf{x}_k(m) \tag{2}$$

$$e_k(m) = v_k(m) - y_k(m) \tag{3}$$

$$\hat{\mathbf{w}}_k(m+1) = \hat{\mathbf{w}}_k(m) + \mu_k \alpha_k e_k(m)\mathbf{x}_k(m) \tag{4}$$

Relation (4) represents the subband normalized LMS update of the branch adaptive filters. In z-domain, relation (3) can be expressed as

$$E_k(z) = V_k(z) - Y_k(z) \tag{5}$$

Where $V_k(z)$ represents the downsample subband noisy signal, and $Y_k(z)$ is the output of the subband adaptive filter. The aim of the adaptive process is to surpress the noisy component in $V_k(z)$ by equating it to $Y_k(z)$ leaving the subband input $S_k(z)$ undistorted. Each subband error signal is then interpolated by upsampling and synthesis filtering. The final output can be expressed as

$$\hat{S}_k(z) = \sum_{k=0}^{M-1} G_K(z) U_k(z) \qquad (6)$$

Where $U_k(z)$ represents the upsampled version of the subband error signals $E_k(z)$.We assume a highly oversampled filter bank, say two fold , the aliasing components in (6) can be considered to be very small and therefore neglected, hence the final system equation can be represented as

$$\hat{S}_k(z) = S(z) \sum_{k=0}^{M-1} G_K(z) H_k(z) \qquad (7)$$

If we constrain the analysis and synthesis filters $H_k(z)$ and $G_k(z)$ respectively to be linear phase finite impulse response FIR filters, then the term $\sum_{k=0}^{M-1} G_K(z) H_k(z)$ in equation (7) describes the amplitude distortion function of the system.

The branch filters $H_K(z)$, and $G_K(z)$, can be derived from single prototype filters $H_0(z)$, $G_0(z)$, according to $H_k(z) = H_0(zW_M^k)$ and $G_k(z) = G_0(zW_M^k)$, where $W_M = e^{-j2\pi/M}$.

This way, a uniform discrete Fourier transform DFT filter banks are created. Let *A(z)* be the distortion function ,in frequency domain, *A(z)* can be represented as

$$A(e^{j\omega}) = \sum_{k=0}^{M-1} H_k(e^{j\omega}) G_k(e^{j\omega}) \qquad (8)$$

The objective is to find prototype filters $H_0(e^{j\omega})$ and $G_0(e^{j\omega})$ to minimize $A_d(z)$ according to

$$A_d = (1 - |A(e^{j\omega})|) \qquad (9)$$

Ideally A_d should be zero i.e. a perfect reconstruction filter bank. Relaxing the perfect reconstruction property, we can tolerate small amplitude distortion; and then we can have frequency selective filters in a near perfect reconstruction NPR filter bank.

3 ANALYSIS/SYNTHESIS PROTOTYPE FILTER DESIGN

The aim of the prototype filter design is to build a complementary analysis and syntheses filter banks, so that the reconstructed output signal has a very low distortion. Different prototype filters for analysis and synthesis filter banks are designed. The analysis and synthesis prototype filters are selected as in Figure 2. The cut off frequency f_c of the analysis prototype filter is given by

$$f_c = (f_{pa} + f_{sa})/2 \qquad (10)$$

provided that $f_{pa} > f_{ss}$.

Where f_{pa} is the end of the passband of the analysis filter, f_{sa} is the beginning of the stopband of the analysis filter; f_{ss} is the beginning of the stopband of the synthesis filter.

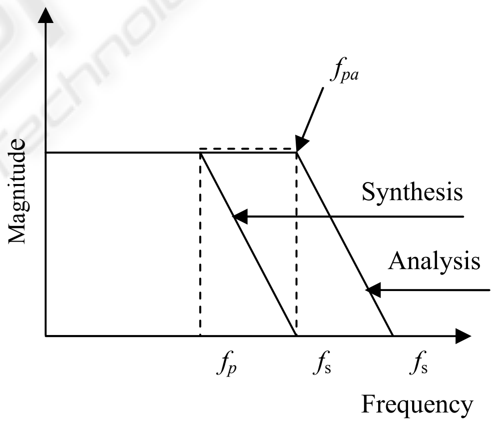

Figure 2: Analysis/Synthesis filters design.

The 3dB down of the prototype synthesis filter (normalized) (*1 /2 M*) is determined by the number of subbands *M*, whereas the analysis filter bandwidth is larger and only limited by the decimation factor *D* according to

$$D \leq \frac{2}{f_{pa} + f_{sa}} \qquad (11)$$

where *D* is the largest integer less than or equal to the right hand side term of (11), f_{pa} and f_{pa} are normalized to the sampling frequency. Values for *D* such that the ratio M/D =2 i.e. two fold over sampled found to be the best in terms of alias cancellation.

The design algorithm starts by designing cut off frequencies for an arbitrary analysis and synthesis prototype filters. The analysis prototype filter is optimized using Parks McClellan algorithm to meet requirements in (10). With analysis prototype filter fixed, the synthesis prototype filter is optimized by minimizing the distortion function given by (9) over the frequency grid in the band [0-π].Figure 3 depicts steps the design procedure and Figure 4 shows a reconstruction error comparison between the modified oversampled filter bank and an equivalent conventional oversampled filter bank.

1. Specify design parameters, number of subbands, M, subsampling factor, filter orders.
2. Design cut off frequencies for prototype synthesis filter
3. Optimize analysis filter with Parks McClellan algorithm satisfying condition in (10).
4. Insert analysis filter in the distortion function, A_d
5. With analysis prototype filter fixed, find the optimum synthesis prototype filter with Hamming window function for which A_d is minimum for certain number of subbands and Hamming window factor β.
6. If satisfied, store analysis and synthesis filter coefficients for later use in subband noise cancellation.
7. Go to step 1 for different optimization parameters

Figure 3: Steps of design algorithm.

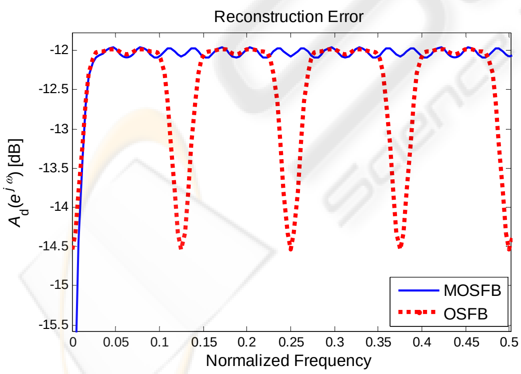

Figure 4: Reconstruction error comparison of the modified filter bank MOSFB and an equivalent conventional oversampled filter bank OSFB.

Analysis and synthesis filter banks can be implemented efficiently using polyphase representation of a single prototype filter followed by fast Fourier transforms FFT.

4 RESULTS AND DISCUSSION

The noise path used in these tests is an approximation of a small room impulse response modelled by a FIR processor of 256 taps. The number of subbands $M=8$, downsampling factor $D=4$, prototype filters order is 128.To measure the convergence of the subband noise canceller, a variable frequency sinusoid was corrupted with white Gaussian noise that was passed through a transfer function representing the acoustic path. The corrupted signal was then applied to the primary input of the noise canceller; regarding zero mean, white Gaussian noise is applied to the reference input. A subband power normalized version of the LMS algorithm is used for adaptation. Mean square error MSE convergence is used as a measure of performance. Plots of MSE were produced and smoothed with a suitable moving average filter. A comparison is made with a conventional fullband system as well as with an oversampled system without modification. The unmodified oversampled subband noise canceller is denoted with OSSNC and the modified system is denoted with MOSSNC. Results are depicted in Figures 5 and 6.

To test the behaviour under real environmental conditions, a speech signal is then applied to the primary input of the proposed noise canceller. The speech is a Malay utterance “Kosong, Satu, Dua, Tiga” spoken by a woman, sampled at 16 kHz. Different types of background interference were used to corrupt the aforementioned speech. MSE plots are produced for two cases: Figure 7 for the case of machinery noise as background interference and in Figure 8 for the case of a cocktail party i.e. disturbance by another speech.

Apart from the fast initial convergence, it is clear from Figure 5, that the mean square error (MSE) plot of the oversampled subband system OSSNC levels off quickly before the MSE plot of the fullband system. This is obviously due to the inability to properly model the presence of coloured components near the band edges of the filter bank. During initial convergence the subband system performs better than the fullband system but is less effective afterward. On the other hand, the MSE convergence of the modified system MOSNC outperforms that of the fullband system during initial convergence and exhibits comparable steady state performance as shown by Figure 6. It is obvious that while the MSE of the fullband system converging in slow asymptotic way, the MOSNC system reaches a steady in 2000 iterations. The fullband system needs more than 6000 iterations to reach the same noise cancellation level. The main difference between Figure 5 and Figure 6 is in the amount of residual noise which has been g reduced with the MOSSNC. Results obtained for actual speech and background noise (Figures 7 and 8) prove that the fullband system cannot model properly with coloured noise as the input to the adaptive filters, and the residual error can be sever when the environment noise is highly coloured. Tests performed in this part of the experiment proved that the MOSSNC does have improved performance compared to OSSNC and the full-band model. Depending on the type of the interference, the improvement in reducing resedual noise n varies from 15-20 dB better than full-band case.

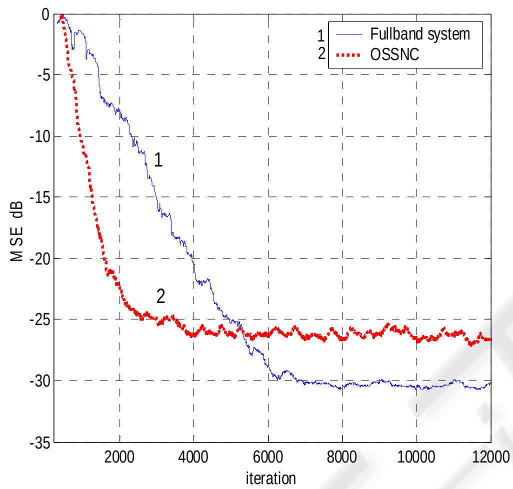

Figure 5: MSE convergence behaviour of OSSNC under white Gaussian noise.

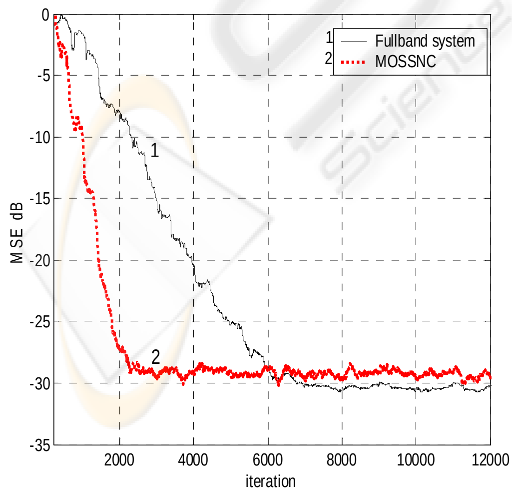

Figure 6: MSE convergence behaviour of MOSSNC under white Gaussian noise.

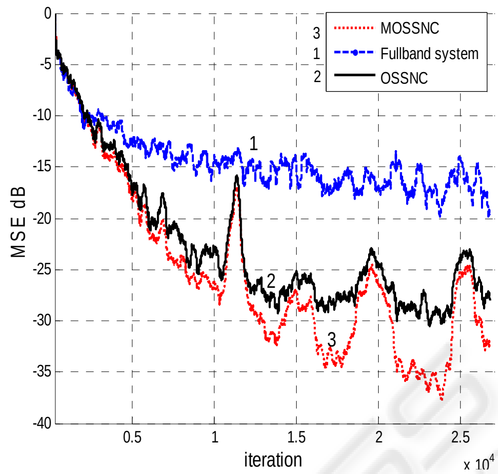

Figure 7: Performance comparison under actual speech and machinery noise.

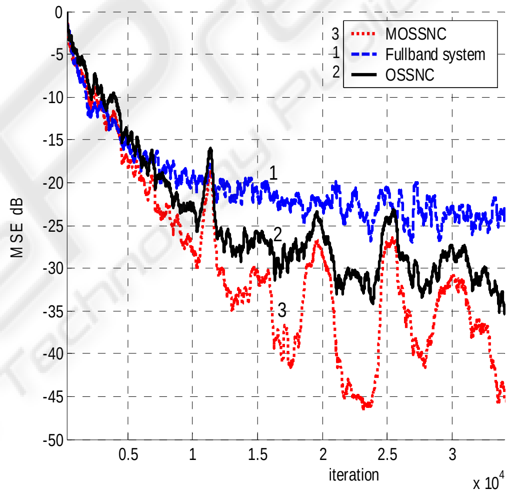

Figure 8: Performance comparison under actual speech disturbed by another speech.

5 CONCLUSIONS

In this work, an oversampled subband noise canceller with modified filter bank is developed to overcome the problem of slowly converging components associated with the usual oversampled subband scheme. An efficient optimized DFT filter bank is used in the canceller. The analysis filter bank is modified to remove slowly converging components near band edges, while the synthesis filter bank is optimized to minimize input/output distortion.The modified system has shown improved performance compared to a similar scheme with conventional oversampled filter bank. The convergence behaviour under white and coloured environments is greatly improved. The amount of residual noise is reduced by 15-10dB under actual speech and background noise. The next logical step is to realize this system on a suitable DSP processor such as the Texas C6000 to prove the validity of the method for noise cancellation. Also, other types of filter banks and transforms can be investigated and used for the same purpose.

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