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Improved, Low Complexity Noise Cancellation Technique for Speech Signals

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Abstract: A noise cancellation system with an improved performance and low computational costs is presented in this paper. In speech applications, slow convergence and high computational burden are the main problems incorporating with conventional noise cancellation method. The proposed noise canceller is based on using multirate filter bank to split the spectrum of the input signals and uses the least mean square (LMS) algorithm in branches to control a finite impulse response (FIR) filter to reduce the noise in the input noisy speech. The computational power is greatly reduced by polyphase implementation and the noble identities. Direct and polyphase models were developed, tested and compared to the equivalent full band model. The proposed scheme shows better convergence behavior compared to classical approach with 50% reduction in computational complexity.

Key words: Noise cancellation · Adaptive filtering · Multirate signal processing · Filter banks · FIR filters

INTRODUCTION

The technique of multirate filtering has been used over the past few years with attention being given to subband decomposition. Subband signal representations are obtained from multirate filter banks [1]. In this technique we encounter time-varying linear systems such as decimators, interpolators and modulators [2].

In applications such as the elimination of background noise from speech signals, a very long adaptive filter length required due to the requirement to model very long acoustic path impulse response. Also, the convergence and tracking of a gradient based adaptive filters such as the least mean square LMS algorithm can be very slow if the input signal has wide spectral dynamic range such as that found in speech [3]. As the length of the adaptive filter increases the system becomes computationally expensive. In many situations we assume limited storage and computational resources, as the design may be realized in limited resources processors. The above two problems are intimately connected and they impose a demand for high computational and storage requirements. A technique used to overcome the above problems is to split the signal into subbands and adapt each subband signal using separate adaptive filter [4]. The filter bank is the primary tool used to perform the decomposition. The first benefit of this system results from the fact that the subband adaptive filters are shorter in length than the equivalent full-band adaptive filter and operate at a downsampled rate. The second benefit comes from the subband decomposition of the input signal, by decomposing the signal; the adaptive filter operates on a smaller bandwidth and can be adjusted to take advantage of this [5]. This paper presents two multirate noise cancellation models. Both models are two band schemes, the filter bank of the first model is implemented using direct quadrate mirror filter arrangement, to reduce computational power this model is implemented using polyphase structure of the quadrature mirror filter QMF [6]. Models were built, tested and assessed using computer simulations. The paper is organized as follows: in addition to this section, section 2 describes the proposed methodology; section 3 presents simulation results and section 4 warps up the paper with conclusions of the main aspects.

MATERIALS AND METHODS

A simplified diagram of the proposed structure is shown at Fig. 1, this is a two sensor scheme, it consists of three sections: analysis which contains analysis filters $H_o(z)$, $H_1(z)$ plus the down samplers, the adaptive section

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contains two adaptive FIR filters with two controlling LMS algorithms and the synthesis section which comprises of two upsamplers and two interpolators $G_o(z)$, $G_1(z)$.The noisy speech is fed from the primary input, whilst, the noise n' is fed from the reference input sensor, n' is added to the speech signal via a transfer function A(z) which represents the acoustic noise path, thus n' correlated with n and uncorrelated with s. In stable conditions, the noise n should be cancelled completely leaving the clean speech as the total error signal of the system [7]. The system equation is formulated as follows:

The transfer function of the kth analysis filter $H_k(z)$ is given by:

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

Where:

$$h(m) = [h(0), h(1), h(2), \ldots\ldots h(L)] \quad (2)$$

Where m is a time index and L is the filter order. The analysis prototype filter H(z) can be represented in polyphase components as follows;

$$H(z) = \sum_{K=0}^{M-1} z^{-k} P_k(Z^M) \quad (3)$$

Where $P_k(z)$ is the kth polyphase component of the prototype filter, M is the number of subbands, in this case M=2. The impulse response of $P_k(z)$ is given by:

$$p_k(m) = h(mM + k)\text{, for k=0, 1} \quad (4)$$

The branch update of the subband adaptive filters with LMS adaptation algorithm is given by:

$$w_k(m+1) = w(m) + \mu_k e_k(m) n'(m) \quad (5)$$

Where, w is the filter coefficient vector, e is the error signal and μ is the adaptation step size. The subband error signal e_k is calculated as follows:

$$e_k(m) = d_k(n) - y_k(m) \quad (6)$$

Where, d is the desired input i.e. the subband speech input (s+n), y is the output of the adaptive filter and k is the decomposition index. The synthesis filters G(z) can be expressed in the same way as the analysis filter, in poly-phase form, using type 2 representations:

$$G(z) = \sum_{k=0}^{M-1} z^{-(M-1-k)} P_k(Z^M) \quad (7)$$

The main problem with filter bank is aliasing distortion due leakage between adjacent bands. For alias free system, filters should possess the following relationships among each other [6]:

$$G_0(z) = H_1(z),\ G_1(z) = -H_0(-z) \quad (8)$$

In transform domain the whole system can be expressed as

$$\sum_{0}^{M-1} G_k(z) E_k(z)\ S_k(z) = z^{-\partial} \hat{S}(z) \quad (9)$$

where ∂ is the total system delay, E(z) is the z-transform of e(n), $\hat{S}(z)$ is a delayed version of the input speech.

Here, we assume that the two sensors are physically separated and isolated from each other so that no substantial speech leakage into the reference input otherwise intelligibility of the speech signal will be degraded. Omnidirectional microphones are traditionally used to prevent speech leakage into the reference input. Also acoustic barriers can be used for this purpose [8].

In practical implementation of modern noise cancellers, a voice activity detector VAD is used to suspend adaptation during periods of speech activity thus prohibiting the input noise from containing some strength of actual speech signal [9].

The first step in constructing the two band multirate canceller is the design the prototype low-pass filter low pass filter LPF $H_0(z)$, this filter was designed using optimization method presented in [6],the optimization performed by minimizing the objective function that is given by the following:

$$\alpha \int_{\omega s}^{\pi} \left|H_0(e^{j\omega})\right|^2 d\omega + (1-\alpha) \int_{0}^{\pi/2} [1 - \left|T(e^{j\omega})\right|^2]^2 d\omega$$
$$0 < \alpha < 1 \quad (10)$$

where ω is the stopband angular frequency and $T(e^{j\omega})$ is a distortion function. Specifications of the prototype filter are given by Table 1, this filter was implemented in a directly connected analysis/synthesis filter bank configuration; a speech signal was applied to the input of the filter bank to verify reconstruction capability. In the first case, a direct implementation model of a quadrature mirror filter QMF as shown in Fig. 2, in a

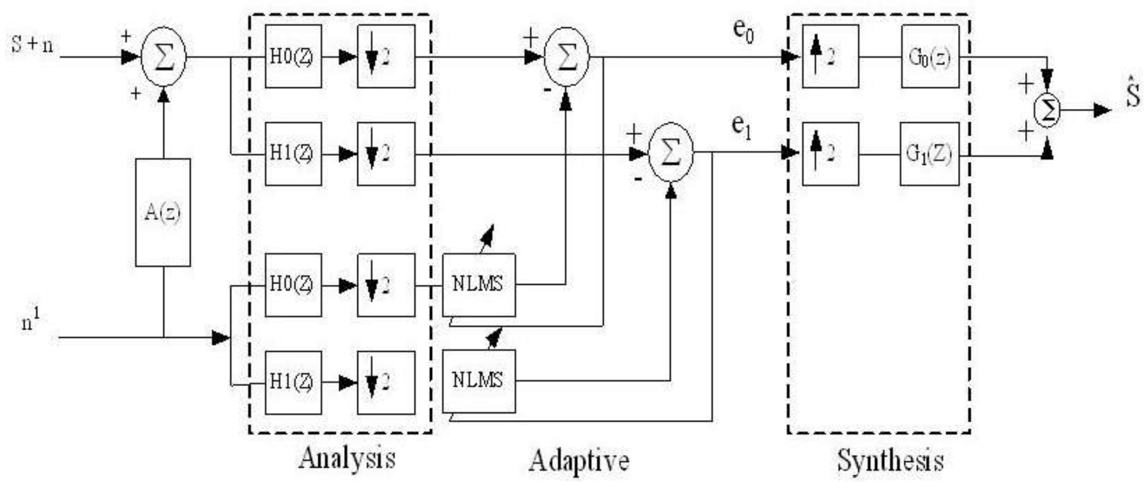

Fig. 1: Proposed structure

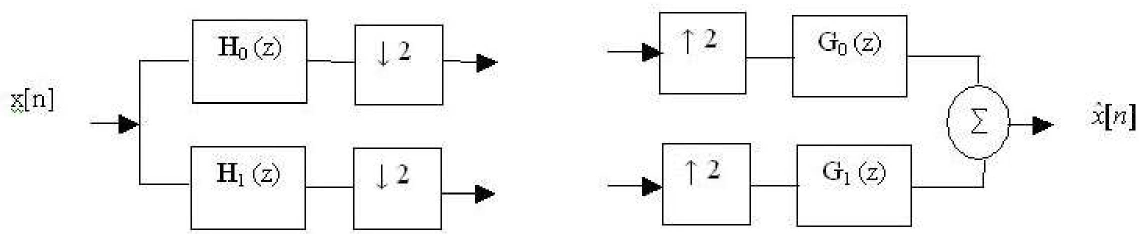

Fig. 2: Direct implementation of QMF

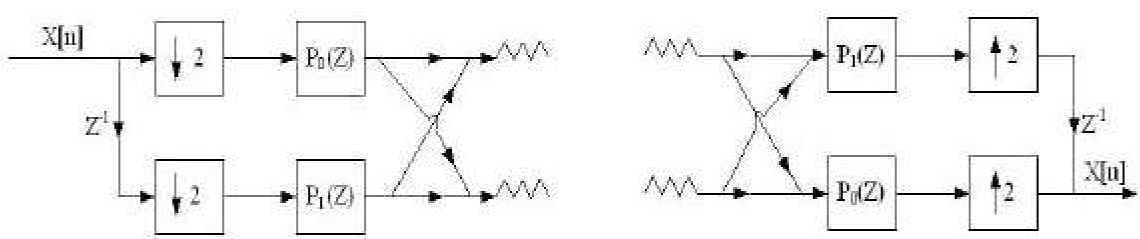

Fig. 3: Polyphase arrangement

Table 1: Prototype filter specifications

Filter order	31
Phase response	Linear
Cut off frequency normalized	0.5
Transition-band width normalized	0.14
Weighting factor α	1
Reconstruction error	0.008 dB
Stop-band attenuation first peak	60 dB
Stop band attenuation least peak	75 dB.

two band noise canceller configuration is constructed. This model was then implemented with computationally efficient polyphase decomposition as shown in Fig. 3, the noble identities are utilized to achieve additional reduction in computational power [10].

RESULTS AND DISCUSSION

The noise path is modeled by FIR filter with 92 tap impulse response. The normalized version of the LMS is used to adapt an FIR filter of 46 tap in each brunch, the step size control μ was chosen to be 0.08 by trial and error. Initially, the proposed model was tested using variable frequency sine wave contaminated with zero mean, unit variance white Gaussian noise. This noise is propagating through a noise path A(z), applied to the primary input of the system. The same Gaussian noise is passed directly to the reference input of the canceller. Figure 4 shows the simulation algorithm and Table 2 lists the various parameters used in the simulations.

In a separate experiment, a speech signal sampled at 8 kHz as shown in Fig. 5 is used for testing. Several types of environmental noise are used to corrupt the speech signal. Convergence behavior using mean square error MSE plots are used as a measure of performance. These plots are smoothed with 200 point moving average filter as shown in Fig. 6 for low frequency sine wave corrupted by white Gaussian noise and in Fig. 7 for speech input corrupted by environmental noise. The processed speech is shown in Fig. 8.

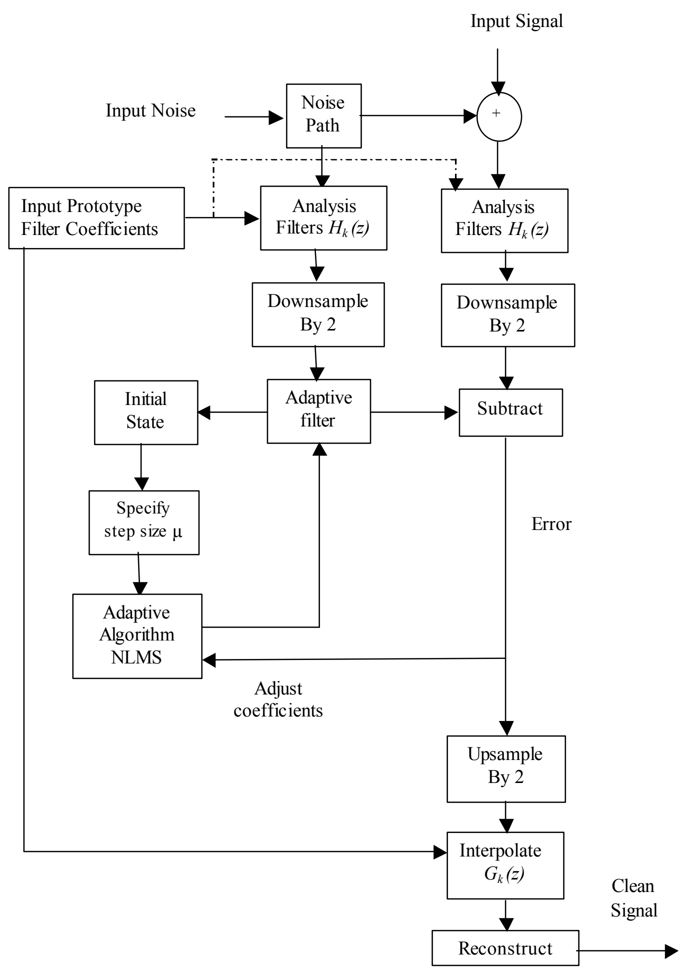

Fig. 4: Simulation algorithm

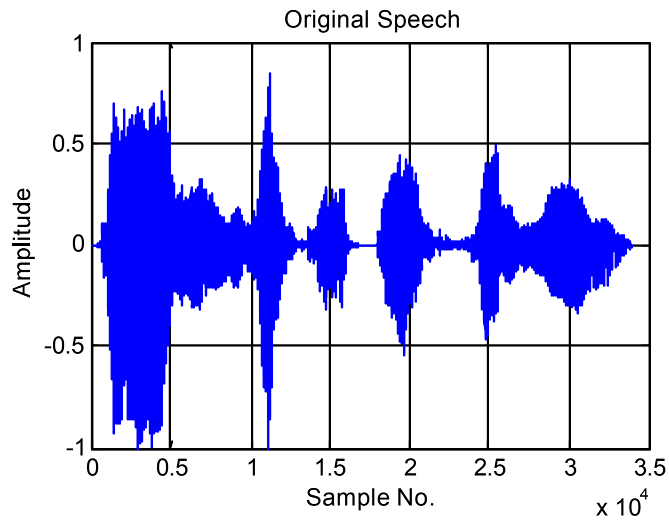

Fig. 5: Speech used in tests

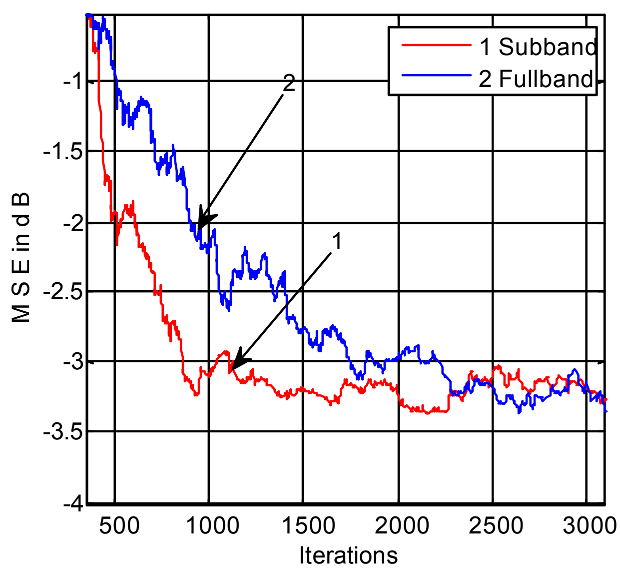

Fig. 6: MSE behavior for variable frequency sine wave and white Gaussian noise

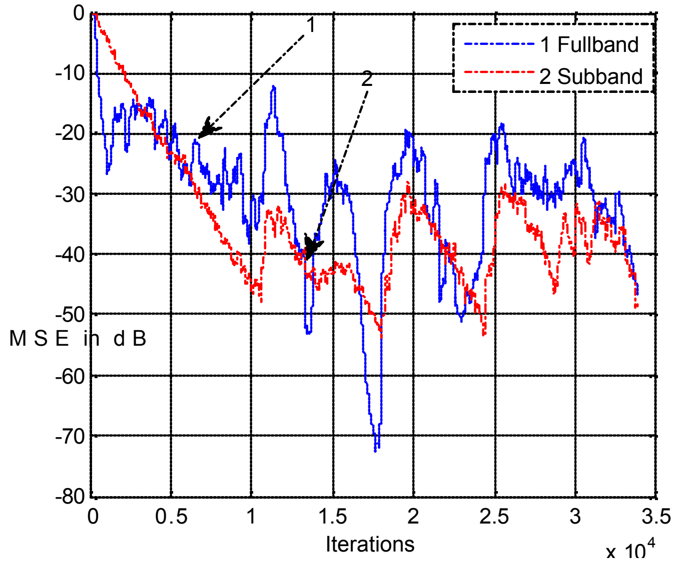

Fig. 7: MSE behavior for speech corrupted by environmental interference

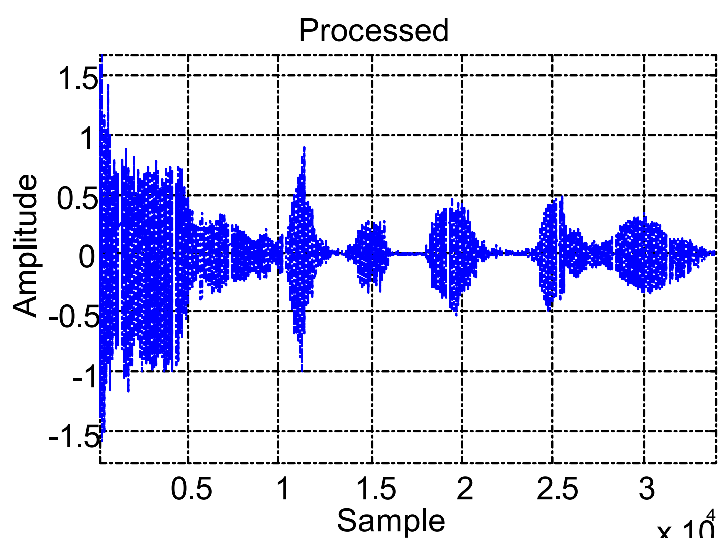

Fig. 8: Reconstructed speech

Table 2: Simulation parameters

Noise path impulse response length	92
Adaptive filter order	45
Step size (μ)	0.08
Sampling frequency	8 kHz
Input (first test)	Variable frequency sine wave
Noise (first test	Gaussian white noise with zero mean and unit variance
Input (second test)	Speech of a woman
Noise (second test)	Ambient
Prototype analysis/synthesis order	31

In comparison to full band system, the convergence speed of the propose system is higher as depicted in MSE plot (Fig. 6),the proposed system reaches a steady state after less than 1000 iterations, while the fullband system approaches the same level after 2500 iterations. Since one of the objectives of this work is to reduce the computatioal expenses, we explain here the calculation of such requirements; the prototype analysis/synthesis filter is of length 32 and since there are a total of two polyphase filters each of length 32/2 = 16 operating at a rate of (1/2) due to downsampling in the filterbank, thus requiring 16 real multiplication per input sample, this operation is performed three times for analysis of noise, the contaminated speech and for the synthesis of the error signal i.e. 48(16X3) multiplication per unit sample, we have the noise path to be modeled is 92 in length, each adaptive filter is of length 92/2 = 46 operating at a down sampled rate. So the total multiplication per input sample in the two-band ploypháse system is = 47 + 48 = 95 operation per input sample compared with 185 multiplication per input sample in the fullband system, there is a savings of more than 50%, therefore in spite of the introduction of analysis/synthesis filter banks there is a considerable savings specially if one considers a very long impulse response and a further decomposition of the two subbands to a higher number of subbands.

CONCLUSIONS

A subband band noise cancellation scheme is proposed, this proposal based on using QMF bank in direct and polyphase implementation. Results were compared with their equivalent fullband model, which was built for the sake of comparison. As far as speed of convergence is concerned, results showed that the subband QMF model outperforms its equivalent classical fullband method. The polyphase arrangements reduce the computational requirements considerably. However the new structure suffers from latency due to the introduction of analysis synthesis filter banks. Amplitude distortion is hardly notice by average listeners.

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