Narrowband Interference Supression in Speech Signals Using Multirate Filter Banks

Srilatha Maganti, Spandana Kolloju, and Anil Kumar Tipparti

***Abstract*—** This paper proposes a perfect reconstruction filter bank based narrow band interference suppression system to eliminate the effects of narrowband interference in a speech signal. The interference suppressor presented in this paper is based on a Cosine modulated filter bank. The Perfect Reconstruction and easy implementation property of the cosine modulated filter bank is considered in designing the analysis and synthesis filter banks. The reconstruction of the original speech signal is done by detecting the interference band from the subband signals at the output of analysis bank and eliminating it in the synthesis filter bank using an M-estimator based iterative method. The performance of the proposed technique method is illustrated through simulation results on a contaminated speech signal.

***Keywords*—** Filter banks, Perfect reconstruction, Modulated Filter banks, Narrow band Interference.

I. Introduction

MULTIRATE signal processing is a fast growing area that enables the use of digital signal processing techniques in applications requiring low-cost and high sample rates. Judicious and creative use of computationally efficient Multirate algorithms in digital signal processing systems allows one to realize a hardware efficient solution, that in many instances would not be feasible using single rate signal processing.

The multirate theory is applied to signal estimation, where one signal is estimated from some other related signal or signals. [1,4] The desired signal may be corrupted by distortion or interference A typical signal estimation application is the recovery of a transmitted signal from a received signal that has been subject to distortion and is corrupted by noise. A number of typical signal estimation applications are presented including the recovery of a transmitted signal from a distorted received signal, subject to amplitude and phase distortions and additive white noise over the communications channel; and image restoration of an image recorded by an imaging system that introduces blurring, nonlinear geometric distortions, and additive white noise. In the multi-channel, multirate optimal filtering problem, there are multiple observation signals and these signals are allowed to be at rates different from the desired signal. The goal is to estimate this underlying signal from the set of related Multirate observation signals.[1] as shown in Fig 1.

During the last several years there has been substantial progress in Multirate system research where in the Multirate filter banks have found various applications in many different areas, such as speech coding, scrambling, adaptive signal processing, image, image compression, as well as transmission of several signals through the same channel Traditionally, a filter bank [2] is a group of filters derived from a prototype filter but working in different frequency ranges. The simplest filter bank separates the input frequencies into two bands, namely a high frequency band and a low frequency band. The bandwidth of sub bands may be same or different depending upon the choice of filter bank; and suitably termed as uniform filter bank or non-uniform filter bank respectively. A two band uniform filter bank that has mirror symmetry in their magnitude characteristics of the sub band filters is called as Quadrature Mirror Filter Bank [2,3].

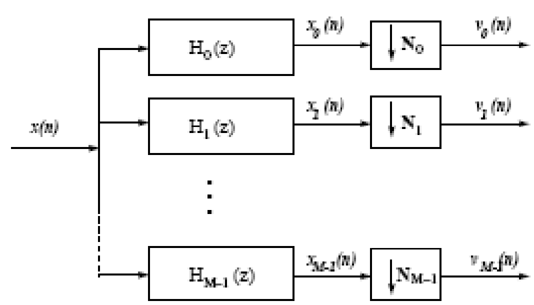

Fig. 1. An M-channel multirate observer system.

A great amount of different filter bank approaches have been developed over last two decades [6]. The Discrete Fourier Transform (DFT) polyphase filter bank has high computational efficiency but, suffers from the fact that it is not able to cancel alias components caused by sub sampling the sub band signals [6]. Among the many filter banks, Cosine Modulated filter banks are optimal filter banks with respect to design ease , implemention and provide Perfect

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Reconstruction(PR) [7].

II. COSINE MODULATED FILTER BANKS

A critically sampled filter bank system is shown in Fig. 2. Modulated filter banks are used to form analysis-synthesis filter banks that divide the received signal into several channels (analysis part), and reconstruct the original signal from the subchannels (synthesis part) [8]. If the filter design parameters are chosen correctly, the filter bank can offer PR, meaning that the output signal is just a scaled and delayed version of the input signal [9]

$$\hat{x}(n) = cx(n-n_0) \tag{1}$$

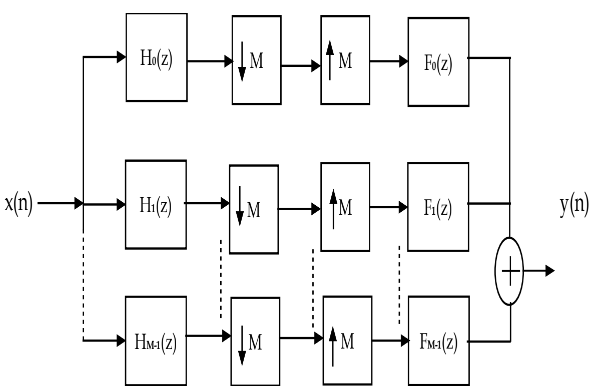

Fig. 2 A critically sampled Multirate filter bank

In a conventional M channel maximally decimated cosine modulated PR or approximately PR system, all filters have the same total bandwidth *2π /M* and their pass-bands do not overlap significantly. When the subband signals are decimated by M**,** aliasing can be eliminated.

The impulse responses of the analysis and synthesis banks $h_k(n)$ and $f_k(n)$ of a Critically sampled FIR filter bank can be generated from cosine modulation versions of the FIR prototype filters h(n)and f(n) given as

$$\begin{aligned} &H(Z) = \sum_{n=0}^{N} h(n)Z^{-n} \\ &and \\ &h(N-n) = h(n) \end{aligned} \tag{2}$$

The impulse response of the modulated filter bank is then,

$$\begin{aligned} h_k(n) &= 2h(n)\cos\left((2k+1)\frac{\pi}{2M}\left(n-\frac{N-1}{2}\right)+\theta_k\right) \\ f_k(n) &= 2f(n)\cos\left((2k+1)\frac{\pi}{2M}\left(n-\frac{N-1}{2}\right)+\gamma_k\right) \end{aligned} \tag{3}$$

Where 0≤n≤N-1, 0≤k≤M-1 and

$$\theta_k = (-1)^k\frac{\pi}{4}, \gamma_k = -(-1)^k\frac{\pi}{4} \tag{4}$$

Then for perfect reconstruction,

$$\begin{aligned} &f_k = h_k(N-n) \\ &and \\ &F_k(Z) = Z^{-N}H_k(Z^{-1}) \end{aligned} \tag{5}$$

III. PROPOSED METHOD

This paper proposes an M-estimator based perfect reconstruction (PR) filter bank based interference suppression system to eliminate the effects of narrowband interference in a speech signal.

The signal with added interference is filtered to sub band signals in a Cosine modulated analysis filter bank. In the frequency domain, the NBI manifests itself as a peak in the spectra. Therefore from the spectral observations of each sub band signal, the interference band can be detected.

The interference suppression technique transforms the received signal into spectral domain and attenuates the coefficients of the filter of the synthesis filter bank recognized to be in the interference frequency band.

This is an iterative method in which the estimated signal spectrum from the output of the synthesis bank is compared with the desired spectrum and the coefficients of the corresponding interference band filter of the synthesis filter bank are adjusted. The iterations may be stopped if the function $\rho(e)$ (known as penalty function), where e is the difference (error) between the actual and the estimated signal, is very small (say 0.001) [10-11]. The derivative of $\rho(e)$ is known as influence function ($\psi(e)$). The technique can be summarized as

$$\text{Minimize:} \quad \rho\left(\widehat{H}(\omega) - H_{\mathrm{d}}(\omega)\right) \tag{6}$$

where δ denotes the upper bound of the squared weighted error(say 0.001), $\hat{H}(\omega)$ is the estimated spectrum of the synthesis signal, $H_d(\omega)$ is the desired spectrum and ρ is a symmetric, positive-definite function with a unique minimum at zero, and is chosen to be less increasing than squared function. Let $C=[c_1,.....c_k]$ be the parameter vector to be estimated. The *M*-estimator of *c* based on the function $\rho(e_i)$ is the vector ***C*** which is the solution of the following *k* equations:

$$\sum_i \psi(e_i)\frac{\partial e_i}{\partial c_j}, \ for\ i=1,...,k \tag{7}$$

Where the derivative $\psi(x) = \dfrac{d\rho(x)}{dx}$ is called the influence function. The influence function $\psi(x)$ measures the influence of a datum on the value of the parameter estimate. In this paper, a new *M*-estimator is proposed for robustifying the algorithm. The penalty function and the influence functions of the proposed *M*-estimator are given by [10] (also see Fig 1).

$$\rho_{PROPOSED}(x) = \begin{cases} \dfrac{x^2}{2}, \; for \quad |x| \le a \\ \dfrac{ab}{2} - a|x|, \; for \quad a < |x| \le b \\ -\dfrac{ab}{2}\exp\left(1 - \dfrac{x^2}{b^2}\right) + d, \; for \quad |x| > b \end{cases} \tag{8}$$

where d is any constant.

$$\psi_{PROPOSED}(x) = \begin{cases} x, \; for \quad |x| \le a \\ a\,\mathrm{sgn}(x), \; for \quad a < |x| \le b \\ \dfrac{a}{b} x \exp\left(1 - \dfrac{x^2}{b^2}\right), \; for \quad |x| > b \end{cases} \tag{9}$$

The choice of the constants $a \left(= kv^2\right)$ and $b \left(= 2kv^2\right)$ depends on the robustness measures derived from the influence function.

IV. Simulation Results

The M channel Nth order Cosine Modulated analysis and Synthesis filter banks for Perfect reconstruction is formed [1] using equation(3) as

$$h_k(n) = 2h_p(n)\cos((2k+1)\frac{\pi}{M}(n - \frac{N}{2}) + (-1)^k \frac{\pi}{4}$$

and

$$f_k(n) = 2h_p(n)\cos((2k+1)\frac{\pi}{M}(n - \frac{N}{2}) - (-1)^k \frac{\pi}{4} \tag{7}$$

where $k = 0,1,2,\ldots\ldots, M-1$ and $n = 0,1,\ldots\ldots\ldots, N$

The prototype filter h(n) is a symmetric lowpass FIR filter, whose bandwidth is π/M

$$\left|H_0(e^{j\omega})\right|^2 + \left|H_0(e^{-j(\omega-\pi/M)})\right|^2 = 1, 0 < \omega < \pi/M \tag{8}$$

and

$$\left|H_0(e^{j\omega})\right| \approx 0, \; \omega > \pi/M \tag{9}$$

In this paper M=16 and N=49 are chosen. The frequency response of the analysis filter bank is shown in the Fig 3. An audio signal clean_speech.wav with spectrum shown in the Fig 4 is applied to the analysis filter bank. An interference of 300 Hz is added to the seventh subband signal. The spectra of the decomposed signals in each band is shown in the Fig. 5. From the spectral plots of the subband signals,, the band of the interference signal can be observed .The signal recovery is done using the synthesis filter bank. The coefficients of the interference affected filter of the synthesis bank channel are adjusted using the proposed technique . The spectrum of the recovered speech is shown in the Fig 6. Simulations are also done adding interference at two different frequencies in two subbands of the decomposed signal .An audio signal clean_speech.wav with spectrum shown in the Fig 7(a) is applied to the analysis filter bank . Interferences at frequencies (300 Hz and 800 Hz) are added to the third subband signal and seventh subband signal. Fig 7(b) The proposed interference suppression technique is applied to the two synthesis filters and the recovered signal is plotted in Fig7(c).It can be observed that the recovered spectrum follows the original.

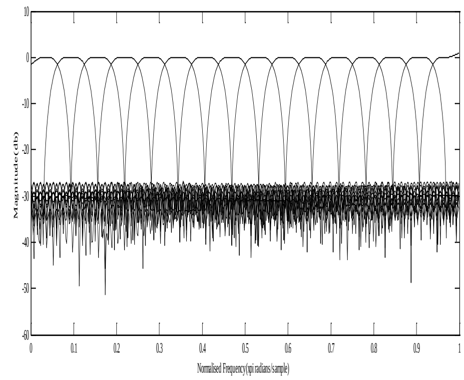

Fig. 3 Magnitude response of CMFB analysis filter bank

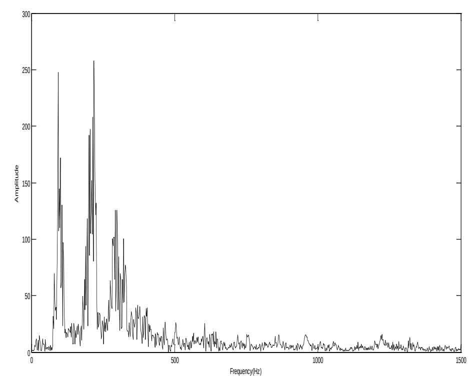

Fig. 4 Spectrum of speech signal clean_speech.wav

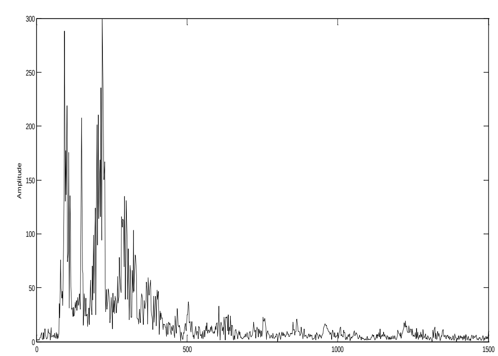

Fig. 5 subband spectra of the decomposed speech signals with interference added in the seventh band

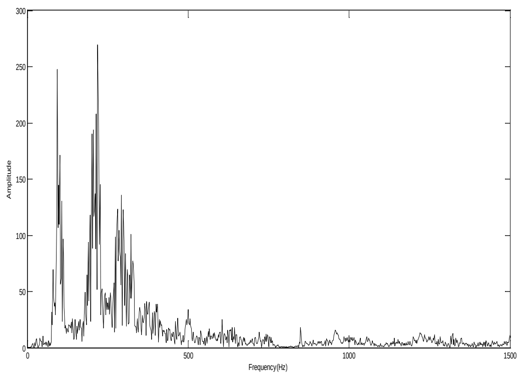

Fig. 6 Spectrum of the recovered speech signal

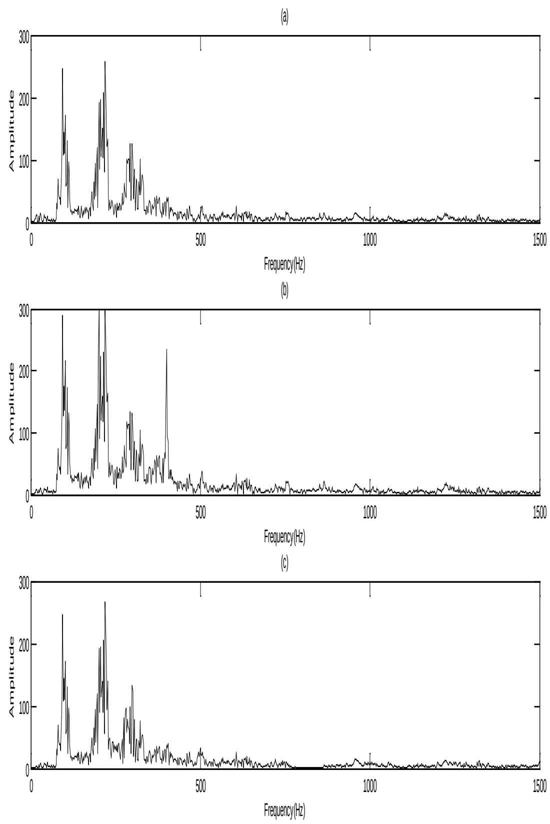

Fig. 7 (a) Orginal speech spectrum (b) subband spectra of the decomposed signal with interference added in two subbands

(c) recovered spectrum

V. Concluding Remarks

In this paper, a filter bank based interference detection and suppression method has been implemented and evaluated. The filter coefficients of the corresponding interference affected subband were adjusted using the proposed technique. The spectrum and the signal recovered follow the original ones. The method worked with interference added in more than one position. The main strengths of the proposed method are the perfect reconstruction property of the filter bank used and its affordable complexity requirements.

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