CASCADE COMBINATION OF WAVELET AND ADAPTIVE FILTER FOR NOISE CANCELLATION

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Abstract: Speech enhancement techniques are very important in the field of signal processing for their numerous applications. They are employed in many contexts such as hands-free telephony, hearing aid systems, re-mastering of audio recordings, preprocessing for speech recognition interfaces, etc. In this paper, an efficient cascade combination approach for cancellation of noises in speech signals is discussed. It is possible to obtain higher performance by cascading two or more algorithms. A better noise removal approach is presented by cascading wavelet and adaptive filters. Results show that the cascade approach gives high Peak Signal to Noise Ratio (PSNR) and low Root Mean Square Error (RMSE) than individual performances of wavelet and an adaptive filter.

Keywords: Wavelet Transform, Adaptive Filters, Noise Cancellation, Least Mean Square Adaptive Filter.

I. INTRODUCTION

Speech enhancement algorithms are used to enhance the speech signals by removing the accumulated noises in the signals. Degenerate Un-mixing Estimation Technique (DUET) based speech enhancement and post signal processing using adaptive noise cancellation technique are explained in [1]. DUET is a generally used for speech babble separation as noise and/or challenging speech as noise from the speech signal. The enhanced approach by combining adaptive noise cancellation and DUET acts as the post-processor.

Short Time Fourier Transform (STFT) based noise cancellation is discussed in [2]. It is developed by mixing the optimal filtering technique and subspace method through joint diagonalization using the noise signal correlation matrix and clean speech signal matrix.

A performance comparison of Least Mean Square (LMS) algorithms are described in [3]. Error Normalized LMS (NLMS) algorithm is discussed in which the variation of step size is inversely proportional to the squared norm of error vectors. The error magnitude needs the signed version of NLMS algorithm. A priori information of a bound is required which is unknown in most applications. Both the signed error and normalization by error function features are used by a very simple algorithm called signed error NLMS.

State space adaptive algorithm for impulsive noise cancellation for speech enhancement is implemented in [4]. The speech signal is enhanced by removing the impulsive noise and which can be used in recognition, medical field and in the case of security purpose. The recursive least square algorithm is used for this purpose, and the state space variant of recursive least square application enhances the result.

A comparative research of different adaptive algorithms for noise cancellation in speech signals is introduced in [5]. Various adaptive filter algorithms are discussed for comparative analysis such as LMS, Block LMS, N-LMS, Block NLMS, Variable step size LMS (VSLMS) and Block VSLMS algorithms.

Optimization algorithms for automatic noise cancellation in speech signals are described [6]. The parameters for adaptive filtering function are generated using the optimization techniques such as artificial bee colony, cuckoo search algorithm. Both techniques are analyzed for different noises at different levels.

A survey about adaptive filtering for noise cancellation is explained in [7]. Adaptive filtering is a very effective technique for noise removal which does not require signal statistics. The improvements in SNR using various adaptive techniques are discussed using the following measures convergence, stability, computation time, and complexity.

New Time Varying LMS (NTVLMS) approach for speech enhancement is discussed in [8]. The performance analysis of NTVLMS algorithm is compared with NLMS, TVLMS, LMS, and robust VSLMS algorithm. The noise cancellation capabilities of aforementioned adaptive techniques are analyzed using speech input signals corrupted with different noises such as conference noise, Gaussian noise, engine noise, and traffic noise.

In [9], noise cancellation for speech enhancement using VSLMS algorithm is implemented. Also, general variable step size adaptive filter is implemented to eliminate the noise effectively. Speech enhancement based on fast affine projection algorithm using matching pursuit in adaptive noise cancellation is introduced in [10].

II. NOISE CANCELLATION APPROACH

In the proposed system, two techniques are applied, the first one being wavelet filters which do not require any reference signal and the second being adaptive filters that use reference signal. They are applied in this order or vice versa so that the proposed system can suppress the remaining noises in the first filter stage output by the second stage filter.

The decomposition of an input signal into sub-bands by wavelet transform forms a multiresolution analysis. The larger magnitude sub-band coefficients are connected with prominent features in the signal. Hence, denoising can be achieved by applying a thresholding operator to the sub-band coefficients followed by inverse wavelet transform. The proposed system uses well-known Stein`s unbiased risk estimator for threshold estimation which is defined in Eqn. 1.

$$SURE = -n\sigma^2 + \|\hat{s} - s\|^2 + 2\sigma^2 \sum_{i=1}^{n} \frac{\partial \hat{s}}{\partial s_i} \qquad (1)$$

where σ is the standard deviation of the noise, s is the clean signal and $\hat{s}$ is the de-noised signal.

The LMS adaptive filtering is the more appropriate and stable algorithm. It can eliminate noisy signal due to minute step size. The proposed system uses fast transversal LMS algorithms to create adaptive filters. The flowchart of fast transversal LMS adaptive filter is shown in Figure 1.

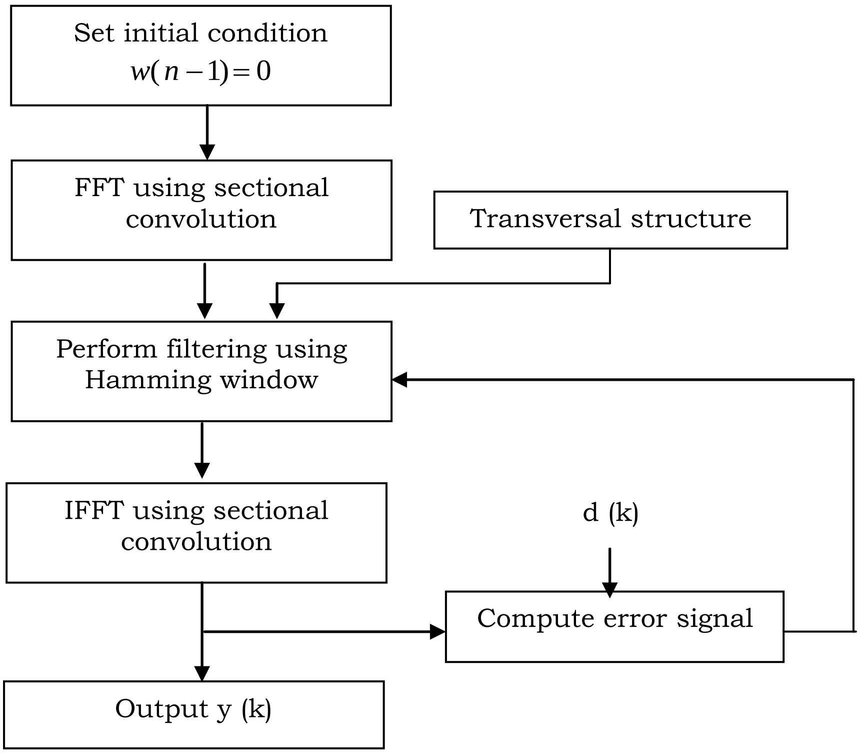


Fig. 1 Flowchart of fast transversal LMS adaptive filter

The hamming window is used to process the output from the FFT block. The transversal structure is adopted by the FIR filter block for processing the complex multi rate input signal. The filtered output is then Inverse Fast Fourier transformed using sectional convolution. The comparison of the sampled output signal with the desired signal produces the error signal.

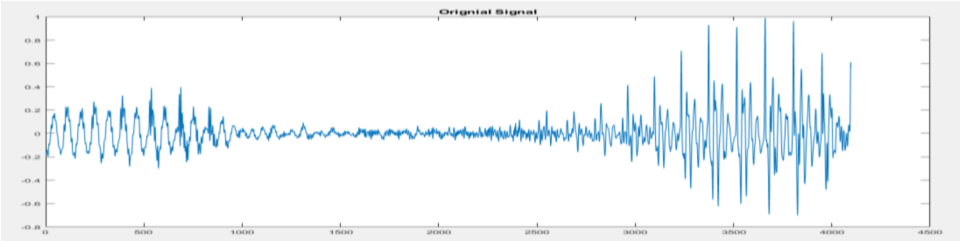
III. RESULTS AND DISCUSSION

The performance is quantitatively evaluated by using RMSE in Eqn. 2, and PSNR in Eqn. 3.

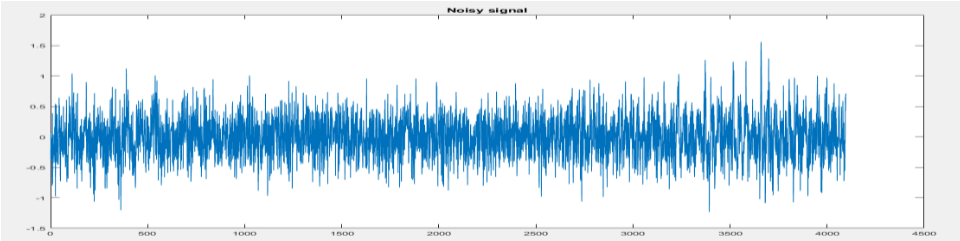
$$e_{rms} = \sqrt{\frac{1}{M}\sum_{m=0}^{X-1}\sum_{n=0}^{Y-1}\left[\check{f}(m.n) - f(m,n)\right]^2} = \sigma_e \qquad (2)$$

$$PNSR = 10 \log_{10}\frac{(peak\ value\ of\ the\ referenced\ signal)^2}{{e_{rms}}^2} \qquad (3)$$

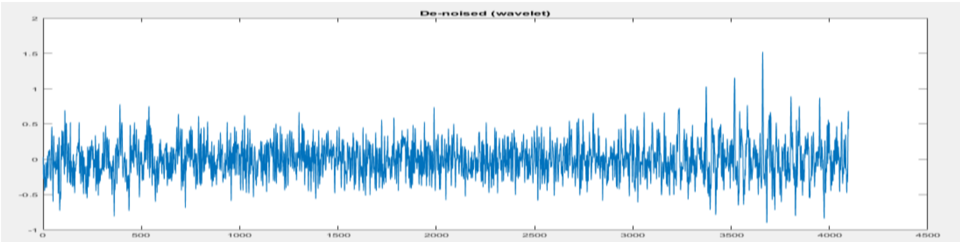
At first, noisy signal is added to the original speech signal. Then, filtering is applied to eliminate the noises using the proposed cascaded system, and the performance measures are calculated. Figure 2 (a) shows the original signal. The noisy version of the signal in Figure 2 (a) is shown in Figure 2 (b). Figure 2 (c) shows the de-noised version by wavelets. The application of fast transversal LMS adaptive filter is shown in Figure 2 (d). The de-noised signal by the cascade approach is shown in Figure 2 (e).



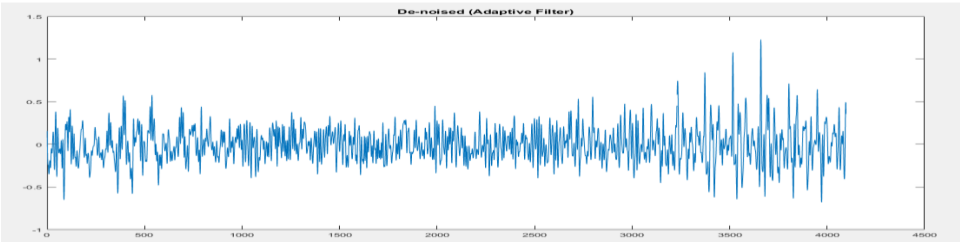
(a) Original Signal



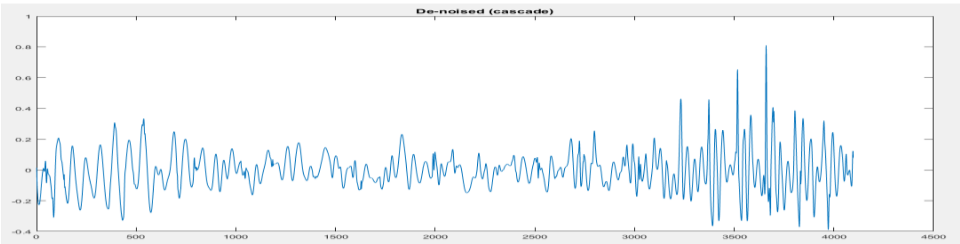
(b) Noisy signal



(c) Noise removal by DWT



(d) Noise removal by fast transversal LMS adaptive filter



(e) Noise removal by cascade approach

Fig. 2 Output waveforms at each step of cascade approach

From the obtained de-noised signal (f) ̆(x,y), the metrics mentioned above are computed with the help of input signal f(x,y) and shown in Figure 3. It is observed that the cascade approach provides better PSNR than wavelet and an adaptive filter.

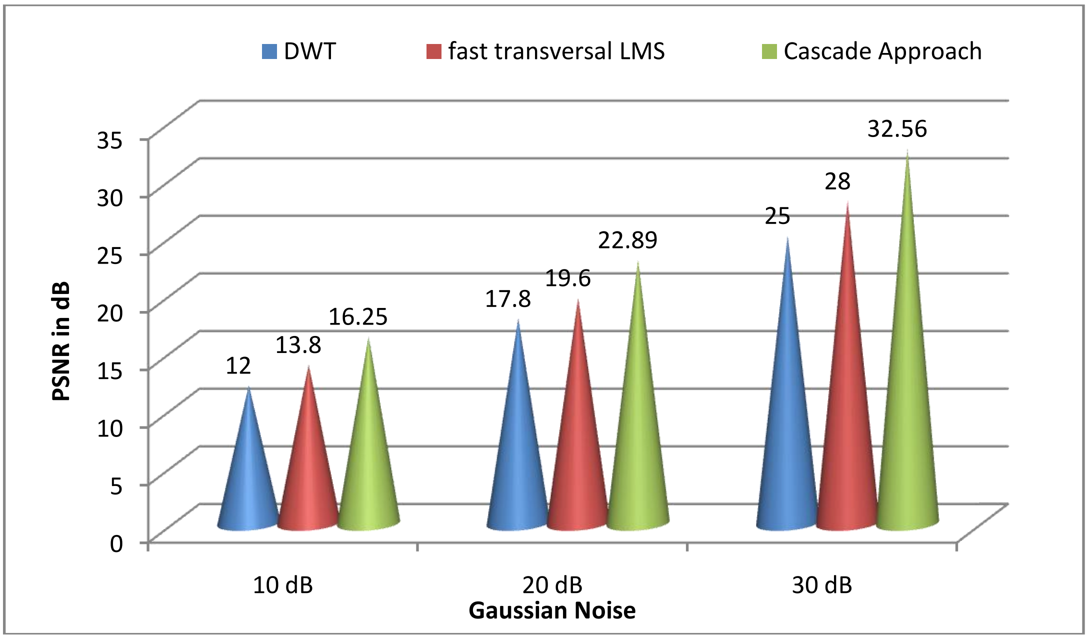


Fig. 3 Performance of noise cancellation by cascade approach using wavelet and adaptive filter

IV. CONCLUSION

In this paper, a cascade approach using wavelet and LMS adaptive filter is presented for noise cancellation. Wavelet transform is widely accepted as an important analysis tool in signal processing. The cascade approach uses fast transversal LMS algorithms to create adaptive filters. In the first stage, denoising is achieved by thresholding in the wavelet domain followed by adaptive filtering. Results show that the cascade approach wavelet and LMS adaptive filter gives promising results in terms of MSE and PSNR.

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