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Signal Adaptive Orthogonal Wavelet Decomposition and Hampel filtering for Impulse De-noising of Speech

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ABSTRACT*:* Impulsive noise is a significant reason for degradation of speech. Intelligibility of speech and performance of automatic speech recognition is minimized.Earlier de-noising performed using various techniques has limitations of detection level and removal of impulse noise. We proposed Signal Adaptive Wavelets for decomposition and used Hampel filtering for de-noising. The proposed methods' performance, efficiency is compared using the metric SNR, significant improvement is obtained.

KEYWORDSImpulse Noise, Adaptive Wavelets, Hampel filter, Intelligibility.

I. INTRODUCTION

Speech being an important medium of communication, degradation of speech through various sources leads to loss of information. Impulse noise is one of such sources that degrades the speech signal. It is a short burst of acoustic energy possessing wide bandwidth. Typical examples of impulse noise are pen clicks, clicks on tape recordings, cracks on gramophone records and popping popcorn. To alleviate the degradation levels in original speech signal, de-noising is necessary. Median filter is the classic method for removal of impulse noise, but it reduces the quality of the speech. Spectral subtraction, a speech enhancement technique that works by subtraction of the estimated noise spectrum from the spectrum of the noisy signal does not de-noise efficiently as impulse noise has very wide bandwidth. De-noising through thresholding and shrinkage of wavelet coefficients is found to be useful[1,2] when compared to the former methods but has limitations. As the number of levels of decomposition increases frequency components of speech is lost after shrinkage of coefficients.In existing method only detection of Impulse noise in speech signal is done. For removal of noise existing methods used linear prediction models which are complex in implementation due to consideration various elements like designing of inverse filters, matched filters and threshold detector.To overcome the said limitations we proposed a new scheme of adaptive wavelets to counter the cited issues.

For designing an impulse noise removal system the following criteria should be considered: 1.Noise and signal characteristics in both frequency and time domains.2.Statistical nature of signal and noise.3.Signal and Noise generation models[3].

For that purpose, the proposed scheme works by decomposing the signal using adaptive wavelets and applying Hampel filter on approximate coefficients and masking the detail coefficients. Hampel filter is a mutant of median filter used for signal smoothening. The efficiency of the proposed scheme is evaluated using SNR(Signal to Noise ratio)and compared with existing methods.

Design procedure for adaptive wavelets is discussed in section 2 of this paper. In section 3 Hampel filter is described. Impulse noise model and simulation results are summarized in sections 4 and 5 respectively. Conclusions are drawn in section 6.

II.DESIGN OF ADAPTIVE WAVELETS

The high pass and low pass filters of a two channel perfect reconstruction (PR) filter bank corresponds to the scaling and wavelet functions respectively. In order to obtain the wavelets a two channel PR filter bank is designed.

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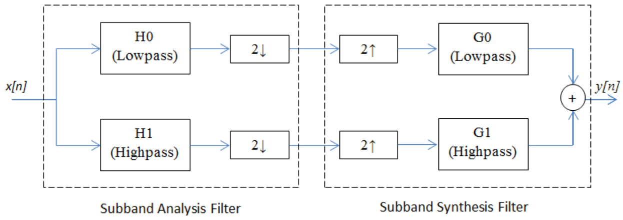

Fig.1 Two channel PR filter bank

The high pass filter H1 is designed such that it maximizes the power of impulse noise relative to that of the speech signal R_i

$$R_i = \sigma_i^2/\sigma_s^2$$

where

$$\sigma_i^2 = \int_{-\pi}^{\pi} \left|H_1(e^{jw})\right|^2 P_i(w)dw \approx \sum_i \left|H_1(e^{jw_i})\right|^2 P_i(w_i)$$

$$\sigma_s^2 = \int_{-\pi}^{\pi} |H_1(e^{jw})|^2 P_s(w)dw \approx \sum_i |H_1(e^{jw_i})|^2 P_s(w_i)$$

$P_i(\omega_n), P_s(\omega_n)$ are power spectrums of estimated noise and underlying speech respectively.

For a given length L the low pass filter is

$$H_0(z) = h(0) + h(1)z^{-1} + \ldots + h(L-1)z^{-(L-1)}$$

High pass filter in terms of low pass filter is

$$H_1(z) = -z^{-(L-1)}H_0(-z^{-1})$$

Orthogonal two channel PR filter bank should satisfy the following conditions.[4]

1.R(z)+R(-z)=2 where R(z) =H_0 (z)*H_0 (z^{-1}).R(z) is the half band filter. Spectral factorization of the R(z) gives $H_0(z)$.

2.r(n) is the autocorrelation sequence of h_0(n)

$$r(e^{jw}) = |H_0(e^{jw})|^2$$

3.The filter coefficients should satisfy double shift orthogonality condition.

$$\sum_n h(n)h(n-2k) = \delta(k), for\ k = 0,1,.,L/2-1$$

where$\delta(k)$ is delta function.

4.The following condition also must be true for wavelet function to exist.

$$\sum_n h(n) = \sqrt{2}$$

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For signal adaptive wavelets convex optimization problem is framed on the autocorrelation sequence[5] of the filter coefficients that maximize the noise relative to the signal.

$$r_h(l) = \begin{cases} \sum_{n=0}^{L-l-1} h(n)h(n+l), & l \geq 0 \\ r_h(-l) \quad , & l < 0 \end{cases}$$

III. HAMPEL FILTER

For each sample of the signal Hampel filter[8] computes the median of the window in which the sample is present. Using median absolute deviation standard deviation of the window about the median is calculated. Each sample is replaced by median if

$$|X_s - median| > \sigma$$

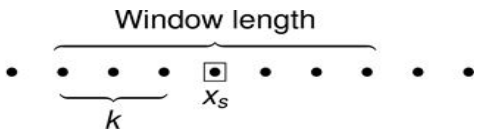

Fig.2. Hampel filter

IV.IMPULSE NOISE MODEL

A binary state model [9] is used to generate the impulse noise.

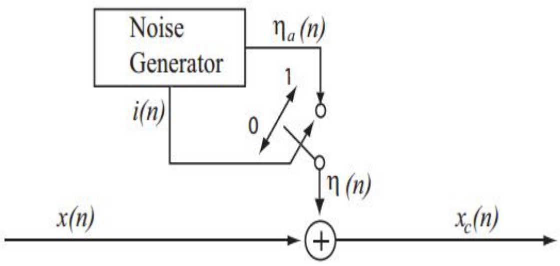

Fig.3. Noise generation model

The process $\eta_a(n)$ generates guassian noise while the binary switch is controlled by the process $i(n)$.The binary switch is operated using Markov model.

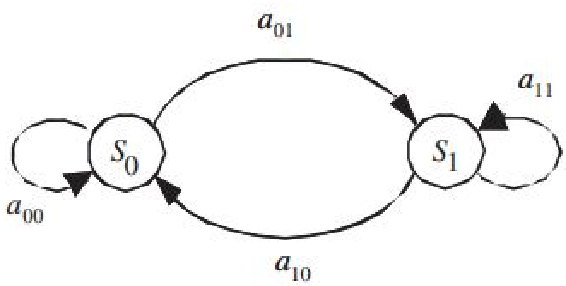

Fig.4.State diagram with transition probabilities

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In this model S_0, S_1 are two states each corresponding to impulse off & impulse on in the noise generation process.a_{00},a_{01},a_{10},$a_{11}$are state transition probabilities. The shape of the impulse noise depends on the state transition matrix

$$T = \begin{bmatrix} a_{00} & a_{01} \\ a_{10} & a_{11} \end{bmatrix}.$$

V. SIMULATION RESULTS

For testing the performance, a clean speech signal of '.wav' file is loaded from the dataset and impulse noise is added at two different SNR levels, 0dB and 5dB.

The figureof clean signal and noisy signal(0dB)is as depicted in figure 5.

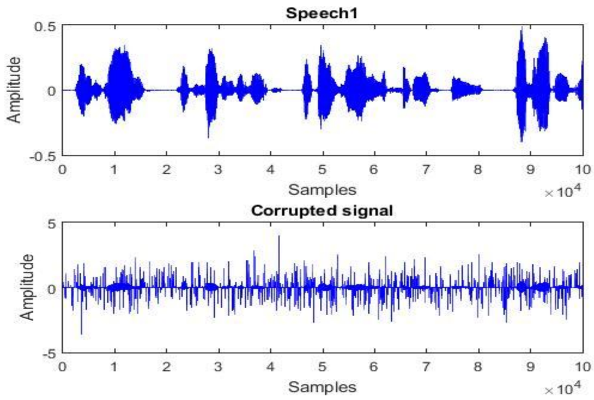

Fig.5.Clean speech and Noisy speech

For the noisy signal using the proposed method signal adaptive wavelets are calculated.After decomposition(One level) the noisy signal with adaptive wavelet with support size2 the approximations and details are as depicted in the figure 6.

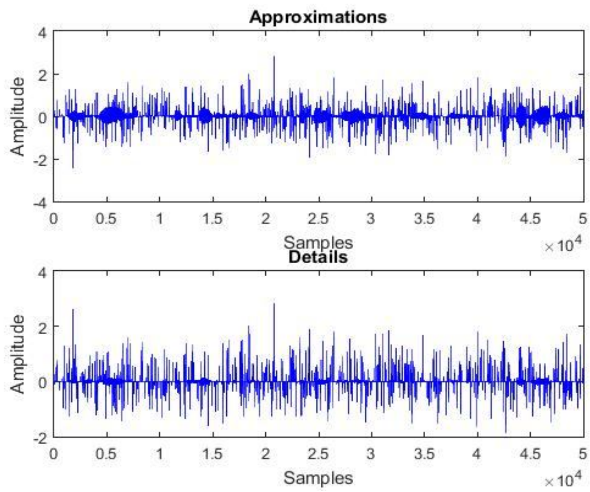

Fig.6.Approximation and Detail Coefficients

After decomposition detail coefficients are discarded and approximations are passed through the Hampel filter and the recoveredde-noised speech signal is shown in the figure 7.

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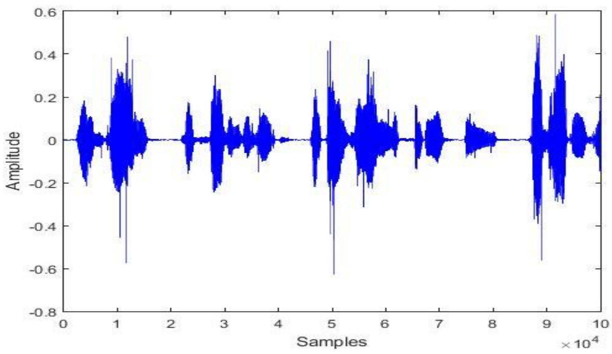

Fig.7.Denoised speech.

In Table I comparison is made between performance of the proposed method and well known wavelets.

Table I

		Spectral subtraction	Duabechies (3 level Decomposition)				Symlets (3 level Decomposition)			Coiflets (3 level Decomposition)			Proposed Method (1 level Decomposition and Hampel filtering)		
			db1	db2	db3	db4	sym2	sym3	sym4	coif1	coif2	coif3	p1	p2	p3
Clean1		SNR													
	0dB	2.1358	3.8134	4.6125	4.8244	4.9025	4.633	4.7938	4.884	4.6225	4.9003	4.95	7.709	6.105	6.597
	5dB	6.3251	4.8069	5.8221	6.134	6.2251	5.802	6.1538	6.257	5.8543	6.2494	6.3	7.779	6.303	7.222
Clean 2															
	0dB	3.7452	3.2768	3.8477	3.9834	4.0452	3.848	3.9834	4.024	3.7473	4.1081	4.02	6.72	5.45	6.208
	5dB	5.7091	4.1464	4.7929	5.0321	5.1155	4.793	5.0321	5.076	4.7306	5.2232	5.06	6.732	5.603	6.65
Clean 3															
	0dB	3.1474	2.9598	3.5334	3.6169	3.6836	3.533	3.6169	3.663	3.5787	3.6385	3.74	6.19	4.853	5.486
	5dB	5.9418	3.7675	4.4618	4.548	4.6466	4.462	4.548	4.619	4.4856	4.608	4.69	6.246	5.028	5.946

VI. CONCLUSION

The SNR of the de-noised signal for different levels of noise are compared with that of spectral subtraction, wavelet shrinkage at 3 levels of decomposition and soft thresholding using Symlets(sym1,sym2,sym3),Duabechies wavelets (db1,db2,db3,db4) and Coiflets(coif1,coif2,coif3) for support sizes of 2,4,6.

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