

Enhancement of Speech Signal Corrupted by Impulsive Noise Using Rank Order Mean

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Abstract: Speech is the medium through which information can be conveyed from one person to another. Audible frequency range of human beings is between 20Hz to 20 kHz, significant frequency components has range up to 4 kHz. Rank Order Mean method has been proposed in this work which is used for ejection of impulse noise from the speech signal using Rank. To estimate the time occurrence of impulsive noise rank order differentiation is applied at the input signal. Then rank order mean is used for replacing the noisy samples to get the noise free signal. The above mentioned technique shows betterment in results in terms of Signal to Noise Ratio (SNR) and Log Spectral Distance (LSD) when compare to the discrete wavelet transform and adaptive median filtering techniques. Simulation work have been done on the Matlab software.

Keywords: Speech enhancement, impulse noise, mean filter, signal to noise ratio(SNR)

1. Introduction

When the information is transmitted from source to receiver, noise from the surroundings gets added to the signal. The resultant signal content two components, one carries the information of interest means the useful signal, the other error generated is because of superimposition with the original signal. Speech is the most important form of human communication. For that reason, there is a big trend to increase the quality of speech signal and improve tele communications limitations. The random error and noise are unwanted because they degrade the accuracy and precision of the measured signal. There are different noises which may affect the signal under consideration, these noises are White noise, Colored noise, Brown noise, Pink noise and Impulse noise. White noise is a sound or signal consisting of all audible frequencies with equal intensity. Any noise that is not white can be termed as colored noise. Colored noise has frequency spectrum that is limited within a range unlike white noise which extends over the entire spectrum. There are different types of colored noise (brown noise, pink noise, etc.) depending upon the gradation in the Power Spectral Density (PSD) of the noise. Colored noise can be generated by passing white noise through a filter with required frequency response.

Impulsive noise refers to sudden bursts of noise with relatively high amplitude. This type of noise causes click sounds in the signal of interest. Impulsive noise is generally modelled as contaminated Gaussian noise [4]. The ratio of the variances of the two Gaussian noises decides the impulsive Character of the noise generated. Impulsive noise corrupt thousands of sample. There are number of methods to overcome this problem. Some of them are methods based on: wavelets [6], soft decision and recursion, diffusion filtering [11] Bayesian frameworks [5]. But above mentioned techniques are not so much efficient. The proposed technique works in following steps:

- First detecting the noisy samples using the technique rank order difference by setting a threshold.
- In this step detected noisy samples will be replaced by clean samples using the technique rank order mean.

2. Detection and Cleaning Algorithm

To accomplish the task of detection and cleaning following steps are followed.

- Taking the sliding window of size five as shown in the figure.

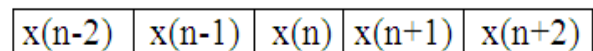


Figure 1. Filter Window

- Consider a vector $x(n)$ of size five centered at n .
- Define a vector of size four that excludes $x(n)$, the centre sample under inspection.

$$w(n) = [w_1(n), w_2(n), w_3(n), w_4(n)]$$

$$= [x(n-2), x(n-1), x(n+1), x(n+2)] \quad (1)$$

- The observation samples from $w(n)$ are sorted,
- $$r(n) = [r_1(n), r_2(n), r_3(n), r_4(n)] \quad (2)$$

where the elements in $r(n)$ are ordered by rank i.e

$$r_1(n) \leq r_2(n) \leq r_3(n) \leq r_4(n)$$

- rank order difference defined as

$$d_i(n) = \begin{cases} r_i(n) - x(n), & \text{if } x(n) \leq \mu(n) \\ x(n) - r_{4-i}(n), & \text{if } x(n) > \mu(n) \end{cases} \quad (3)$$

Where

$$\mu(n) = [r_2(n) + r_3(n)]/2$$

is called the rank order mean (ROM). For a window of size five, $i=1,2$.

- The algorithm decides $x(n)$ is a noise if any of the following conditions hold

$$d_i > T_i, \text{ for } i = 1, 2$$

- Where T_1 and T_2 are two appropriately chosen threshold.
- Every detected impulse is replaced by the ROM, $\mu(n)$.

3. Methodology

- Signal is segmented using the window of size 5.
- Then Rank Order Differentiation is applied.
- To determine the impulse noise location in the signal rank order differentiation is applied.
- The noisy samples are replaced by rank order mean.

4. Implementation Results

A. Speech data base

In the test data set, the voice was recorded with a Compaq iPAQ 3630 built-in microphone and an Optimus Nova 80 close-talk microphone. The iPAQ data were recorded using the AD converter in the iPAQ, and the close-talk data were recorded using a Creative Sound board . Both channels were recorded with a 11.025kHz sampling.

B. Impulse noise modeling

The noise is modeled in 2 stages. In the first stage of noise modeling process, 2-state Markovian state transition is used[12]. One state consider as impulsive and the other state consider as non impulsive state. In the Figure 1 state B is considered as impulsive state i.e this stage is contaminated with noise. State A is considered as non impulsive state i.e noise process is considered to be switched off in this state.

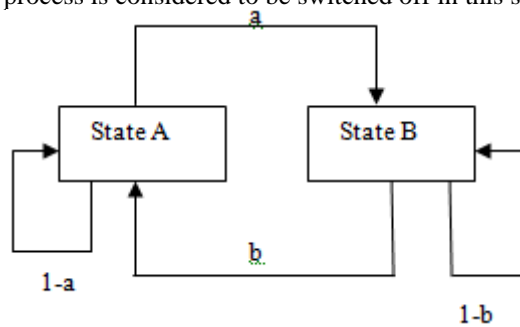


Figure 1: 2 State Markovian process[12]

Second stage of modeling completes with the multiplication of Gaussian distribution having zero mean with the data stream generated in the above stage. the result will be the very short occurring Gaussian distributed noise. Following waveforms are given using Matlab to accomplish the task.

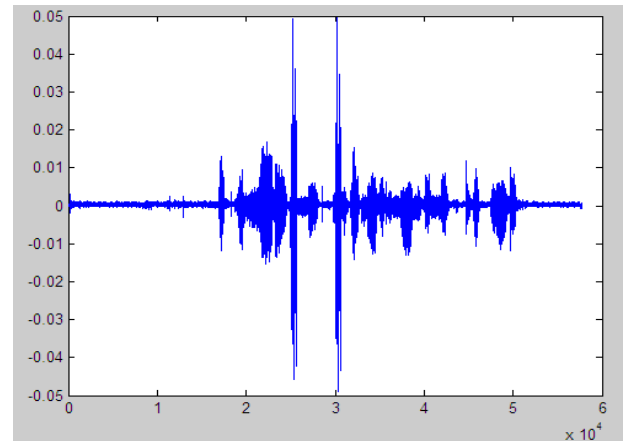
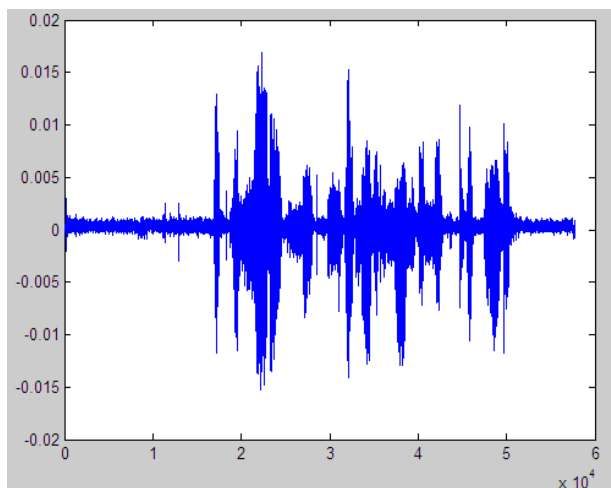
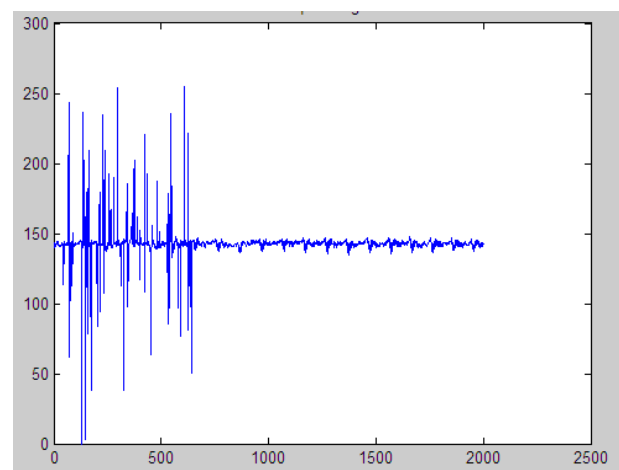
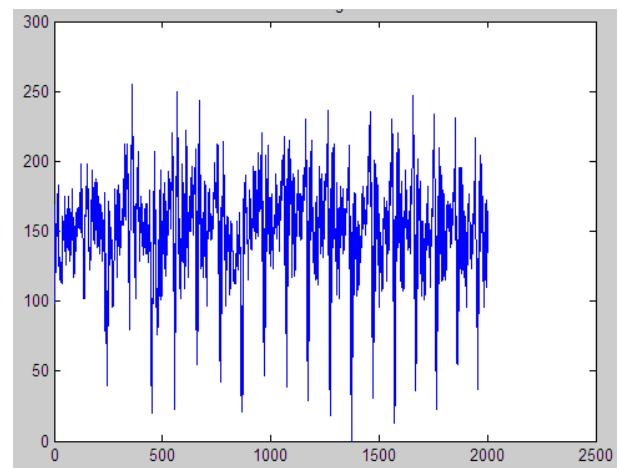


Figure 2: The first plot shows a given number of samples of a speech signal which is uncorrupted by the impulsive noise. The second plot shows the speech signal of given number of samples corrupted by the impulsive noise at a certain range of samples with the higher magnitude compare to the uncorrupted signal.

5. Signal to noise ratio and Log Spectral Distance

The above described techniq shows improvement in terms of Signal to Noise Ratio (SNR) and Log spectral distance(LSD). All the Simulations are done using matlab.



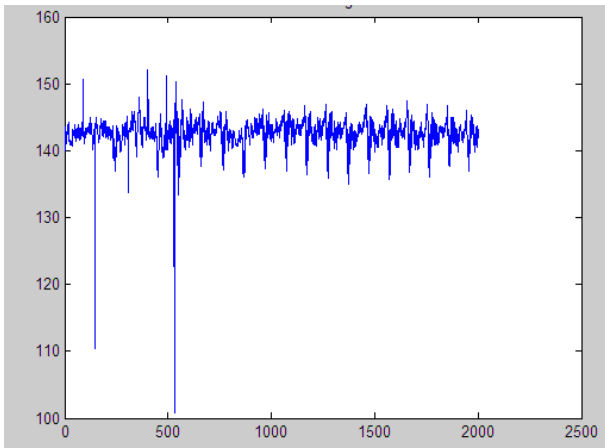


Figure 3: The first shows a pure speech signal at a sampling frequency of 11.025 kHz. The second plot shows the speech signal corrupted by the impulsive noise. The third plot shows the output of the proposed technique. It can be seen that impulsive noise removed effectively.

Table 1: The improvement in the SNR and LSD for proposed technique when compare to the discrete wavelet transform and Adaptive Median filtering technique

Performance Parameter	Proposed technique	Adaptive Median filtering
Signal to noise ratio(SNR)	11.4921	-7.3164
Log spectral distance (LSD)	2.3397	1.5996

6. Conclusion

Speech enhancement is a special case of signal estimation as speech is non stationary and hence human ear is the final judge and it requires a mathematical error criterion. To evaluate their speech enhancement capabilities, all the parameters are performed on computer simulations. The results show that Rank Order Mean method gave the better noise reduction capability in comparison to given speech enhancement method. Future work might involve a real time implementation of the system so that it could be used in devices where speech quality and speech intelligibility is must.

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