Impulsive Noise Removal from Speech Signals using Rank Order Difference

Praveer Chanchal
Global Institute of Technology & Management, Gurgaon

Prof. (Dr) Satya Prakash
Global Institute of Technology & Management, Gurgaon

Abstract

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 Perceptual Evaluation of Speech Quality (PESQ) w. r. t to the existing techniques.
Keywords: Speech Enhancement, Digital Signal Processing, Rank Order Mean

I. 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. The random error and noise are unwanted because they degrade the accuracy and precision of the measured signal. Therefore the effective removal or reduction of noise in the field of signal is an active area of research. The utilization of Adaptive filter is to eliminate the signal corruption caused by predictable and unpredictable noise. An Adaptive filter has the property of self-modification of frequency w.r.t variation in time. Filter alter itself accordingly the variation in input signal characteristics .Due to this capability and the construction flexibility, the adaptive filter have been employed in lot of different applications like radar signal processing, navigation system, channel equalization, communication, biometric signal processing etc. First Stochastic Gradient based algorithm and Second Recursive Least square based algorithm helps in forcing the filter to adapt its coefficients. Their implementation and adaptation properties are the determining factors for choice of application. The main requirement and the performance parameter for adaptive filter are the convergence speed and the asymptotic error. Property of an adaptive filter which enables one to measure how quickly the filter is converging to the desired value is convergence speed. Convergence speed is major need and it act as a limiting factor for most of the application of adaptive filters. The quantity of error introduced by filter at steady state is represented by asymptotic error when it has converged to the desired value. The function of algorithm is to perform the adaptation and the configuration of the filter depends directly on the application of the filter. And the basic computational structure that performs the adaptation of the filter coefficients can be similar for different algorithms which depend on the statistics of the input signals to the system. Generally, adaptive filters are considered as nonlinear systems; therefore their behaviour analysis is more complicated than for fixed filters. As the adaptive filters are self-designing filters from the practitioner's point of view therefore their designing can be considered less involved than the digital filters with fixed coefficients. The most important form of human communication is speech. Though communication languages were first conceived over the ages and different ways of communication techniques have been developed and implemented to convey appropriate information across a distance. In the recent years telephone has revolutionized this process and therefore the demand for efficient communication and data storage is continuously increasing.

In recent communication technology developments like speech, image and video signals play a vital role in the Cellular and Mobile Communication. In digital communication world, Signal representation is one of the important parameter. Demand for cellular, Mobile and convenient forms of communication have been an explosion in the use of Cellular and Satellite Telephony, both of which have significant capacity constraints. Speech coding research functions to address the problem of accommodating more users over such limited capacity by coding speech before transmitting it across a network.

II. RANK ORDER MEAN

A computationally efficient algorithm is proposed to remove noise impulses from speech and audio signals while retaining its features and tonal quality. The proposed method is based on the SD-ROM (Signal Dependent Rank Order Mean) algorithm. This technique has successfully been used to remove impulse noise from images. It has the advantage of being relatively fast, simple and robust. The algorithm estimates the likelihood the sample under inspection is corrupt relative to the neighboring samples and replaces a sample detected as corrupted by a value based on the neighboring samples. This algorithm also has the advantage of being 'configurable' to the type of noise impulses in the sample, as the thresholds used to detect noise impulses can be varied to suit the signal. Standard global filtering techniques like lowpass filtering do not differentiate between impulses corrupted samples and uncorrupted samples.

Median filters and other order static filters that operate on a localized area typically modify uncorrupted samples as the filtering is applied uniformly over the whole signal. The median filter is a highly robust estimator of the signal value in the presence of impulse noise. However, a median filter will eliminate changes in the input signal with duration less than half the size of the filter window. When the signal is heavily corrupted with impulse noise, a large median window **is** needed and this leads to more uncorrupted samples being replaced by the median value within the window and more high frequency components being removed by the filter. As a result, conventional techniques for impulse noise removal perform poorly for an acceptable level of feature and tonal quality.

The ideal objective would be to replace only the noisy samples, leaving uncorrupted samples unchanged. The technique presented in this paper attempts to do this using a detection-estimation approach. Each impulse is first detected in the sample stream and then is replaced by an estimate based on neighboring samples. We assume that the noise impulses can take any arbitrary value within the dynamic range with some error probability. Let *s(n)* denote the uncorrupted l-D sequence and **x(.)** denote its corrupted version containing some impulse corrupted samples. Then, at time instant *n*, for an impulse noise model with an error probability *p e* , we have(1) *x(n)* = { ~ *(n)* ,w ith probability 1 - *p* , ~ *(n)* , with probability *p*, where ~ *(n*is*)* a n identically distributed, independent random process with an arbitrary underlying probability density function.

III. Results and Discussions

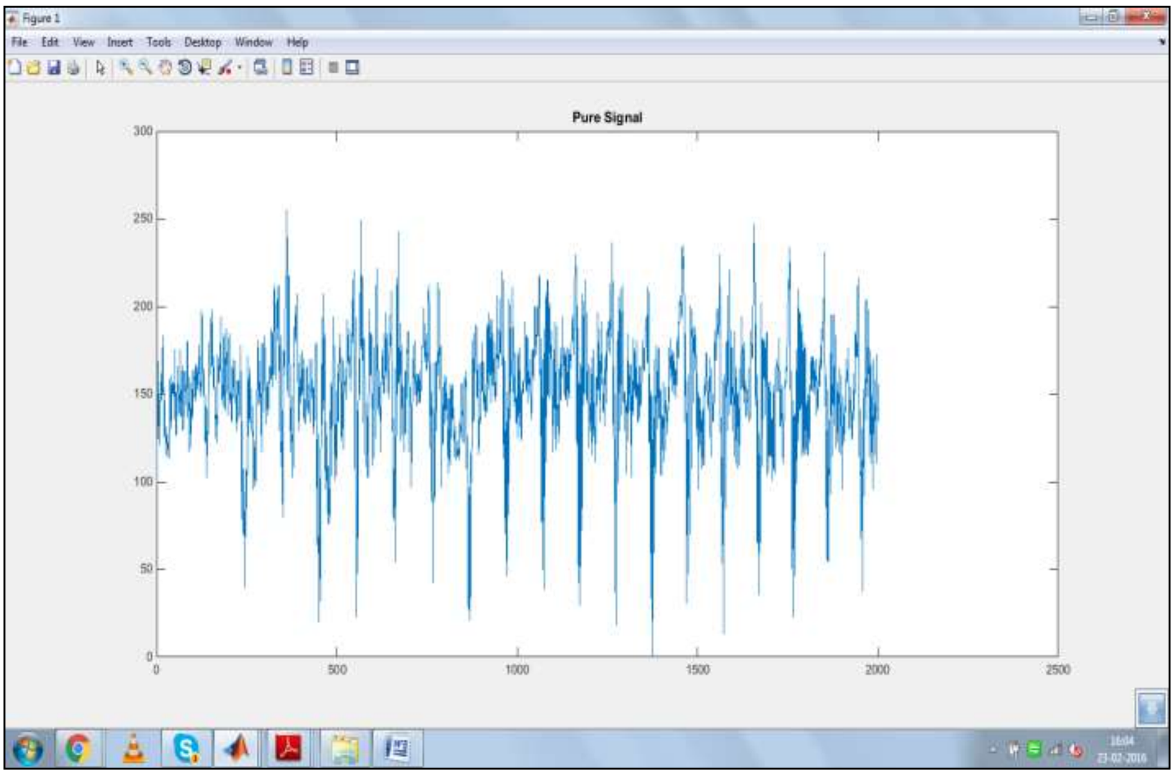

Fig. 1: Pure Audio Signal

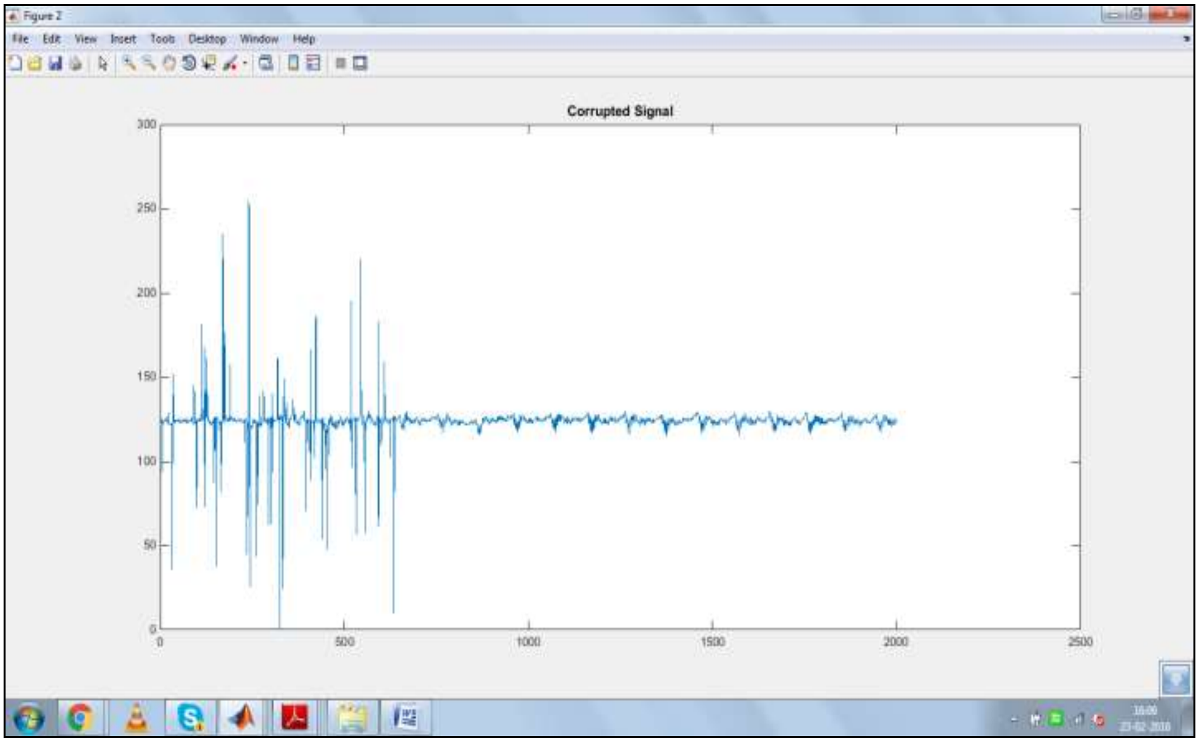

Fig. 2 Corrupted Audio Signal with Impulsive Noise

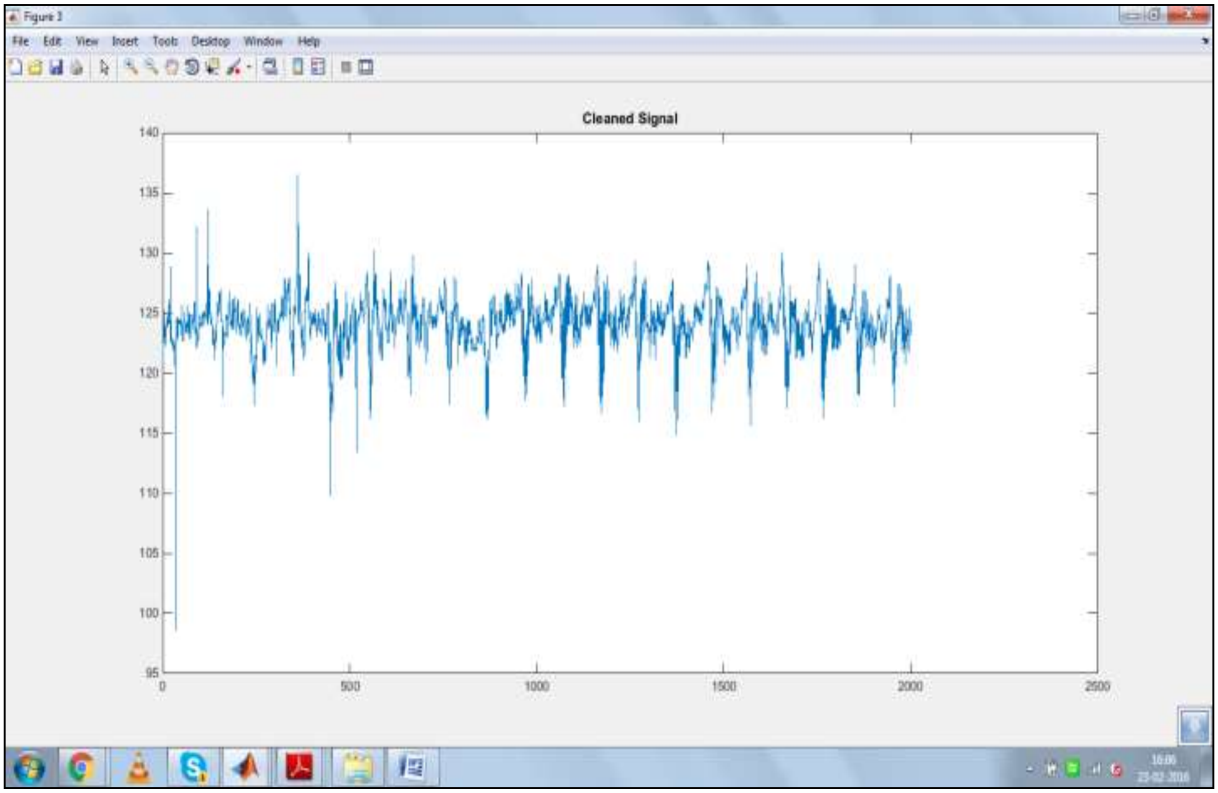

Fig. 3: Cleaned Signal through Rank Order Mean

IV. METHODOLOGY

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

Table – 1
Comparison between results in proposed method and the base paper method

S.No.	*Improvement in SNR*	
	Proposed Technique	*Adaptive Median Filtering*
SNR	*13.4982dB*	*8.93dB*
LSD	*2.3358*	*Not mentioned in the paper*

In this work a method for removal of impulse noise from the speech signal using Rank Order Mean is proposed. The rank order differentiation is applied to input signal to estimate the time occurrence of impulsive noise. Then rank order mean is used for replacing the noisy samples to get the noise free signal. The above described technique shows improvement in terms of Signal to Noise Ratio (SNR) and Perceptual Evaluation of Speech Quality (PESQ) w. r. t to the existing techniques.

V. 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. The rank order differentiation is applied to input signal to estimate the time occurrence of impulsive noise. Then rank order mean is used for replacing the noisy samples to get the noise free signal.

REFERENCES

[1] S. V. Vaseghi, “Advanced digital signal processing and noise reduction.” Wiley, 2008, pp. 29–43.
[2] R. C. Nongpiur, “Impulse noise removal in speech using wavelets,” in Acoustics, Speech and Signal Processing, 2008. ICASSP 2008. IEEE International Conference on. IEEE, 2008, pp. 1593–1596.
[3] A. Subramanya, M. L. Seltzer, and A. Acero, “Automatic removal of typed keystrokes from speech signals,” Signal Processing Letters, IEEE, vol. 14, no. 5, pp. 363–366, 2007.
[4] C. Chandra, M. S. Moore, and S. K. Mitra, “An efficient method for the removal of impulse noise from speech and audio signals,” in Circuits and Systems, 1998. ISCAS’98. Proceedings of the 1998 IEEE International Symposium on, vol. 4. IEEE, 1998, pp. 206–208.
[5] Z. He, X. Guo, and M. Zhang, “Detection and removal of impulsive colored noise for speech enhancement,” in Information and Automation (ICIA), 2010 IEEE International Conference on. IEEE, 2010, pp. 2320–2324.
[6] S. Zahedpour, S. Feizi, A. Amini, M. Ferdosizadeh, and F. Marvasti, “Impulsive noise cancellation based on soft decision and recursion,” Instrumentation and Measurement, IEEE Transactions on, vol. 58, no. 8, pp. 2780–2790, 2009.
[7] R. Talmon, I. Cohen, and S. Gannot, “Speech enhancement in transient noise environment using diffusion filtering,” in Acoustics Speech and Signal Processing (ICASSP), 2010 IEEE International Conference on. IEEE, 2010, pp. 4782–4785.