Impulsive Noise Removal from Speech Signals using Rank Order Mean

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Abstract— In this Paper represent the impulsive noise Removal speech using rank order mean. Speech is a very rudimentary way for persons to transport information to one additional with a bandwidth of only 4 kHz; speech can convey information with the feeling of a humanoid voice. The language indication has convinced properties: It is a one-dimensional signal, with time as its self-governing variable, it is chance in countryside, it is non-stationary, and i.e. the frequency spectrum is not continuous in time. Though humanoid existences have an audible frequency range of 20Hz to 20 kHz, the humanoid speech has important occurrence components only up to 4 kHz. 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. Noise cancellation is the process of removing background sound from language sign. The squalor of speech due to presence of contextual noise and several other noises reason problems in various sign processing errands like speech credit, speaker recognition, and speaker verification etc. Numerous methods have been extensively used to remove noise from speech signal like linear and nonlinear filtering methods, adaptive noise annulment, total difference demising etc. This paper addresses the problem of reducing the impulsive sound in talking signal using compressive sensing method. The results are compared against three well known speech improvement means, spectral deduction, Total difference denoising and signal dependent rank order mean algorithm.

Key words: Signal to Noise Ratio (SNR), Evaluation of speech Quality (PESQ), Frequency range of 20Hz to 20kHz, Bandwidth of only 4kHz etc

I. Introduction

When the information is transmitting from source to receiver, noise from the surroundings gets added to the signal. The resultant signal content two components, one carry the information of interest means the useful signal, the other error generated is because of superimposition with the original signal. The random fault and noise are unwanted because they corrupt the accuracy and precision of the considered signal. Therefore the efficient removal or reduction of noise in the field of signal is an active area of do research.

The consumption of Adaptive filter is to eliminate the signal corruption caused by expected and impulsive noise. An Adaptive filter has the property of self-adaption of frequency w.r.t variation in time. Filter alter itself consequently the variation in input signal characteristics .Due to this capacity and the construction stiffness, the adaptive filter have been employed in lot of different application like radar signal handing out, navigation system, channel equalization, communiqué, biometric signal processing etc. First Stochastic grade base algorithm and Second Recursive slightest Square based algorithm helps in forcing the filter to adjust its coefficients. Their functioning and adaptation properties are the decisive factors for choice of application. The main requirement and the performance parameter for adaptive filter are the union speed and the asymptotic fault. Property of an adaptive filter which enables one to measure how rapidly the filter is converging to the desired value is union speed. Union speed is major need and it act as a off-putting factor for most of the application of adaptive filters.

II. Literature review

Over the past few decades, most of the attention was given to system modeling of speech signals and has been incorporated in several commercial speech coding standards. First scalar quantization is introduced and has come out with some of the disadvantages like space-filling, the shape advantage, and the memory advantage. Most of the researchers have proposed that the quantization technique used should have less computational and memory requirements and it should not result in suboptimal quantization performance of intelligibility. Speech coders operating at low bit rates demand efficient encoding of Linear Predictive Coding (LPC) coefficients. Line Spectral Frequencies (LSF) parameters are currently one of the most efficient choices of transmission parameters for the LPC coefficients [8].

III. Speech analysis filter

Linear Predictive Coding is most efficient form of coding technique and it is used in different speech processing applications for representing the envelope of the short-term power spectrum of speech. In LPC analysis of order ‘’ the linear combination of p past samples helps in prediction of current speech sample s (n).

After implementation of analysis filter, the quantization techniques are implemented and the speech signal is to be brought from the quantized signal at the receiver and so the quantized signal is to be synthesized to get the speech signal.

The synthesis of speech signal can be represented by the following diagram:

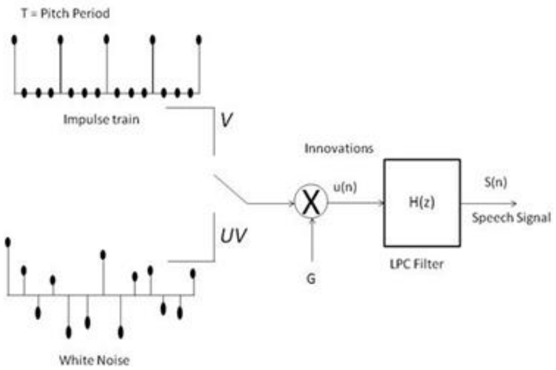

Fig. 1: Diagram For Synthesis Of Speech Signal

To identify the sound whether it is voiced or unvoiced, the LPC analysis of each frame can act as decision-making process. The impulse train is used to represent voiced signal, with nonzero taps occurring for every pitch period.

To determine the correct pitch period / frequency, pitch-detecting algorithm is used. The pitch period can be estimated using autocorrelation function. However, if the frame is unvoiced, then white noise is used to represent it and a pitch period of T=0 is transmitted.

Therefore, either white noise or impulse train becomes the excitation of the LPC synthesis filter [6]. Hence it is important to emphasize on pitch, gain and coefficient parameters that will be varying with time and from one frame to another.

The above model is often called the LPC Model. This model speaks about the digital filter (called the LPC filter) whose input is either a train of impulses or a white noise sequence and the output is a digital speech signal. The relationship between the physical and the mathematical model is given below:

– Vocal Tract	H (z) (LPC Filter)
– Air	u (n) (Innovations)
– Vocal Cord Vibration	V (Voiced)
– Vocal Cord Vibration Period	T (Pitch Period)
– Fricatives and Plosives	UV (unvoiced)
– Air volume	G (gain)

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.

V. Algorithm Approach

This section begins with a broad overview, covering the "high-level" operation of one form of this filter. After presenting this high-level view, the specific equations and their use in this discrete version of the filter are focused.

Firstly, it estimates a process by using a form of feedback control loop whereby the filter estimates the process state at some time and then obtains feedback in the form of (noisy) measurements. As such, these equations for this filter fall into two groups: “Time Update equations” and “Measurement Update equations”.

The responsibilities of the time update equations are for projecting forward (in time) the current state and error covariance estimates to obtain the priori estimates for the next time step. The measurement update equations are responsible for the feedback i.e. for incorporating a new measurement into the priori estimate to obtain an improved posteriori estimate.

The time update equations can also be taken as “predictor” equations, whereras the measurement update equations can be thought of as “corrector” equations. By and large, e final estimation algorithm loop process similar as that of a predictor-corrector algorithm to resolve out numerical problems just like the one shown in Figure below.

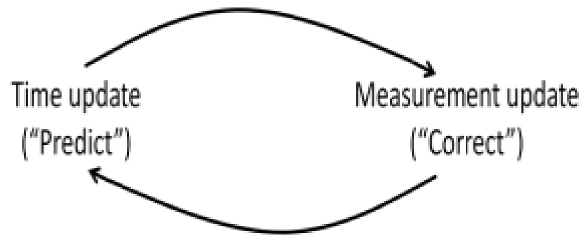

Fig. 2: Process of prediction and correction

As the time update projects the current state estimate ahead in time, the measurement update accordingly adjusts the projected estimate from the time update by an actual measurement at that particular time. The above said procedures are also followed for speech signal corrupted by Real world noise signals and all the parameters are calculated and tabulated.

All the Simulations are done using MATLAB.

VI. Results and Conclusion

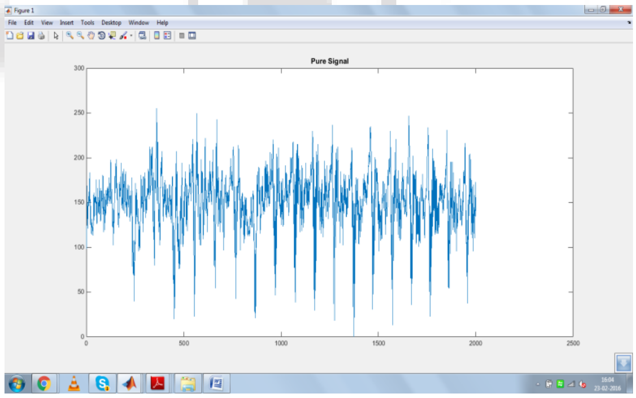

Fig. 3: Pure Audio Signal

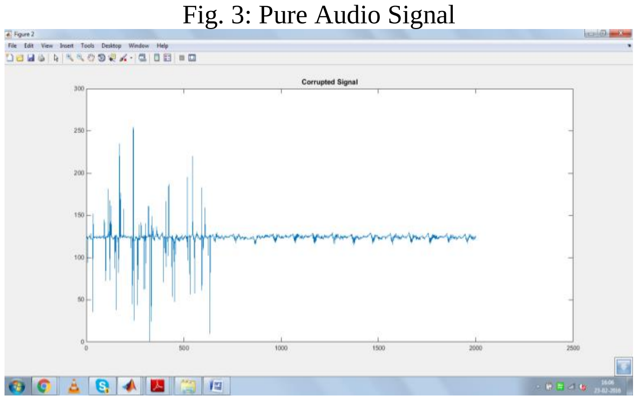

Fig. 4: Corrupted Audio Signal with Impulsive Noise

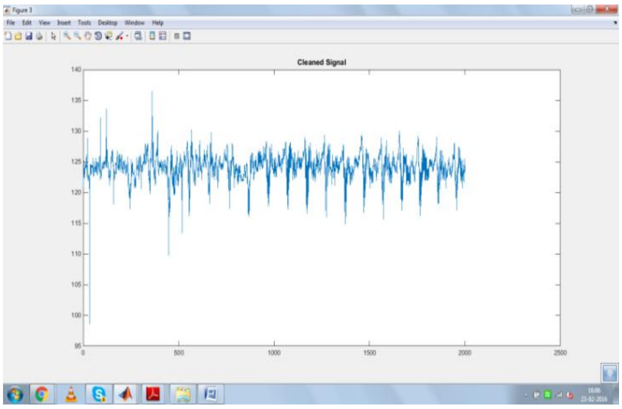

Fig. 5: Cleaned Signal through Rank Order Mean

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 Mentione d in paper

Table 1:

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.

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.

This thesis has presented a historical review about some speech estimation techniques and explicitly states the difference between their theoretical back-ground. Moreover, to evaluate their speech enhancement capabilities, all the parameters are performed on computer simulations. The results show that Recursive filter method gave the best noise reduction capability in comparison to other speech enhancement methods presented in this thesis.

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References

[1] M.Satya Sai Ram, P. Siddaiah and M.Madhavi Latha, “Multi Switched Split vector quantizer,” International Journal of Computer, Information, and Systems Science, and Engineering, Winter 2008, pp. 1-6.

[2] M.Satya Sai Ram, P. Siddaiah and M.Madhavi Latha, “Multi Switched Split vector quantization of narrow band speech signals,” Proceedings of world academy of science, Engineering and Technology, WASET, Vol 27, Feb 2008, pp. 236-239.

[3] M.Satya Sai Ram, P. Siddaiah and M.Madhavi Latha, “Speech Coding & Recognition,” Proceedings of world academy of science, Engineering and Technology, WASET, Vol 39, March 2009, pp. 54-58.

[4] So. Stephen and K. K. Paliwal, “Efficient product code vector quantization using switched split vector quantizer,” Digital Signal Processing journal, Elsevier, Vol. 17, Issue 1, Jan 2007, pp.138-171.

[5] Paliwal, K.K, Atal, B.S. Efficient vector quantization of LPC Parameters at 24 bits / frame. IEEE Trans. Speech Audio Process, pp.3- 14, 1993.

[6] F.K.Soong and B.H.Juang, “Line spectrum pair (LSP) and speech data compression,” in: Proc. IEEE Int. Conf. Acoust., Speech, Signal Processing, Vol 9, Issue 1, March 1984, pp. 37-40.

[7] W.B. Kleijn and K.K.Paliwal, “An introduction to Speech coding,” Speech coding and synthesis, Elsevier science, 1995, pp. 1-47.

[8] P. Kroon and W. B. Kleijn. “Linear Predictive Analysis by Synthesis Coding, in Modern Methods of Speech Processing” Chapter 3. (Kluwer Academic Publishers, 1995)

[9] P. Kroon and E. F. Deprettere, A class of analysis-by-synthesis predictive coders for high quality speech coding at rates between 4.8 and16 kbit/s.," IEEE J. Selected Areas Comm., vol. 6, pp. 353-363, 1988.

[10] K.K. Paliwal, B.S. Atal, Efficient vector quantization of LPC parameters at 24 bits/frame, IEEE Trans. Speech Audio Process. Vol.1, No. 1, pp.3–14, Jan.1993 pp.3–14.