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A Hybrid Model for Acoustic Feedback Cancellation

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Abstract – Acoustic feedback is the problem which occur where a speaker or microphone device reproduce the sound signal. This reproduction can occur because of wall refection and affects the signal quality and frequency. In this work, a layered model is defined to reduce the speech signal noise and acoustic feedback. In first stage of this layer, the high level impurities are removed using spectral subtraction method. Later on LPC and DWT are used collectively to analyze the signal and remove the signal noise. The experimentation is applied on multiple real time speech signals. The results shows that the work provided the effective MSE and PSNR values.

Index Terms – Speech, Acoustic Feedback Cancellation, LPC, Spectral Subtraction, DWT.

1. INTRODUCTION

Speech is the raw form of biometric signal used to provide the communication or information transition by a person. The human to computer interaction is also performed using speech. There are number of voice commanders available to provide the speech adaptive operations. Speech is also having the significance in many of the authentication systems. These authentication systems include the speech content based authentication and the speaker level authentication. Another associated application area for speech processing is the language adaptive translation. It means, the speech contents of one language can be transformed to other language. Such kind of transformation can be performed between different language forms.

But the outcome of these all applications and processes depends on the speech quality. As the speech is collected from raw source it can have number of integrated impurities. These impurities disrupt the speech signal feature so that the overall accuracy of any of speech processing is degraded. There is the requirement to remove these signal impurities and provide the effective speech communication. The speech signal enhancement is considered as the preprocessing activity included with each speech processing application and process. The effectiveness of these applications depends on the accuracy of the speech signal enhancement methods. In this paper, a more robust and adaptive method is defined for speech signal enhancement. This paper is basically focused on the acoustic feedback cancellation over the speech signal.

1.1. Acoustic Feedback

The acoustic path exist between the device and speaker, it actually generally a loop so that the function level application is obtained over the signal. The block diagram of acoustic feedback loop is shown in figure 1.

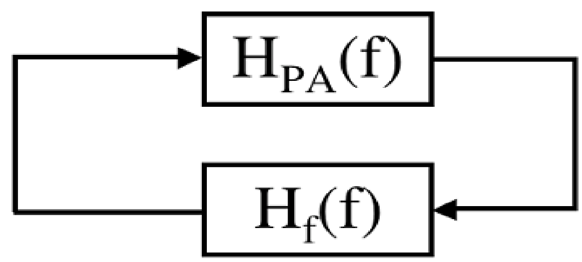

Figure 1: Block Diagram of Acoustic Feedback

The public communication system mostly infected with acoustic feedback which is also called howling. This kind of signal grows the signal unboundedly and gives the element based clipping. The howling is generally loud and disturbed form of noise. It is required to identify the acoustic feedback and remove it. The stability adaptive criteria are required to analyze the adaptive problem and resolve it.

The speech Acoustic Feedback Cancellation is the major requirement for any of the speech processing which is been resolved in this paper by defining a hybrid model. The presented work is defined as a layered approach for speech enhancement. In this section, the speech signal problems and processing applications are explored. The section also explained the concept of acoustic feedback and its effect on speech signal. In section II, the work presented by the earlier researchers is discussed. In section III, the proposed work model is presented and discussed. In section IV, the conclusion obtained from the work is presented.

2. RELATED WORK

In this section, the work defined by earlier researchers is discussed. Most of the researchers defined work for noise reduction and speech signal enhancement. Author [1] has presented a work on speech signal improvement using vector quantization model. Author defined performance adaptive approach to identify the parametric change based on the fractional adaptation. Author defined the analysis under different statistical vector. Author presented the noise adaptive improvement to the signal so that the overall performance will be improved. Author [2] has provided an adaptive model to improve the degraded quality of speech signal. Author analyzed the signal under frequency domain and provided the adjustment to the signal so that the high quality improvements will be achieved. Author provided the improved signal communication for internet application. Author provided the improvement in the quality of signal.

Author [3] has provided a work on speech signal improvement using the adaptive filtration model. Author used the wavelet based decomposition model for frequency based signal separation and reduced the complexity of the signal. Author applied work based on five different signal improvements. Author adopted the model under the bit rate estimation with signal improvement and provided the speech enhancement so that the adaptive signal improvement is achieved. The method discussed in this work includes wavelet transformation, DWT, quantization model, thresholding, encoding etc. The signal reconstruction is here performed to provide the adaptive signal communication. Author [4] has provided a comparative model for filtration modeling and relative improvement in the parameters. The work includes the signal level enhancement so that quality signal adaptation is achieved from the work.

Author [5] has provided a work on speech signal processing for Tamil datasets so that the improved signal transition and improvement will be achieved. Author provided the signal enhancement using DCT, LPC and the DWT approaches. Author provided the high quality improvement over the speech so that the improved signal communication will be achieved from the work. Author obtained the improved signal communication with derived signal value so that the applied signal communication will be formed from the work. Author provided the adaptive signal communication so that the effectiveness of signal will be improved. Author [6] has provided a work on high quality signal processing and the signal encoding using different encoders. Author provided the improvement over the speech with relative delay specification. Author achieved the shape driven analysis over the signal with time and frequency domain and provided the codec integrated improvement so that the improved signal form will be obtained from the work. Author [7] has provided a work on signal compression modeling under the speech enhancement adaptive modeling. Author analyzed the signal under decomposition model and provided the bandwidth adaptive optimization so that the improved signal form is obtained from the work. Author provided the high level signal improvement so that the effective signal improvement can be gained from the work. Author [8] has provided the work coefficient vector adaptive communication of speech signal so that the prediction to the noise constraints can be obtained from the work. The analysis can be here done under specification of different associated noise vectors with specification of threshholding and the band adaptive modeling so that the improved signal communication will be formed from the work. Author provided the signal improvement under the variation analysis so that the change analysis will be performed and the signal will be improved. Author [9] defined a work on sub band division based capacity derivation model for speech signal improvement. Author presented the adaptive analysis model for transition of signal in actual signal form. Author [10] provided a work on the encoding method applied to analyze the error over the signal at early stage and provided the adaptive signal improvement will be done. Author [11] provided a work on signal analysis under spectral method so that the frequency based modification over the signal can be obtained.

3. RESEARCH METHODOLOGY

In this present work a layered model is presented to identify the acoustic feedback problem over the speech signal and provided the adaptive improvement over the speech. The complete work is here divided in three subsequent stages. Where the outcome of one phase is taken as the input to the other. In first phase of this model the frequency signal analysis is obtained. To resolve the signal level problem, the Spectral subtraction method is applied. This method is frame adaptive analysis performs the signal level encoding to identify the high frequency noise over the speech signal. Once the problem is identified, the relative absolute value is subtracted from the speech and the improved signal form is obtained from the work.

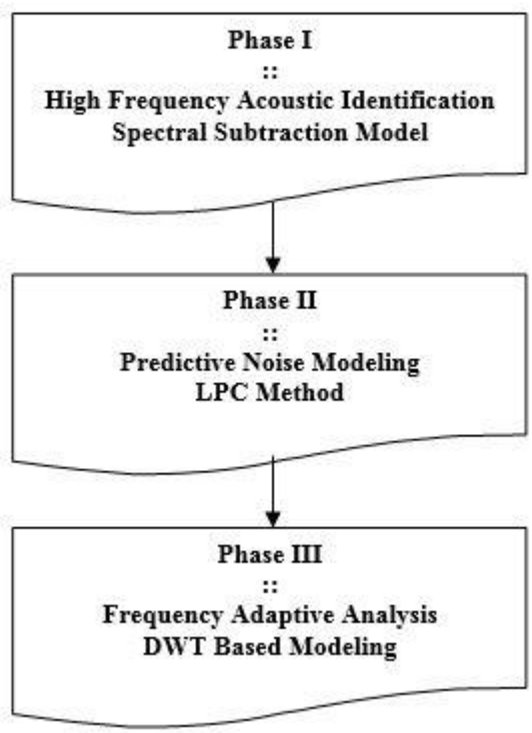

Figure 2: Presented Model

Here figure 2 is showing the complete descriptive model of this work along with associated problem resolution and the method applied. In first phase, where the high level acoustic identification is done using spectral subtraction method. This model is defined under the frame adaptive modeling so that the improved signal form is obtained from the work. Once the improved signal form is obtained, in second stage, the probabilistic analysis over the signal is done. In this stage, the LPC modeling is applied to improve the signal. The LPC is the predictive model applied in the window adaptive form. In final stage of this model, the DWT adaptive modeling is defined for frequency adaptive analysis. In this stage, the frequency adaptive analysis is applied over the signal so that the improved signal form will be obtained from the work. The work model has analyzed the signal adaptively and provided the complete solution in terms of noise reduction.

4. RESULTS

The presented work is implemented in matlab environment and the analysis of the work is performed on different speech signals. These signals are collected from external web source and present in the form of .wav file. One of the sample signal is shown in figure 3.

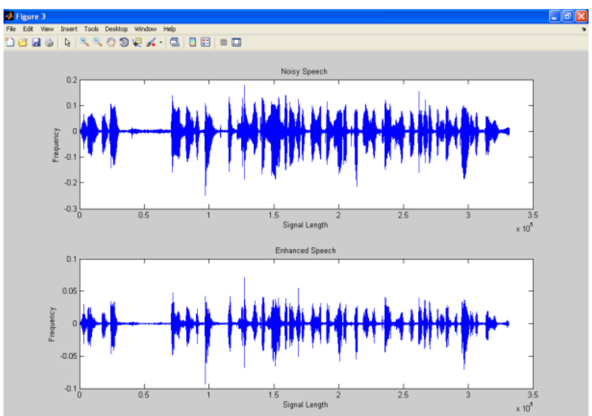

Figure 3 : Speech Signal (Noisy and Improved)

Here figure 3 is showing the noise inclusive speech signal and also showing the first level enhanced signal obtained after spectral subtraction method. The error point identification is shown here in figure 4.

Here figure 4 is showing the identification of problem point over the speech signal. Here x axis is showing the signal length and y axis is showing the signal frequency. Based on this analysis, the signal improvement is done using proposed hybrid model.

The proposed model is applied over the signal for speech signal enhancement. The analysis is here done on 5 different speech signal and the parameter taken for analysis is MSE (Mean Square Error). Lower the MSE value, more effective the signal is considered. The obtained results are discussed here.

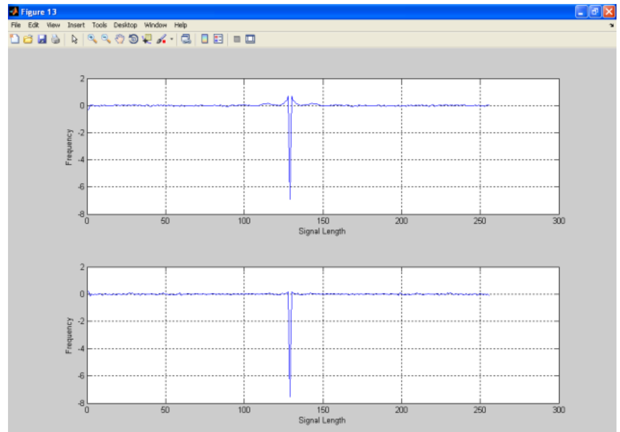

Figure 4: Problem Point Identification

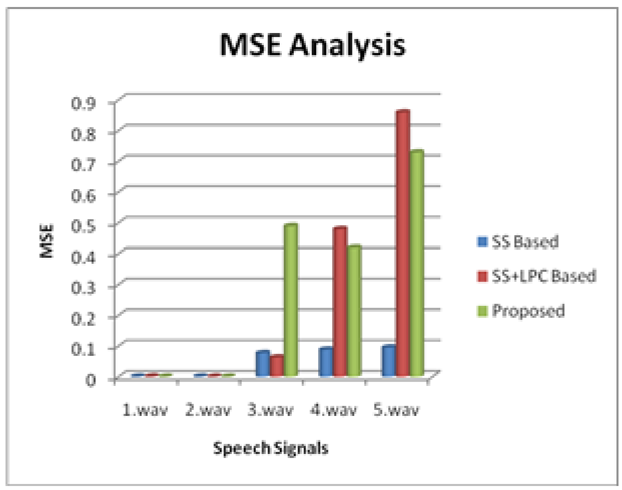

Figure 5 : MSE Analysis (Comparative)

Here figure 5 shows that the maximum MSE value otbtained here is 7 which shows that the work has provided the signifcant results for most of the signals. The analysitcal results of first two signal is very much improved and for other speech signals, the results are significant. The comparative analysis shows that the presnted hybrid model has reduced the error rate.

5. CONCLUSION

In this paper, an improved method is defined for speech signal enhancement in case of acoustic feedback problem. The presented model is applied on real time speech signals. The analysis results show that the work has improved the signal up to an extent.

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