An Algorithm for Speech Enhancement

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***Abstract-* The speech enhancement has become a challenging task in the real word environment. The speech which having the noise is not much clear to understand. In order to make much clear speech to understand an speech enhancement technique is introduced. Which remove the unwanted noise in the speech and make much clear to understand? Some of the algorithm has been introduced in order to make clarity of speech such as MCRA, STSA, Gamma, Laplacian method and DWT methods so on. In this paper, we introduced an algorithm which removes the unwanted sound present in the speech by applying two level speech enhancement algorithms, In order to get very clear Speech.**

Key words: MCRA, STSA, GAMMA, DWT, Winner pass filtering, Band Pass filtering

I. INTRODUCTION

The real challenge for removal of noise from the audio or speech has become the very critical task. Because of noise in the audio it is very difficult to hear audio or speech clearly. The noise in the sense unwanted sound present in the audio track. There are different type of noise can present in the audio such are white noise, gussien noise, guesion white noise and color noise. Here we introduced an algorithm which removes the noise and produce the clarity of audio sound. In this methodology, first the audio frames are segmented and then remove the unwanted sound or noise from the frame. Finally it will combine the segmented frame of non noise frames and also give the much clear audio track. So many different existing techniques are used for removal of unwanted sound form audio or speech track. Some of techniques are Discrete wavelet transformation(DVT), Fourier transformation technique(FTT), Discrete cosine transformation(DCT), spectral coefficient, linear perceptual coefficient(lpc).

II. RELATED WORK

Cohen et al [1] Introduced an approach which make the speech much more enhancement. This author has used an MCRA (Minima controlled recursive algorithm) for noise variance estimation. This approach provides the high performance compare to weighted average method. And also compare the MCRA algorithm with the spectrogram of speech enhancement.

Berdugo B[2] This author has updated the MCRA algorithm for noise estimation by adding the average part spectral value for noise speech. This is used to control by the time and frequency dependent smoothing factors.

R martin et al[3] has introduced an approach on gussian statistical model. This provides a good approach for the noise DFT co efficienc. This approach work on DFT frames size for large for speech correlations of the signal.

Y Ephraim et al[4] This author has worked on short time spectral amplitude for the noise speech signal. This method provides the minimum mean square error of the log spectrum.

Israel cohen et al[5] Here author has introduced an minimum control recursive averaging for estimation of noise using the average part spectrum power value and smoothing parameter.

R Martin et al[6] The author has worked on real and imaginary part of the signal. Which are estimated in MMSE and he combined the gamma and laplacian distribution for noise filtering. In order to provide the speech much clear.

Discrete wavelet transformation: Is the technique which is used to remove the unwanted sound or any disturbance which occur during the recoding sound. But this technique is not much efficient in order to remove the unwanted sound form the audio track. [7].

Linear perceptual coefficient technique: This technique is much efficient then the discrete wavelet transformation technique. This approach remove the unwanted sound present in the audio track and then produce the much clear sound [8].

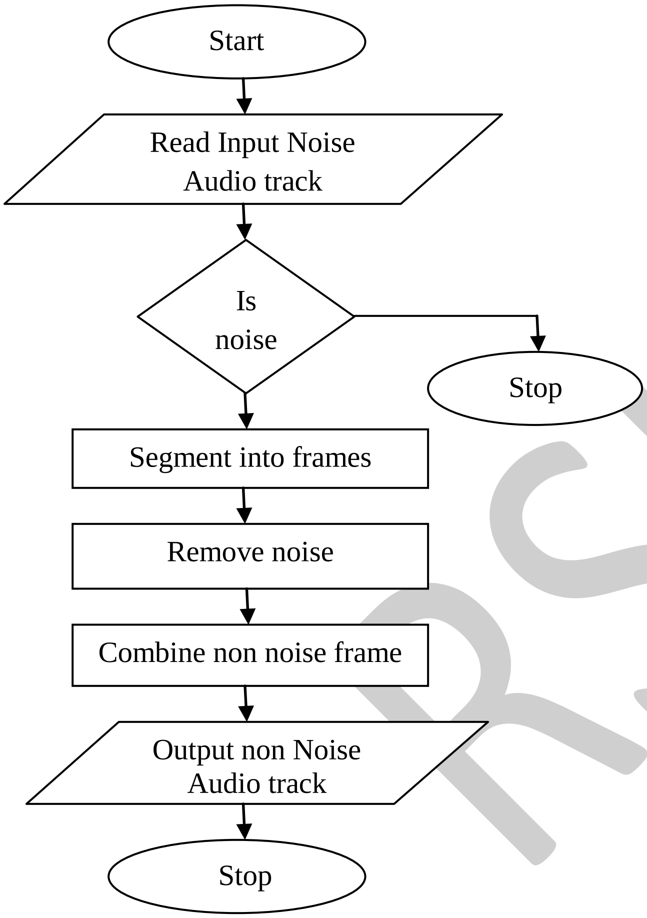
III. METHODOLOGY

In this approach, first we read the input as noise audio track and then segmentation concepts is applied , The we apply the two level speech enhancement concept. First we apply high band pass filtering algorithm and in second level winner filtering concept is applied then we will remove the unwanted Sound from the segmented frames as audio track and finally post processing steps will combine the all the noise removed frames in order to produce the final frame, such as much clear audio track.

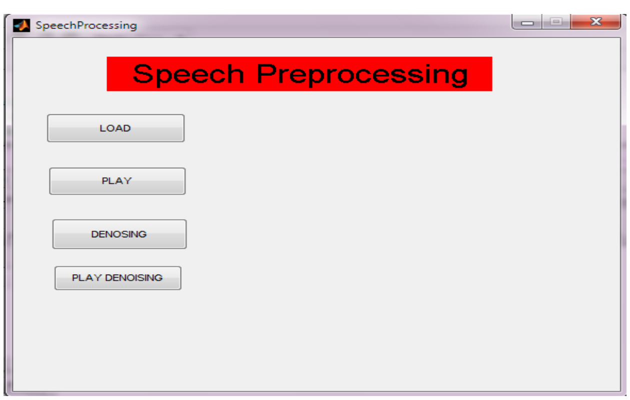
Algorithm:

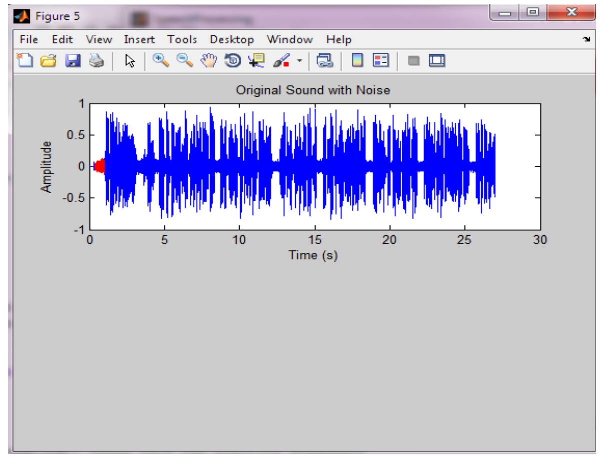
Step 1: Read the input as noise audio track
Step 2: Inputted audio track are segmented into frames.
Step 3: From segmented frame unwanted noise are removed.
Step 4: Finally non noise frames are combined.
Step 5: Gives the much clear audio track.

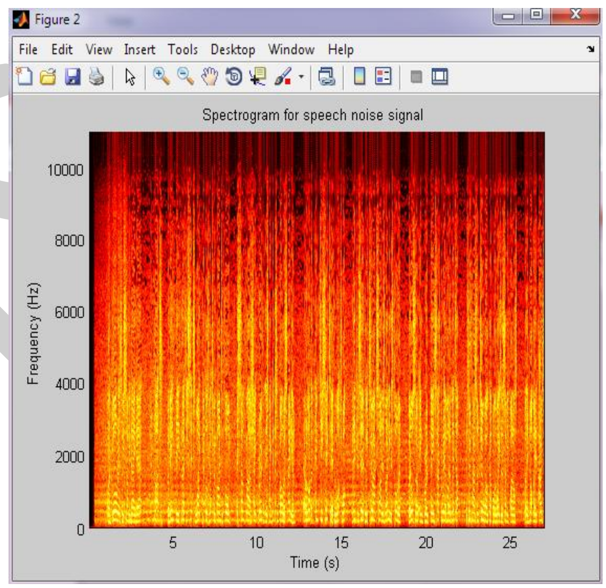
Flowchart:

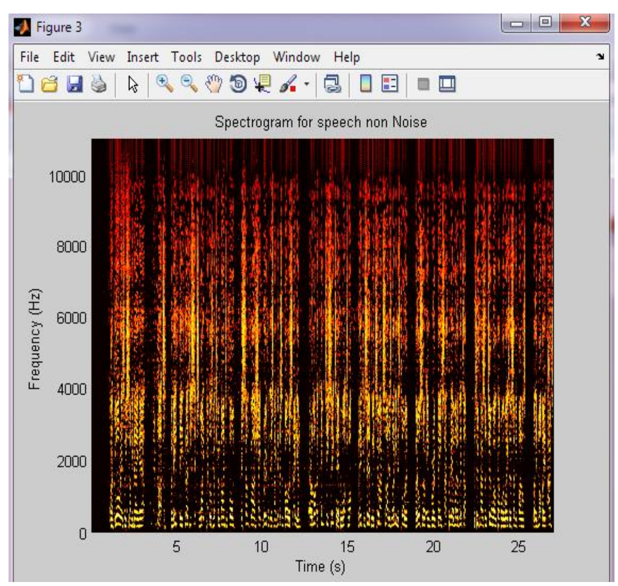


IV. EXPERIMENTAL RESULT









V. CONCLUSION

The speech contain the noise is unable to understand clearly. Here we introduced an two level noise filtering algorithm which produce the much clear speech. But this process will work only for the 20min of audio speech. Further enhancement is to work for long duration of the audio track.

REFERENCE

[1]. Cohen et al ―Noise spectrum estimation in adverse environments: Improved minima controlled recursive averaging‖, IEEE Trans. on speech and audio processing, vol. 11, no. 5, pp. 466-475
[2]. Berdugo B―Noise estimation by minima controlled recursive averaging for robust speech enhancement‖, IEEE Sign al Proc. Letters, vol. 9, no. 1, pp. 12-15
[3]. R martin et al ―Speech enhancement using a minimum mean-square error short-time spectral amplitude estimation,‖ in Proc. IEEE Int. Conf. Acoust., Speech, Signal Processing, pp. 504—512
[4]. Y Ephraim et al - Speech Enhancement Using a Minimum Mean-Square Error Log-Spectral Amplitude Estimator‖, IEEE Transactions On Acoustics, Speech and Signal Processing
[5]. Israel cohen et al - Speech Enhancement Using a Minimum Mean-Square Error Log-Spectral Amplitude Estimator‖, IEEE Transactions On Acoustics, Speech and Signal Processing
[6]. R. Martin and C. Breithaupt, ―Speech enhancement in the DFT domain using Laplacian speech priors,‖ in International Workshop on Acoustic Echo and Noise Control (IWAENC)
[7]. Discrete wavelet transformation
[8]. Linear perceptual coefficient technique