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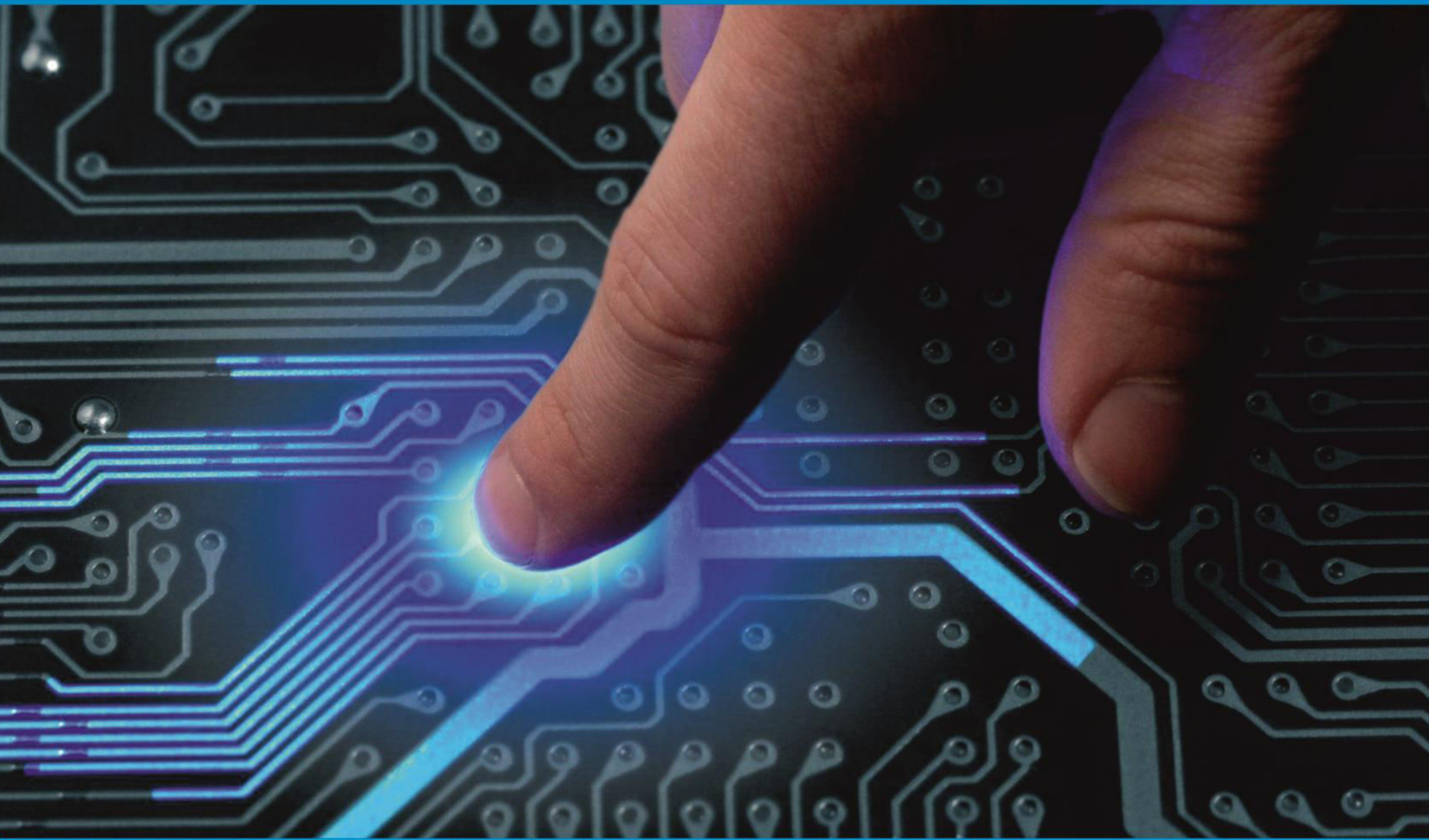

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Single Channel Speech Enhancement Using An Adaptive Wiener Filtering Approach

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ABSTRACT: In our project we have used the application of the Wiener filter in an adaptive manner. The proposed adaptive Wiener filter depends on the adaptation of the filter transfer function from sample to sample based on the speech signal statistics (mean and variance). The adaptive Wiener filter is implemented in time domain rather than in frequency domain to accommodate for the varying nature of the speech signal.

The proposed algorithm is simulated using MATLAB version R2017a. The adaptive wiener filtering approach for speech enhancement to predict the speech quality in both subjective and objective manner with an adaptive gain function under five different noises at five different ranges of input SNR. The results for different SNR values ranges 0, 5, 10, 15, ∞dB shows that adaptive wiener filtering approach improves the speech perception in high SNR values. The simulation results reveal the superiority of the wiener filtering method in the case of Additive White Gaussian Noise (AWGN).

I.INTRODUCTION

Noise reduction is one of the most essential parts of digital hearing aids because hearing-impaired people have difficulty understanding speech, especially in noisy environments. As sound should ideally be processed in real-time with minimum processing delay. Digital hearing aids generally employ single microphone noise reduction algorithms due to size constraints and limitations in computational complexity. Noise reduction techniques can be classified as single channel and multi-channel techniques. Single-channel noise reduction schemes are designed to reduce noise associated with speech signal by using only single microphone. Multi-channel noise reduction algorithms use two or three microphones as they are very tiny. The speech and noise sources usually come from different locations; hence the spectral, temporal, and spatial differences between the required signal and the noise are used.

In our project, we have simulated single channel adaptive wiener filter applicable for hearing aids. Analysis of algorithm with respect to subjective and objective measures was carried out. Subjective analysis using mean opinion scores (MOS) will be conducted using 5 or 6 subjects with normal-hearing and hearing-impaired with mild to moderate sensorineural hearing loss (SNHL). Objective measures using perceptual evaluation of speech quality (PESQ).

II.PROPOSED METHODOLOGY

Noise reduction technique is the process of removing background noise present in the actual speech signal [27]. The classification of noise reduction system is based on the number of microphones of information available for processing into a single channel, dual channel or multi-channel noise reduction. Single channel noise reduction is still an important field because of their simple implementation and effectiveness. The single channel is especially useful in mobile communication applications, where only a single microphone is available due to cost and size considerations. The block diagram of single channel noise reduction system is shown in Fig. 5.1.

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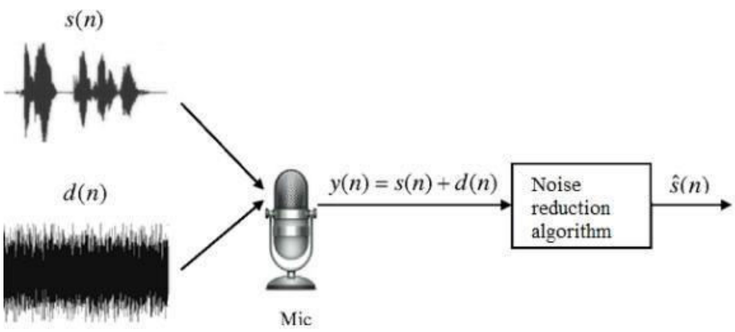

Fig. 5.1 Block diagram of single channel noise reduction system.

In our proposed work, we have used single channel adaptive wiener filter for reducing the background noise and to get an enhanced speech signal. The proposed adaptive wiener filter is discussed in the following section.

Adaptive wiener filter

In this approach, the estimated speech signal mean and variance σ_x^2 are exploited. It is assumed that the additive noise v(n) is of zero mean and has a white nature with variance of σ_v^2. Thus, the power spectrum $P_v(\omega)$can be approximated by: m_x

1) $P_v(\omega) = \sigma_v^2$ (1)

Consider a small segment of the speech signal, in which the signal x(n) is assumed to be stationary, the signal x(n) can be modelled by $x(n) = m_x + \sigma_x \omega(n)$ (2)

Where and are the local mean and standard deviation of $x(n)$. $w(n)$ is a unit variance noise. Within this small segment of speech, the Wiener filter transfer function can be approximated by: $m_x \sigma_x$

2) $$H(\omega) = \frac{P_s(\omega)}{P_s(\omega)+P_v(\omega)} = \frac{\sigma_s^2}{\sigma_s^2+\sigma_v^2} \quad (3)$$

From Eq. (3), because H(ω) is constant over this small segment of speech, the impulse response of the Wiener filter can be obtained by:

$$h(n) = \frac{\sigma_s^2}{\sigma_s^2+\sigma_v^2}\delta(n) \quad (4)$$

From Eq. (4), the enhanced speech signal $\hat{s}(n)$in this local segment can be expressed as:

$$\hat{s}(n) = m_x + (x(n) - m_x) * \frac{\sigma_s^2}{\sigma_s^2+\sigma_v^2}\delta(n) = m_x + \frac{\sigma_s^2}{\sigma_s^2+\sigma_v^2}(x(n) - m_x) \quad (5)$$

If $m_x$and σ_s are updated at each sample, we can say:

3) $$\hat{s}(n) = m_x(n) + \frac{\sigma_s^2(n)}{\sigma_s^2(n)+\sigma_v^2}(x(n) - m_x) \quad (6)$$

In Eq. (6), the local mean $m_x(n)$and $(x(n) - m_x)$are modified separately from segment to segment and then the results are combined. If σ_s^2 is much larger than $\sigma_v^2$the output signal $\hat{s}(n)$ will be primarily due to $x(n)$ and the input signal $x(n)$is not attenuated. If σ_s^2 is smaller than σ_v^2, the filtering effect is performed.

Notice that is identical to when is zero. So, we can estimate $m_x(n)$ in Eq. (6) from $x(n)$ by: m_x m_s m_v

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$$\hat{m}_s(n) = \hat{m}_x(n) = \frac{1}{(2M+1)} \sum_{k=n-M}^{n+M} x(k) \quad (7)$$

where $(2M+1)$ is the number of samples in the short segment used in the estimation.

To measure the local statistics of the speech signal, we need to estimate the signal variance σ_s^2. Since $\sigma_x^2 = \sigma_s^2 + \sigma_v^2$, then $\sigma_s^2(n)$ may be estimated from $x(n)$ as follows:

$$\hat{\sigma}_s^2(n) = \begin{cases} \hat{\sigma}_x^2(n) - \hat{\sigma}_v^2, & if\ \hat{\sigma}_x^2(n) > \hat{\sigma}_v^2 \\ 0, & otherwise \end{cases} \quad (8)$$

Alogrithm

1. Read the clean speech signal sampled at 8000 Hz sampling frequency.
2. Add AWGN (Additive White Gaussian Noise) to the speech signal at different SNR values ranging (0dB, 5dB,10dB,15dB, ∞dB).

Adjust the length of speech signal equal to an integral multiple of frame length.

4. Framing of noisy speech signal with frame duration of 20 msec and 50% overlap.
5. Apply hamming window of frame length duration for each segment.
6. Windowing of each frame using hamming window of frame length.
7. Calculate the mean, variance for local segment.
8. Compute FFT and noisy speech power spectrum for each local segment.
9. Calculate the noise power spectrum of noisy speech signal (for 120ms duration)
10. Calculate the clean speech power spectrum using equation 8
11. Compute the enhanced speech power spectrum using equation no 6.
12. Modified speech spectrum is converted to time domain using IFFT.
13. Re-synthesis of clean speech signal using overlap adds method.

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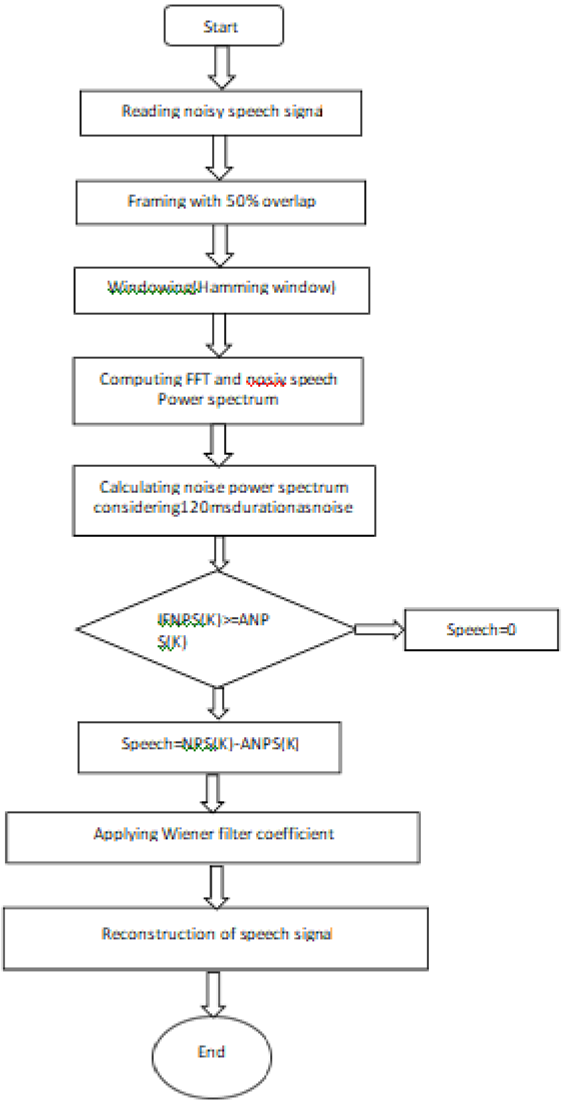

III.SPECTROGRAMS

Figures5.2.1 and 5.2.2 shows the unprocessed and processed speech spectrograms for the sentence "Sky that morning was clear and bright blue".

Unprocessed speech spectrogram

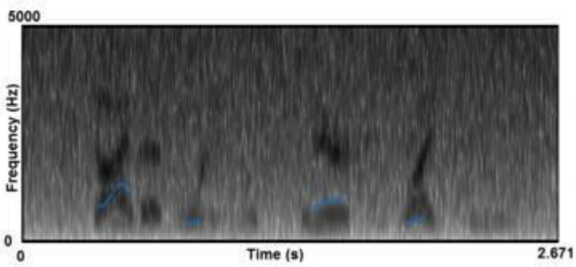

Fig 5.2.1 SNR = 0dB

Processed speech spectrogram

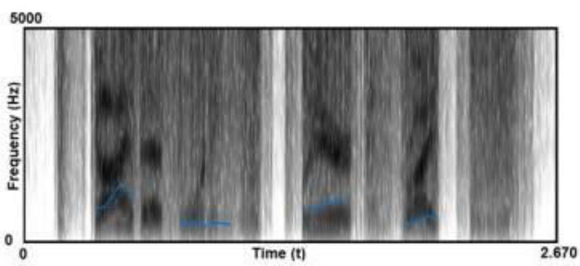

Fig 5.2.2 SNR = 0dB

Figures 5.2.3 and 5.2.4 shows the unprocessed and processed speech spectrograms for the sentence "The lazy cow lay in the cool grass".

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Unprocessed speech spectrogram

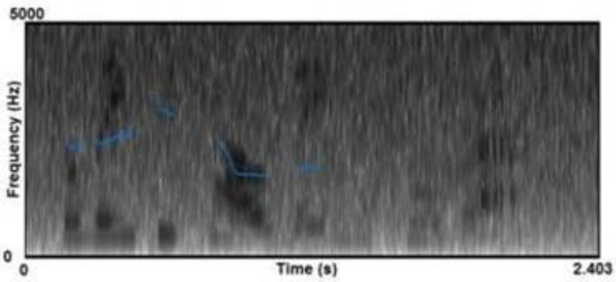

Fig 5.2.3 SNR = 0dB

Processed speech spectrogram

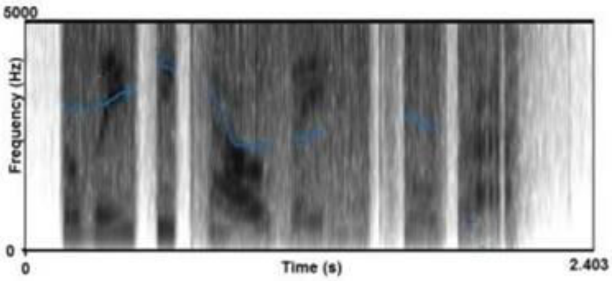

Fig 5.2.4 SNR = 0dB

Figures 5.2.5 and 5.2.6 shows the unprocessed and processed speech spectrograms for the sentence "We find joy in the simplest things".

Unprocessed speech spectrogram

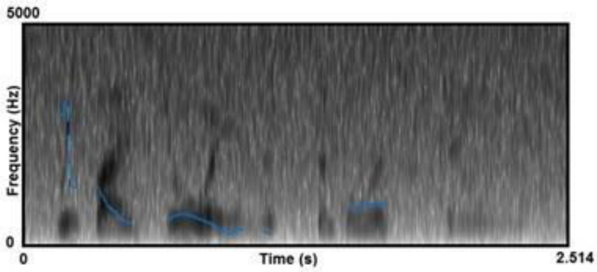

Fig 5.2.5 SNR = 5dB

Processed speech spectrogram

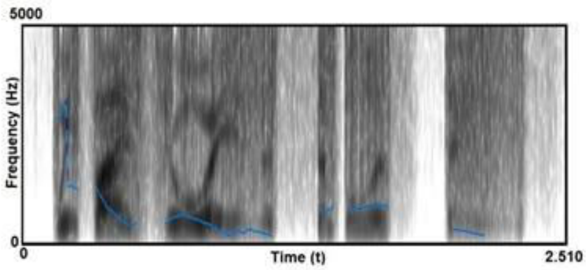

Fig 5.2.6 SNR = 5dB

ADVANTAGES

It removes the additive noise and inverts the blurring simultaneously.

A.F can complete some signal processing tasks that traditional digital filters cannot.

IV.APPLICATIONS

Wiener filter play a central role in a wide range of applications such as linear predication, echo cancellation, signal restoration, channel equalization and system identification

V.CONCLUSION AND FUTURE SCOPE

In this project, we used adaptive wiener filter approach for speech enhancement. The adaptive wiener filtering approach for speech enhancement to predict the speech quality in both subjective and objective manner with an adaptive gain function under five different noises at five different ranges of input SNR.Performance evaluation is done using listening test based on mean opinion score (MOS). The test shows significant improvement in speech recognition score, it includes five different subjects and those are compared to unprocessed speech for SNR values 0,5,10,15, ∞ dB. It is also observed that at higher SNR value the speech recognition score is maximum improvement in speech intelligibility under adverse listening conditions.

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