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Speech Enhancement Using Modified Spectral Subtraction Algorithm

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Abstract: The term "Speech Enhancement" refereed as to improve quality or intelligibility of speech signal. Speech signal is often degraded by additive background noise like babble noise, train noise, restaurant noise etc. In such noisy environment listening task is very difficult at the end user. Many times speech enhancement is used for pre processing of speech for computer speech recognition system. This paper presents speech enhancement methods like Spectral Subtraction and Modified Spectral Subtraction to reduce additive background noise. Basically these methods are single channel speech enhancement methods. Here , we have tested the clean wave file and observe the spectrogram for the noisy wave file and effect of existing spectral subtraction and proposed spectral subtraction algorithm on the file.

Keywords: Speech enhancement, spectral subtraction, modified spectral subtraction, noise, clean wave, and spectrogram

1. Introduction

In speech communication, the speech signal is always accompanied by some noise. In most cases background noise of the environment where the source of speech lies, is the main component of noise that adds to the speech signal. Though the obvious effect of this noise addition is to make the listening task difficult for a direct listener, there are many more far reaching negative effects when we process the degraded speech for some other applications. A related problem is processing degraded speech in preparation for coding by a bandwidth compression system. Hence speech enhancement not only involves processing speech signals for human listening but also for further processing prior to listening. Main objective of speech enhancement is to improve the perceptual aspects of speech such as overall quality, intelligibility, or degree of listener fatigue. Speech enhancement technique through noise reduction due to its simple underlying concept and its effectiveness is enhanced the speech degraded by additive noise. The technique is based on the direct estimation of the short-term spectral magnitude. Noise reduction or speech enhancement algorithms in general, attempt to improve the performance of communication systems when their input or output signals are corrupted by noise. The main objective of speech enhancement is to improve one or more perceptual aspects of speech, such as the speech quality or intelligibility. In this paper, a speech enhancement algorithm using spectral subtraction and modified spectral subtraction algorithm are proposed for hearing aids. NOIZEUS database are used for testing.

1.1 Motivation

Most speech enhancement algorithms have been observed to work well under some conditions. As such the problem of enhancing speech corrupted by a noise source has not yet been fully resolved. While methods based on mathematical and statistical models of speech/noise signals have shown to be effective, they have a key drawback due to the fact that they incorporate some very crucial assumptions about the speech and noise characteristics. However, real-world noise is highly random in nature. Moreover, the spectral content of speech can vary significantly from speaker to speaker and with the emotional state of the speaker. Hence it becomes imperative to exploit as much of the palpable properties of the speech and noise signals as possible.

1.2 Background

The spectral subtraction method is a well-known noise reduction technique most implementations and variations of the basic technique advocate subtraction of the noise spectrum estimate over the entire speech spectrum. However, real world noise is mostly colored and does not affect the speech signal uniformly over the entire spectrum. In this paper, we propose a multi-band spectral subtraction approach that takes into account the fact that colored noise affects the speech spectrum differently at various frequencies. This method outperforms the standard power spectral subtraction method resulting in superior speech quality and largely reduced musical noise.

1.3 Objective

To improve the intelligibility and quality of digitally compressed speech by implementing Spectral Subtraction method for reducing ambient acoustic noise.

2. OVERVIEW OF GENERAL SYSTEM EXECUTION

Speech enhancement process can be achieved by using various speech enhancement algorithms .In this process first speech signal is segmented for 20-30ms i.e. short terms Fourier transform (STFT) is taken and windowed. Hamming window is used for windowing. Then Discrete Fourier transforms (DFT) or Fast Fourier Transform (FFT) of segmented and windowed. Generally in speech enhancement Fast Fourier Transform (FFT) is used. Noisy speech signal is taken. FFT of noisy signal is then given to noise estimation block and speech enhancement block. Noise estimation block estimate the noise during the speech pauses and find the noise spectrum. In most speech enhancement algorithms, it is make assumed that an estimate of the noise spectrum is available. The noise estimate can have a major impact on the quality and intelligibility of the enhanced signal. If the noise estimate is too low, unwanted residual noise will be audible, if the noise estimate is too high, speech will be distorted. The simplest approach is to estimate and update the noise spectrum during the silent segments of the signal. Speech enhancement block enhance noisy speech spectrum to generate clean speech signal.

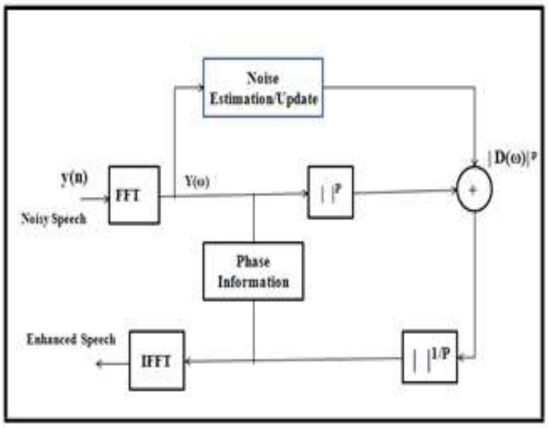

Fig.1.1 Block diagram of spectral Subtraction

A. Basic Spectral Subtraction Method

The spectral subtraction method is historically one of the first algorithms proposed for noise reduction. It is very simple method and easy to implement, it based on the principle that we can obtain an estimate of the clean signal spectrum by subtracting an estimate of the noise spectrum from the noisy speech spectrum. The noise spectrum can be estimated, and updated, during the periods when the signal is absent or when only noise is present i.e. during speech pauses‟. Basic assumption is noise is additive¸ its spectrum does not change with time means noise is stationary or it's slowly time varying signal¸ whose spectrum does not change significantly between the updating periods. Block diagram as shown in figure 3.1.

Let y (n) be the noise corrupted input speech signal which is composed of the clean speech signal x(n) and the additive noise signal d(n). In mathematical equation form we can write y (n) in time domain and Fourier domain as given in equation 1and 2 respectively

$$y(n) = x(n) + d(n) \quad (1)$$

$$Y[\omega] = X[\omega] + D[\omega] \quad (2)$$

Y[w] can be expressed in terms of Magnitude and phase as,

$$Y[\omega] = Y\,|(\omega)|\,e^{j\,\text{Ø}\,y}$$

Where |Y(w)| is the magnitude spectrum and Ø is the phase spectra of the corrupted noisy speech signal. Noise spectrum in terms of magnitude and phase spectra as,

$$D[\omega] = | D[\omega] | e^{j Ø y}$$

The magnitude of noise spectrum | D (w) | is unknown but can be replaced by its average value computed during non speech activity i.e. during speech pauses. In speech enhancement we are keeping phase spectra constant. The noise phase is replaced by the noisy speech phase $Ø_y$ that does not affect speech ineligibility. We can estimate the clean speech signal simply by subtracting noise spectrum from noisy speech spectrum¸ in equation form

$$X(\omega)= [|Y(\omega)|-|D (\omega) |]\, e^{j Ø y} \qquad (3)$$

Where X(ω) is estimated clean speech signal. Many spectral subtractive algorithms are there depending on the parameters to be subtracted such as magnitude (Amplitude) spectral subtraction¸ power spectral subtraction, autocorrelation subtraction. The estimation of clean speech magnitude signal spectrum is

$$X[\omega] = |Y[\omega]| - |D[\omega]|$$

Similarly for Power‟ spectrum is,

$$X[\omega]^2 = |Y[\omega]^2| - |D[\omega]^2| \qquad (4)$$

The enhanced speech signal is finally obtained by computing the Inverse Fast Fourier Transform of the estimated clean speech | X[ω]| for magnitude. Spectrum subtractions and |X[ω]|2 for power spectrum subtraction using the phase of the noisy speech signal. The more general version of the spectral subtraction algorithms is,

$$X [\omega]^p = |Y[\omega]^p| - |D [\omega] |^p \qquad (5)$$

Where p is the power exponent when p=1 yielding the magnitude spectral subtraction algorithm and when p=2 power spectral subtraction algorithm .The spectral subtraction algorithm is computationally simple as it only involves a forward and inverse Fourier transform.

3 Modified Spectral Subtraction

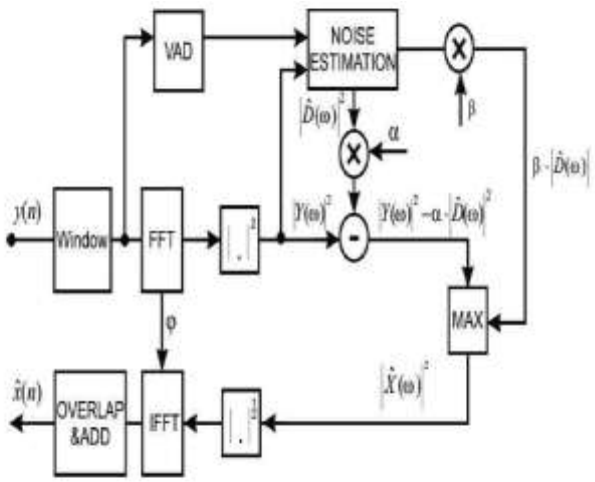

Fig.3.1 Block diagram of Modified spectral Subtraction

Modifications made to the original spectral subtraction method are subtracting an over estimate of the noise power spectrum and preventing the resultant spectrum from going below a preset minimum level (spectral floor).This modifications lead to minimizing the perception of the narrow spectral peaks by decreasing the spectral excursions and thus lower the musical noise effect. Taken a different approach that does not required to access to future information. This Method consists of subtracting an overestimate of the noise power value. This algorithm is given in equation (6), where |X j(ω)|denotes the enhanced spectrum estimated in frame j and |D (ω)| is the spectrum of the noise obtained during non speech activity value. This algorithm is given in equation (6), where |X j (ω)| denotes the enhanced spectrum estimated in frame j and |D (ω)| is the spectrum of the noise obtained during non speech activity.

$$|X_j (\omega)|^2 = |Y_j (\omega)|^2 - |D (\omega)|^2$$

if $|Y_j (\omega)|^2 - (\alpha+\beta)|D (\omega)|^2 = \beta\, |D (\omega)|$ else (6)

With $\alpha > 1$ and $0 < \beta \leq 1$.

Where α is over subtraction factor and β is the spectral floor parameter. Parameter β controls the amount of residual noise and the amount of perceived Musical noise. If β is too small, the musical noise will became audible but the residual noise will be reduced .If α is too large, then the residual noise will be audible but the musical issues related to spectral subtraction reduces.

Parameter α affects the amount of speech spectral distortion. If α is too large then resulting signal will be severely distorted and intelligibility may suffer. If α is too small noise remains in enhanced speech signal. When $\alpha > 1$, the subtraction can remove all of the broadband noise by eliminating most of wide peaks. The parameter α varies from frame to frame according to as given below,

$= \alpha_o - 3/20\ \text{SNR} - 5\ \text{dB}$

$< \text{SNR}\ 20\text{dB}$

4. Result and Discussion

The modified Spectral Subtraction algorithm was simulated using Mat lab. Figure 1 shows the result of SS algorithm and Modified SS algorithm showing the output of algorithm for clean wave and noisy wave .Generally spectral subtraction is used for speech enhancement. Figure 2 shows the clean wave spectrogram output. Figure 3 shows the spectrogram of noisy speech signal corrupted by Babble noise at 0dB SNR and enhanced speech signal by Spectral Subtraction. The performance of spectral subtraction algorithm and proposed spectral subtraction algorithm is evaluated by according to NOZEUS database. The modified algorithm was simulated using Mat lab. Figure 4 shows the existing spectral subtraction output. Figure 5 shows the desired signal derived from the input signal shows the output of proposed spectral subtraction output which is highly stable as compare to spectral subtraction algorithm.

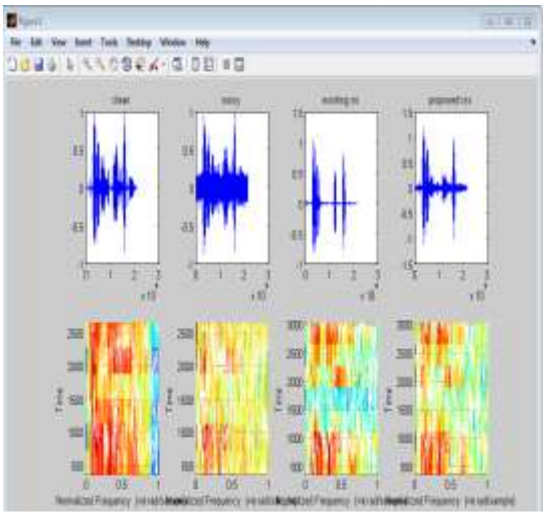

Fig 1 Result of SS algorithm and Modified SS algorithm

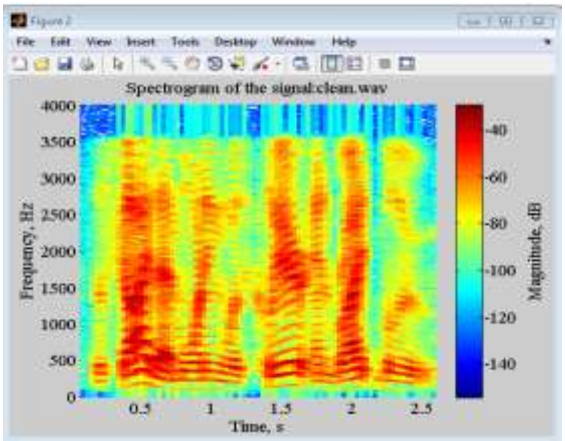

Fig 2 Graphical representation of spectrogram of the clean signal

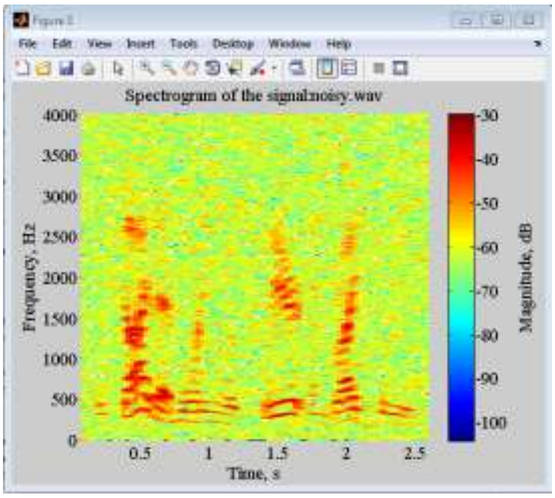

Fig 3 Graphical representation of spectrogram of the noisy signal

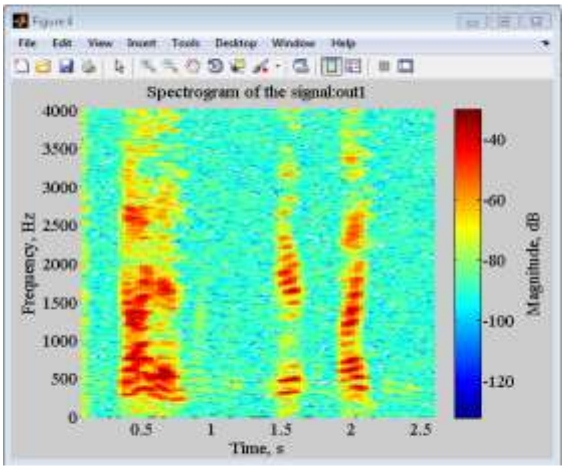

Fig 4 Graphical representation of spectrogram of the spectral subtraction signal

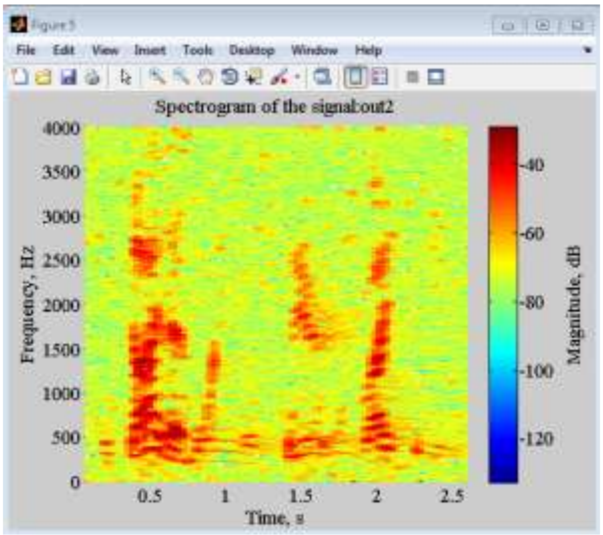

Fig 5 Graphical representation of spectrogram of the improved spectral subtraction signal

4. Conclusion

The main objective of the speech enhancement is to bring up the performance in the presence of noise and echo interference to the performance obtained with pure speech signals, which is the ideal case. Thus, our aim was to approach the performance of single channel based speech enhancement techniques to that in the case of ideal signal. Another objective of this work is to compare the performance of SS Algorithm and Improved SS Algorithm.

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