Review Paper of Speech Enhancement Techniques using Wiener Filter and Subspace Filter

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Abstract— In the speech enhancement method by using the wiener filter and subspace filter. Because of uses advantages in reduction in noise with the subspace speech enhancement technology and stable characteristics of the wiener filter. These proposed enhancement of speech method has a better performance. It can be removed colored noise from noisy speech signal .The proposed enhancement of multi-channel speech signal can be obtain a better speech recovery result as compared to the trandition multichannel wiener filter and the subspace filter.

Key words: Enhancement Speech, Wiener Filter, Subspace Filter

I. Introduction

Speech is most important factor of communication for human. Speech can be defined may be delivering thoughts and ideas with the help of vocal sound. Speech captured by microphones in the hearing aids is always corrupted by additive noise [5]. Speech needs to be clean off irrelevant contents. However, removed the irrelevant information. The object of this paper is to enhancement of the speech quality signal[1-3].

Enhancement of speech has been studied of many applications such as voice communication, transmitted speech signal and voice control [1]. Noise is everywhere around most of the places we feel is silent will have noise floor well below the full scale level. During the conversation on mobile phone between the person A and person B then these conversations is meaningful speech conversation. The direct sound contaminated by early and the late reflections. This paper will remove the additive noise from the signal recorded and to improving the speech quality, improving speech intelligibity and speech recognition rates [1-6].

II. Speech enhancement

Speech enhancement aims to improve quality of speech signal by using various technique. The main objective of enhancement is improvement in intelligibility and/or overall perceptual quality of degraded speech signal using audio signal processing techniques [1]. Speech enhancement degraded by noise, or noise reduction, is the most important field of speech enhancement, and used for many applications such as mobile phones, VoIP, teleconferencing systems, speech recognition, and hearing aids[6].

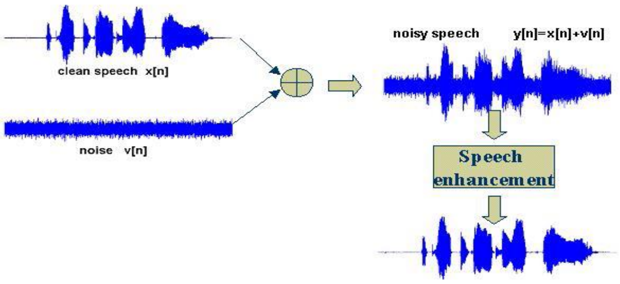

Fig. 1: Basic overview of speech enhancement system.

III. Speech enhancement method

The speech enhancement techniques can be divided into three categories [7].

A. *Time-Domain Methods*

Time domain methods provide trade-off between speech distortion and residual noise. Example of time domain method is a Subspace approach.

B. *Frequency-Domain Methods*

Frequency domain methods include the minimum mean square error (MMSE) estimator, spectral substraction, and the wiener filtering. And in this method provides advantages to the real-time processing with the less computational load.

C. *Time-Frequency Domain Methods*

Time frequency domain method can be involved family of wavelets.

IV. Wiener fitter

The aim of the Wiener filter is to filter out noise that has corrupted a signal. The basic principle of the wiener filter is to obtain clean signal from corrupted by additive noise. The wiener filter is based on a statistical approach. The design of the Wiener filter takes a different approach one is assumed to have knowledge of the spectral properties of the original signal and the noise, and other is seeks the linear time-invariant filter whose output would come as close to the original signal as possible.

1) Assumption: The assumption of the wiener filter is a signal and (additive) noise are stationary linear Stochastic processes. But the known spectral characteristics or known autocorrelation and cross-correlation.
2) Requirement: In the wiener filter the filter must be physically realizable/causal.
3) Performance criterion: The estimation is obtained by minimizing the mean- square error (MMSE) between the desired signal and the estimated signal.

V. Subspace filtering

A. *Fundamentals*

Any of the noise reduction technique requires assumptions about the nature of the adding noise signal. Subspace based speech enhancement also makes some basic assumptions of the desired signal (clean speech) as is the case in many but not all signal enhancement algorithms [7]. The separation of the speech and noise signals will be based on their different characteristics. Since the characteristics of the speech (and also of the noise) signal(s) are time varying, the speech enhancement procedure is performed on overlapping analysis frames.

B. Speech Signal

A key assumption in all subspace-based signal enhancement algorithms is that every short time speech vector s = [s(1),s(2),...,s(q)]T can be written as a linear combination of p<q ,i=1,..., p,

$$s=My \quad (1)$$

Where M is a (q × p) matrix containing the basis functions and it can be a column wise ordered and y is a length-p column vector containing the weights.

An choice for mi are sinusoids motivated by the traditional sinusoidal model for speech signals. A observation here is that the consecutive speech vectors s will occupy a (p<q) dimensional subspace speech signal of the q-dimensional Euclidean space (p equals the signal order). Because of the time-varying nature of speech signals, the location of this signal subspace and the dimension of the signal subspace will consequently be frame dependent.

C. Noise Signal

The additive noise is assumed to be zero mean, white, and uncorrelated with the speech signal. Its variance should be slowly time varying such that it can be estimated from noise only segments. The speech signal consecutive noise vectors n will occupy the whole q-dimensional space.

D. Speech/Noise Separation

Based on the above description of the speech and noise signals, the mentioned q-dimensional observation space can be split up into two subspaces. One is namely a p-dimensional (signal + noise) subspace in which the noise interferes with the speech signal, and other is a (q−p) dimensional subspace that containsonl noise (and no speech). The enhancement of speech signal procedure can now be summarised as follows:

1) Separate the (signal+noise) subspaces from the (noise- only) subspace,
2) Remove the (noise-only) subspace,
3) Optionally, remove the noise components in the (signal + noise) subspace.

The first operation is straight forward for the white noise condition under consideration here, but can become complicated for the colour noise case. The second operation is applied in all implementations of subspace based signal enhancements, whereas the third operation is indispensable to obtain an increased noise reduction. Nevertheless, the last operation is sometimes omitted because of the introduction of speech distortion. The latter problem is inevitable since the speech and noise signals over lap in the signal subspace.

VI. SPEECH QUALITY MEASUREMENT

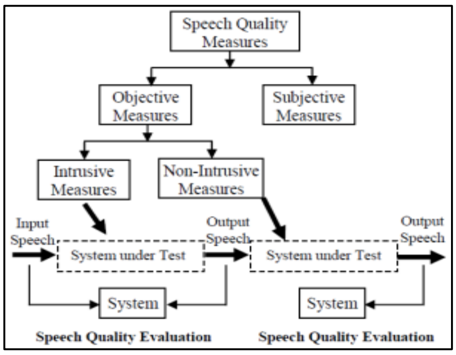

Fig. 2: The classification of speech quality measurement

It is very essential to the software of interest in order to evaluate its performance based on sound quality[6-7]. Speech Quality Measurements are of two types:

– Objective measurements
– Subjective measurements

In this paper only work can be focus in objective speech enhancement technology. Because subjective speech enhancement technology are time consuming and expensive.

VII. CONCLUSION

Various speech enhancement approaches has been approached in this paper. Because of using wiener and subspace speech enhancement technology both of these advantage in noise reduction of the subspace speech enhancement technology and the stable characteristic of the Wiener filtering technology. The proposed multi-channel speech enhancement has a better performance in robustly removed colored noise from noisy speech signals. In these paper work focuses on the improving the quality of speech signal and improving the speech intelligibility and speech recognition rates.

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