

A Review on Speech Enhancement System using FIR Filter

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Abstract - Speech Enhancement is necessary for many applications in which clean speech signal is important for further processing. The speech enhancement techniques mainly focus on removal of noise from speech signal. The present system of speech enhancement is developing by adaptive filtering approach using digital filter. FIR filter is used widely to remove the noise from noisy speech signal. By using adaptive filtration method. FIR is digital filter and it acts as low pass filter to remove high frequency noise component. The adaptive coefficient is adjusted in such a way that error converge become zero. When error converge is zero then input signal is equal to output signal without noise. The error of converge is adjusted according to Lyapunov stability theory. These method is useful where stability and accurate result is demanded. The algorithm is suitable for linear and nonlinear signal. For nonlinear signal, adaptive coefficient is adjusted time to time on information available.

Key Words: Speech Enhancement, Adaptive Algorithm, FIR Filter, Noise cancellation.

1. INTRODUCTION

Speech is one of best communication mediums and is widely used in mobile communication. But spoken speech signal gets distorted due to noise present at surrounding. Hence, need of speech enhancement is essential in order to recover the original signal [1]. Many different techniques are used for speech enhancement, some of them are spectral subtraction, active noise cancellation and dual microphone. But these techniques fail at some point to achieve desired result. These traditional methods do not give the desired result where stability and non-linearity are needed [2]. LMS (i.e. least mean square) cannot apply here because it needs heavy mathematical formulation and has high complexity. By performing all these techniques, still some background noise is present which could not achieve good SNR ratio and thus, some original information is vanished [3]. Hence, FIR filter is used to enhance speech signal. It removes the noise from noisy speech signal to get noise free signal with improved quality and intelligibility. FIR filter is used as adaptive controller. These adaptive controller adjust according to Lyapunov stability theory and information available. This algorithm is simple, requires less mathematical calculation, ideal size and power consumption. Once adaptive coefficient is set, algorithm parameters are automatically adjust according to available information.

2. LITERATURE REVIEW

There are many different methods used to remove noise from speech signal. Many of them achieve different parameter [1]. Firstly spectral subtraction which works on step size of signal is used to reduce background noise. It subtracts the noise signal but with noise signal some sort of information is also removed or corrupted. The two type of subtraction is done over subtraction and nonlinear subtraction. Normally, in nonlinear subtraction it produces the distorted signal.

Dual microphone is another method which uses separate microphone to record noise and speech signal. It achieves fast converge and stability to some extent but complex circuit make the damage. Next is active noise cancellation. It uses FL algorithm to remove noise. But, with active noise some information is also lost and hence many useful harmonic component is lost. Many traditional LMS method is also used as noise removal. Some of them are concept of adaptive algorithm but they do not achieve desired result. In many traditional method they concentrate on step size of speech signal rather than adjusting the parameter according to reference signal. Some of them use winner filter on perceptual base but when we increase signal power mean square error is also increased. CC-WSLMS is used as a combinational adaptive filter on the basis of updated step width. It takes filter of different tap length. In the proposed system algorithm we define each and every parameter. Here, filtering frequency is defined over 300 Hz to 3.4 KHz. This frequency is nothing but sound producing capability of human. The length of filter tap is decided according to type of speech signal which is needed to filter.

3. ADAPTIVE ALGORITHM

Speech enhancement using FIR filter sets adaptive coefficient algorithm. It is nothing but procedure to solve problem according to the sequence of specific action. Similarly, adaptive algorithm means it changes its behavior according to time and information available. Algorithm parameters are used for processing signal that change according to some criterion. Here, we deal with non-linear, linear signal also enhancement parameter. While processing nonlinear signal adaptive filter parameters are continuously changing in order to meet performance criterion. And filtering signal always matched with reference signal. The signal is continuously filtered according to adaptive coefficient and noise is removed. But before going to result it refers reference signal.

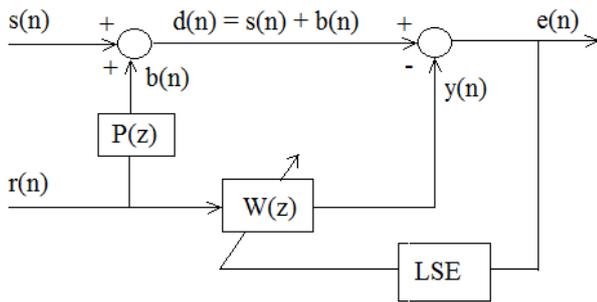


Fig -1: Block diagram of speech enhancement system using adaptive algorithm

The block diagram of adaptive algorithm is shown in fig 1, $r(n)$ is the noise reference signal. $p(z)$ is primary path between noise signal and speech reference signal. The $b(n)$ is background noise and $s(n)$ is desired speech signal. The combination of $b(n)$ and $s(n)$ produces noise signal. $W(z)$ represents control filter whose filter tap is already known. $e(n)$ is error signal which is ideally zero.

4. PROPOSED SYSTEM

Speech signal are continuous in nature but when it is stored in computer or any storing device it becomes discrete in nature. Then stored signal is used for computational purpose. In computational processing, the wave of speech signals are directly stored in MATLAB Tool. The voice frequency of human is 300 Hz -3.4 KHz. The audio range of human ear is 20 Hz-20 KHz and many signals lie in the same range. Hence, it is hard to identify the signal. When sound is produced by human lies in the audible frequency range many unwanted sound or noise is present causing signal to get distorted. In order to remove such a serious error from speech signal, FIR filter is used.

Here, FIR filter acts as a low pass filter which blocks high frequency component because noise is high frequency component. The flow chart of speech enhancement technique is shown in fig 2. The first block contains speech signal which is corrupted by some sort of noise. Signal is continuous in nature but for computational purpose it can be stored in computer or storing device. Hence, it becomes discrete signal. Here, signal is any linear or non-linear and different type of noise is taken. In second block consists of noisy signal and it cuts into different frame. This process is known as Framing. Windowing is nothing but selection of particular frame containing noise. Framing and windowing is one of most important blocks of proposed system. Every frame have different adaptive coefficient proportional to noise and level of noise component. The length of filter tap depends upon framing and windowing block. Fast convergence of error depends on how many filter tap are used.

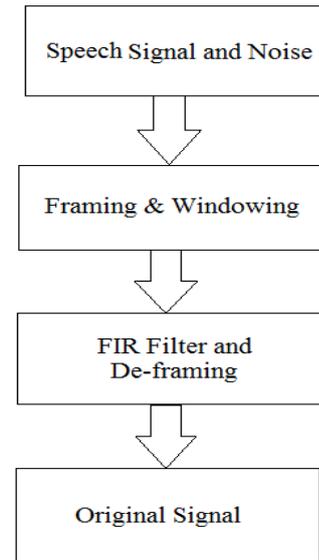


Fig -2: The overview of speech enhancement technique

Third block contains FIR filter with adaptive coefficient which is set according to available information. For nonlinear signal the parameter are adjusted time to time automatically and removed noise. The filter signal is always referred the reference signal before giving the final result. MATLAB Tool can be used to process the signal and in the end original signal or noise free signal is obtained.

5. CONCLUSIONS

Noise corrupts the signal and for proper communication noise free signal is very important. Hence, using speech enhancement technique in order to remove the noise is important. Proposed FIR filter method uses adaptive coefficient for linear and non-linear signal for stability purpose. Adaptive coefficient is adjusted according to lyapunav stability theory so that error is made smaller and ideal error of converge and process toward analytical solution are achieved. The performance of system is better than traditional method.

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