CHAPTER III

MOTIVATION FOR RESEARCH

3.1 MOTIVATION

In the past decades, research in the field of speech enhancement has focused on the suppression of additive background noise. The presence of background noise in speech significantly reduces the intelligibility of speech. Degradation of speech severely affects the ability of person, whether impaired or normal hearing, to understand what the speaker is saying. The ultimate goal of speech enhancement is to eliminate the additive noise present in speech signal and restore the speech signal to its original form. Several methods have been developed as a result of these research efforts. Most of these methods have been developed with some or the other auditory, perceptual or statistical constraints placed on the speech and noise signals. However, in real world situations, it is very difficult to reliably predict the characteristics of the interfering noise signal or the exact characteristics of the speech waveform. In this research the main focus is on the enhancement of speech for human listeners and for speech processing systems in noisy environment conditions, as this is the most commonly encountered problem. The main objectives of the research are:

- To investigate the FFT/IFFT processor and to propose new efficient pipelined architecture for real-valued FFT/IFFT processor, thereby making the proposed method more suitable than the existing processor for speech processing applications like speech recognition, speaker recognition and digital hearing aids.
- To compare the performance of the proposed schemes with existing methods by conducting both objective and subjective test.
- To overcome the problem of matrix inversion in the design of Wiener filter, the noise degradation system for noisy speech signals in Digital Hearing Aid is implemented and incorporating a powerful and efficient hardware wiener filters.
- To find whether the proposed speech enhancement method are suitable to increase the recognition accuracy of speaker recognition system under real life noise conditions.
3.2 AN EFFICIENT PIPELINED ARCHITECTURE FOR REAL-VALUED FAST FOURIER TRANSFORM

Real valued FFT plays a major role in today’s digital world as most of the signals contain real values. The design was used earlier pipelined RFFT was a four-parallel model for radix-2 RFFT, and different structures have equipment for comparable many-sided quality, various methodologies are accomplished for the same models. For radix-2^3 RFFT calculation, more multipliers are required for the design of two parallel. The conventional techniques require more hardware space with high power consumption in the FFT computation that is the most important task for a researcher during the VLSI architecture design. By using symmetric property analysis of the real-valued signals, the drawbacks of conventional technique can be eliminated. The symmetric property is adopted, and an efficient pipelined architecture for 16-point DIF FFT is designed. The scheme of pipeline reduces the time taken for processing at the cost of some registers and to focus on power reduction and the modified complex multiplier with less internal real multipliers that are replaced by a modified canonical signed digit multiplier (CSDM) with the technique of resource sharing technique.

3.3 SPEECH SIGNAL ENHANCEMENT IN DIGITAL HEARING AID

The problem of matrix inversion was considered in the Wiener filter design and the system of noise degradation of noisy speech signals in Digital Hearing Aid can be implemented that incorporates power and hardware efficiency of the Wiener filter. The Wiener filter with pre-processing and post processing for the input speech and noise signal in time domain is transformed to the frequency domain and processed by pre-designed real-valued FFT processor. The FFT is defined with complex data, but the input is real in many applications. The algorithm of real-valued FFT algorithms had properties of symmetry in the FFT and had a higher speed over complex algorithms of the same length. The higher throughput area and lower latency can be achieved. The implementation of RFFT can be preferred rather than FFT processor since a real-valued signal is the input signal. The modified Wiener filter application can be efficient power spectrum and the technique of energy analysis. The floating point adders with low power and multipliers are adopted to contribute the reduction of reduction.
3.4 CONTRIBUTIONS TO THE PROPOSED WORK

- Current speech enhancement methods belong to two categories namely, time domain methods such as the subspace approach and frequency domain methods such as the spectral subtraction, Minimum Mean Square Error (MMSE) estimator and Wiener filtering. Both methods have their own advantages and drawbacks.

- The subspace methods provide a mechanism to control the trade-off between speech distortion and residual noise, but with the cost of a heavy computational load. Frequency domain methods, on the other hand, usually consume less computational resources, but do not have a theoretically established mechanism to control trade-off between speech distortion and residual noise.

- Among them, spectral subtraction is computationally efficient and has a simple mechanism to control trade-off between speech distortion and residual noise, but suffers from a notorious artifact known as “musical noise”.

- The MMSE estimators and Wiener estimator have a moderate computation load, but have no mechanism to control trade-off between speech distortion and residual noise. The main contribution is a low power and hardware efficient matrix inversion module design, has used in the decomposition of QR with givens rotation. As the process of real-valued FFT/IFFT is done using a single processor and also by the utilization of a modified analytic method and the design is an efficient system of speech signal enhancement.