CHAPTER 1

INTRODUCTION

1.1 Overview

This project is about Speech recognition system using Digital Signal Processing (DSP). This system can recognize speech of makhraj such as ا alif, ب ba, ت ta, ث tsa, ج jim, ح ha. Recognition is generally more difficult when vocabularies are larger or have many similar-sounding words. The Digtal Signal Processing (DSP) programmed to act any kind of filter. This recognition system use signal filtering to reduce noise in waveform. According to the signal filtering, this project use analog and digital filter such as Finite Impulse Response (FIR) and Infinite Impulse Response (IIR) digital filter. IIR digital filter one of the techniques is essentially based on transformation of efficient analog filters such as Butterworth and Chebyshev filters, into corresponding digital filters. One of the most widely uses method of FIR digital filter design is the window method. To perform this project, a simulator program, Matlab is applied in speech recognition to assign an input pattern in DSP. According to the Matlab software, there is a Filtering Toolbox that helps this project to successful.

To make pattern recognition easier, the Pulse Code Modulation (PCM) digital audio is transformed into the "frequency domain"[1]. Transformations are done using a windowed fast-Fourier transform. The output is similar to what a spectrograph produces. In frequency domain, it can identify the frequency components of a sound. From the frequency components, it’s possible to approximate how the human ear perceives the sound. Unfortunately, this does not work because of a number of reasons such as a user speaks a word it sound different. User do not produce exactly the same sound for the same phoneme and also the background noise from the microphone and user’s office causes the recognizer to hear diferent vector than it would have if the user was in quiet room with a high quality microphone.

1.2 Objective

The objectives of this project are:

i. To design a makhraj recognition system based on human voice.

ii. To design a system based on FIR filter.

1.3 Scope of Project

i. To develop a makhraj speech recognition system based on human voice. These systems able to identify the user have difference speech of makhraj alphabet from ا alif, ب ba, ت ta, ث tsa, ج jim, ح ha.

ii. To remove noise from human voice that produces clear speech makhraj recognition by using FIR filter.

1.4 Problem Statement

Today’s voice communication systems, with their increasing demand for user comfort, necessitate a growing focus on acoustic noise suppression and echo reduction A Practical Approach addresses proven methods for suppressing acoustic echoes and noise in various sound systems, from hands-free telephones to video conferencing systems, hearing aids, and speech recognition systems.