1.1 Preface

The fundamental purpose of speech is communication, i.e., the transmission of messages. According to Shannon's information theory, a message represented as a sequence of discrete symbols can be quantified by its information content in bits, and the rate of transmission of information is measured in bits/second (bps) [1].

speech coding has been and still is a major issue in the area of digital speech processing in which speech compression is needed for storing digital voice and compression makes it possible to store longer messages, speech compression is often referred to as speech coding which is defined as a method for reducing the amount of information needed to represent a speech signal. Most forms of speech coding are usually based on a lossy algorithm. Lossy algorithms are considered acceptable when encoding speech because the loss of quality is often undetectable to the human ear. Linear predictive coding (LPC) is one of the most powerful speech analysis techniques, and one of the most useful methods for encoding good quality speech at a low bit rate and provides extremely accurate estimates of speech parameters [2].

Under normal circumstances, speech is sampled at 8000 samples/second with 8-bits used to represent each sample. This provides a rate of 64000 bits/second. Linear predictive coding reduces this to 2400 bits/second. At this reduced rate the speech has a distinctive synthetic sound and there is a noticeable loss of quality [3].

1.2 Problem Statement

Speech coding needs high bit rate for representation, therefore, greater bandwidth is required, which increases cost. However, the decreasing of the data rate affects negatively on the quality.

1.3 Proposed Solution

The proposed solution to overcome the above problem is to use the linear predictive coding technique (LPC) vocoder, with optimization of some of its parameters.

1.4 Objectives

The aim of the project is to represent the samples of a speech signal with a minimum number of bits with acceptable quality; this can be achieved through detailed objectives as follow:

- ✓ To mathematically represent the LPC parameters.
- ✓ To simulate the parameters of the LPC.
- ✓ To implement the LPC vocoder in real time.
- ✓ To enhance the quality of synthesize speech signal.

1.5 Methodology

To accomplish this project, the work was divided into four phases:

1. In the first phase the details of digital signal processing and its techniques such as convolution and correlation were studied.
2. The procedures of LPC were reviewed in the second phase, in which, the process of linear predictive coding technique is to transmit information about the voice signal not the signal itself by transmitting parameters, which enable the receiver to re-synthesize the original signal at low bit rate within acceptable quality. Vocal tract is represented as a time-varying filter and speech is windowed about

every 22.5ms. For each frame, the voiced and unvoiced detection, gain, pitch period and only 10 of the coefficients of a linear prediction filter are coded for analysis and decoded for synthesis.

3. The simulation by matlab was created and tested the output voice quality by different techniques in the third phase.
4. The LPC vocoder was implemented in DSK C6713 chip in the fourth phase.

1.6 Research Outlines

The reminder of the document is organized in the following manner:

Chapter Two describes the speech production model, classification of speech coder, basic concepts of linear predictive coding as well as some of its applications, and background research relevant to LPC.

Chapter Three presents the operations of linear prediction code and the mathematical representation of the parameters of speech segment.

Chapter Four discusses the results of simulation and implementation.

Chapter Five draws the conclusion and the future ideas that can be performed.