DESIGN OF INTERPOLATION FILTER IN DELTA MODULATION TECHNIQUE FOR MULTIRATE DIGITAL SIGNAL PROCESSING

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ABSTRACT

Delta Modulation is a simple and robust technique for speech coding implementation and yet offers the best technique for multirate digital signal processing. The performance of digital signal processing and communication system is generally limited by the precision of the digital input signal which is archived at the interface between analog and digital information. This project will focus on designing an Interpolation Filter in Delta Modulation (DM) Technique for Multirate Digital Signal Processing. Method to be used is a kind of waveform coding that is Delta Modulation. This project explores DM because it only used a single bit Pulse Code Modulation (PCM) to achieve digital transmission of analog signal. Therefore, the bits rates associated with DM are lower than conventional PCM. Since DM transmits signals at lower bit rates, the bandwidth consume will reduce. Therefore, this project will require a design of interpolation filter at the receiver. The developed model using the SIMULINK in MATLAB environment will be test using the real speech and the performance of the speech quality can be measured based on objective measurement that is signal to noise ratio (SNR) [1] [8]. There are four different techniques of interpolation filters used in this project. The interpolation factor that were propose in this project are L=2 and L=4. At the end of this project, the interpolation filter in the Delta Modulation system has managed to filter the speech that was transmitted.

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CHAPTER 1

INTRODUCTION

1.0 INTRODUCTION

In recent years, much progress has occurred regarding speech coding technology on the telephone band. This has accelerated the development of international, mobile and satellite communications. Although current techniques allow a telephone band network quality speech transmission rate at 64kbit/s that's standard for Pulse Code Modulation (PCM), there are still demand for lower rates.

Speech coding attempts to achieve toll quality performance at a minimum bit rate so that it can improve the efficiency of the transmission, reduce cost, increase security and robustness in transmission [1]. All of this is the advantages in using the lower rates in transmission. The main target of all speech coding systems is to transmit speech with the highest possible quality using the least possible channel capacity.

This project used Delta Modulation due to its simplicity in hardware implementation with multirate technique to achieve lower bit rate transmission and also to fit speech signals into a considered ideal transmission channel. The speech signal is to be transmitted at lower bit rate than standard practice. Then to obtain the acceptable performance in perception, an interpolation is to be designed at the receiving end. At this point, this project aim is to design the up sampling filter at receiving end. Beside that, this project will propose several different techniques of interpolation filters. This is important so that all the result can be compared and from this, the performance of filters can be recognized [1, 3].

The goal of speech coding is to represent in digital form with as few as possible while maintaining the intelligibility and quality required for the particular application. Interest in speech coding is motivated by the evolution to digital communication and the requirement to minimize bit rate hence conserve bandwidth.