Design of Interpolation Filter in Delta Modulation Technique for Multirate Digital Signal Processing

Asraf Ahmad

Bachelor of Engineering (Honours) Electrical
Faculty of Electrical Engineering
Universiti Teknologi MARA Malaysia
40450 Shah Alam, Selangor
Email: asrafahmad85@yahoo.com.my

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 [6]. 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 [1]. 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.*

Keywords: Delta Modulation, Multirate Digital Signal Processing, SIMULINK, Signal to noise ratio

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 high 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. All of this is the advantages of 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 [5]. The speech signal is to be transmitted at lower bit rate than standard practice. Then to obtain the acceptable performance in perception, an interpolation filter is to be designed at the 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.

2.0 OBJECTIVES

The objective of this project is to design interpolation filter at the receiver for Delta Modulation. The main purpose of this interpolation makes up the signal that transmits during the transmission in the channel. Beside that, this interpolation filter will reconstruct the voice that was transmitting from input so that the quality of the voice will be maintained. The interpolation filter is important because there will be loss in the voice when it's transmitting from the input. As a result, the interpolation filter is design to solve the problem.

3.0 SCOPE OF WORK

This project was conducted by using Simulink and MATLAB device as simulation software to design the circuit of interpolation filter in Delta Modulation technique. Method to be used is waveform coding because of its relatively simple algorithms, better adaptive capability and better speech quality.

There were two types of interpolation filter that have been designed. The designing include standard interpolation filter and FIR interpolation filter. Each of the filter are design at two interpolation factor that are L=2 and L=4.

All of this design will be implemented and evaluated in SIMULINK and MATLAB. Finally, the performance of the designs will be tested to compare their respective performance. Performance of design will be measured with objective measurement and subjective measurement.

4.0 METHODOLOGY / DESIGN PROCESS

This chapter will explain more detail on the process involved in modeling and analysis the performance of Delta Modulation Technique for Multirate Digital Signal Processing [4] [5] [7].

It will include the step of the setting for each block in the Delta Modulation design and also the block design for the interpolation filter. The different for each block in interpolation filter is shown in this chapter.

4.1 FLOW CHART FOR THE PROJECT IMPLEMENTATION

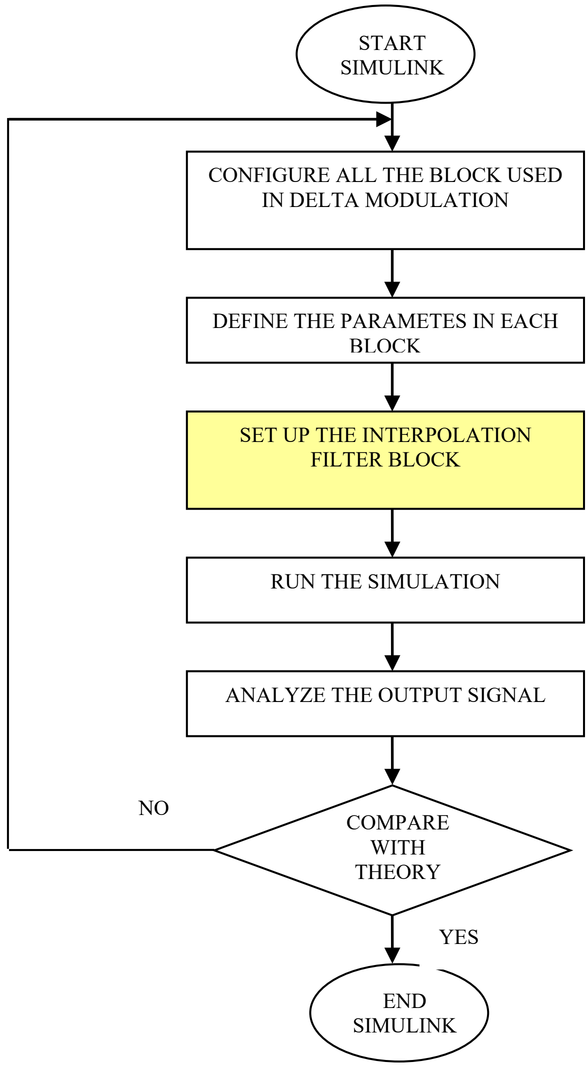

FIGURE 4.1 : Flow Chart for Project Implementation

4.2 BLOCK DIAGRAM

The main focus on this subject will be on modeling of the interpolation filter receiver in the delta modulation systems. Figure 4.2 below show delta modulation block diagram.

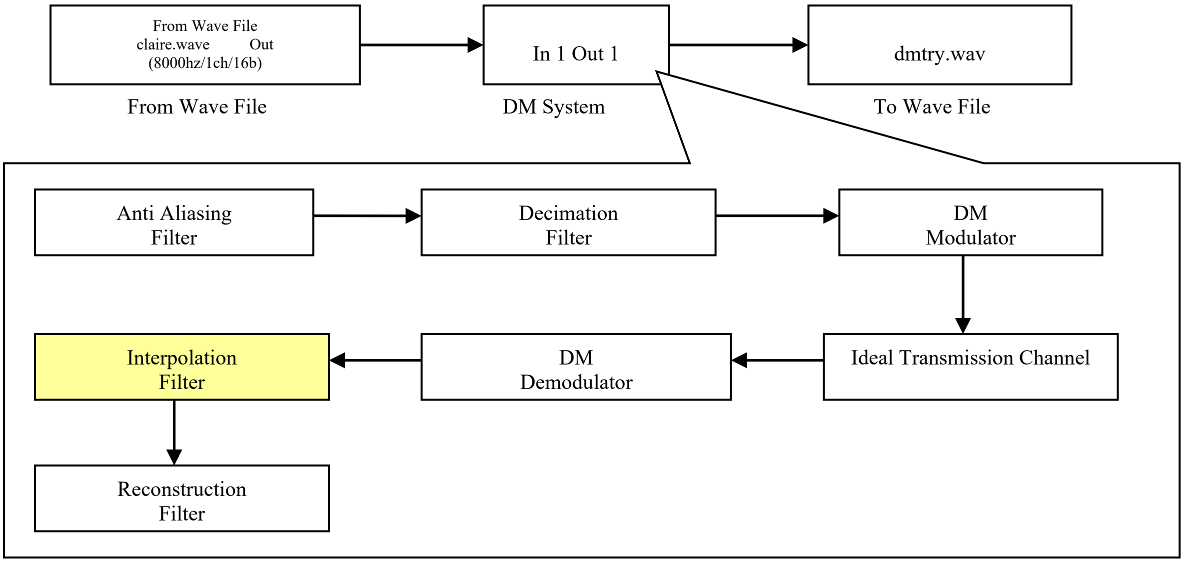

Figure 4.2 : Delta Modulation Block Diagram

From Figure 4.2, simulation accept input signal from .WAV file, then, the speech was transmitted into decimation filter. The decimation filter also known as down sampling filter. It functions to decrease the sampling rate. Next, it's transmitted to the anti aliasing filter. The main purpose of the filter is to band limited the input signal.. In DM modulator, the digital signal is converted to analog signal. The signal will transmitted through ideal transmission channel to the DM demodulator. The analog signal will be converted back to digital signal in DM demodulator.

At the next stage of the transmitting signal is an interpolation filter. The main function of interpolation filter is to increase the signal rate and filter it. The interpolation factor is simply the ratio of the input rate to the output rate. The primary reason to interpolate is to increase the sampling rate at the output of the interpolation filter so that the interpolation filter will operating at a higher sampling rate at the input signal. The interpolation filters will reconstruct signal and remove the undesired signal. As a result, it also will improve the transmitted signals. After that, the zero-order hold bock will implement the sample and holds function operating at the specified sampling rate. The block accepts one input and one output. Finally, the speech is generated into .WAV output file.

4.3 CIRCUIT DESIGN

4.3.1 BLOCK DIAGRAM OF DELTA MODULATION

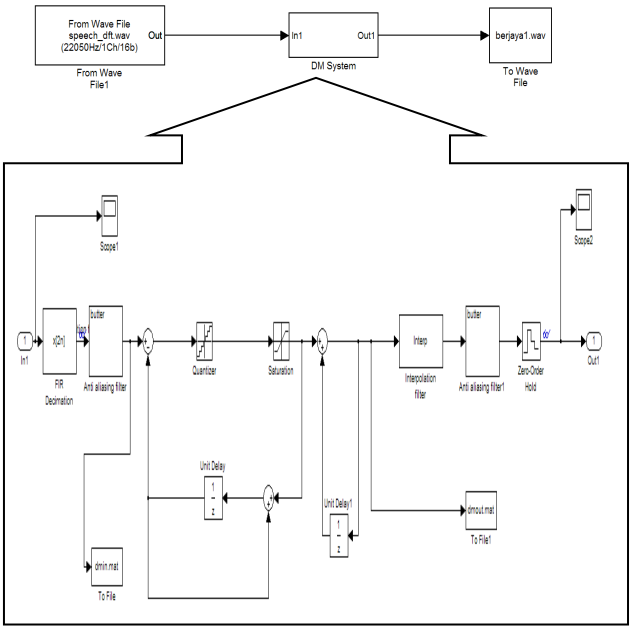

Figure 4.3 : Delta Modulation System Block

4.4 INTERPOLATION FILTER DESIGN

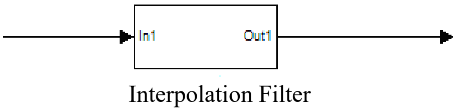

Figure 4.4 : Interpolation Filter Block

4.4.1 DESIGN A : INTERPOLATION FILTER 1

These interpolations filter design with the interpolation factor, L is equal to 2.

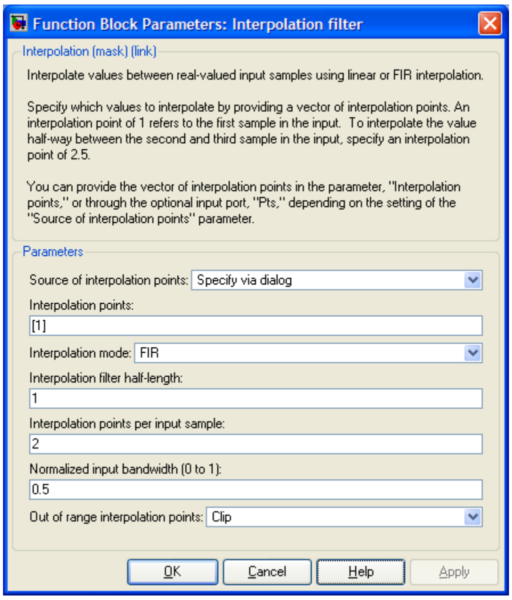

Figure 4.5 : Block Diagram of Interpolation Filter 1

4.4.2 DESIGN B : INTERPOLATION FILTER 2

These interpolations filter design with the interpolation factor, L is equal to 4.

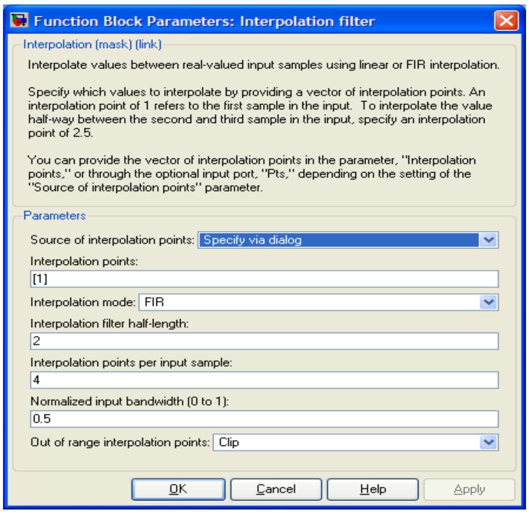

Figure 4.6 : Block Diagram of Interpolation Filter 2

4.4.3 DESIGN C : INTERPOLATION FILTER 3

These interpolations filter design with the interpolation factor, L is equal to 2.

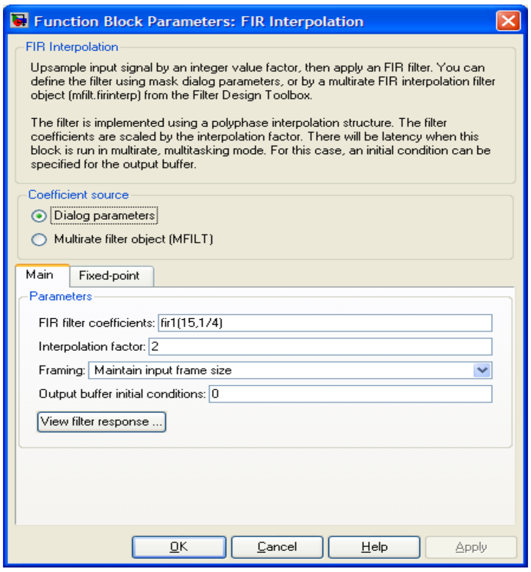

Figure 4.7 : Block Diagram of Interpolation Filter 3

4.4.4 DESIGN D : INTERPOLATION FILTER 4

These interpolations filter design with the interpolation factor, L is equal to 2.

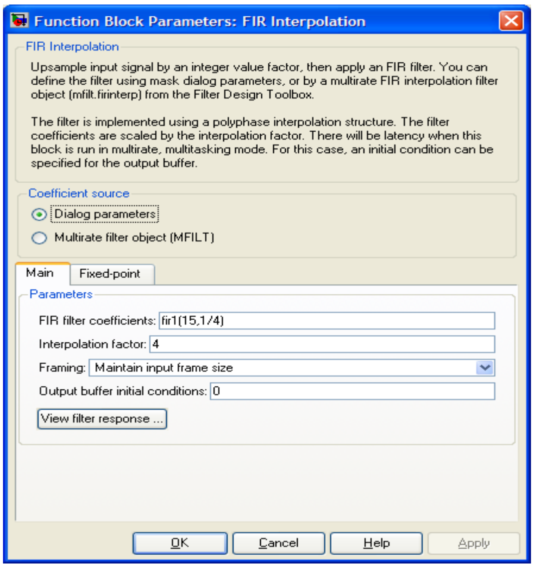

Figure 4.8 : Block Diagram of Interpolation Filter 4

4.5 SIMULATION IN SIMULINK

4.5.1 DEFINE THE PARAMETER IN EACH BLOCK

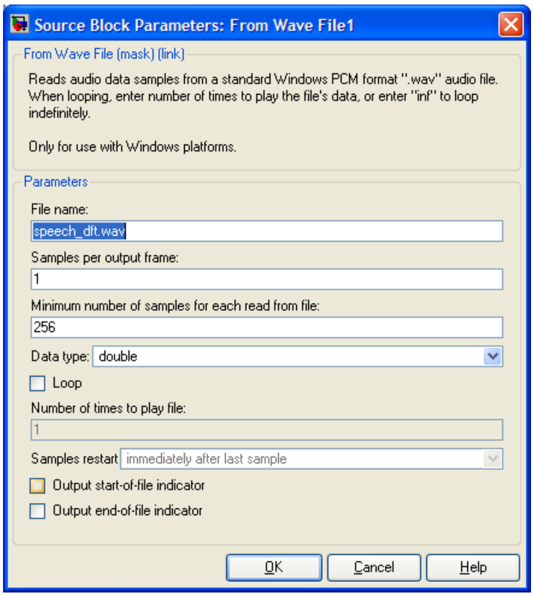

Figure 4.9 : From Wave File Block Parameter

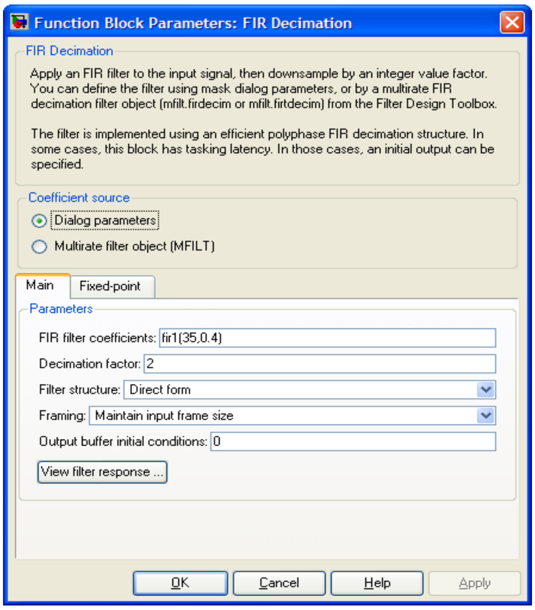

Figure 4.10 : FIR Decimation Block Parameter

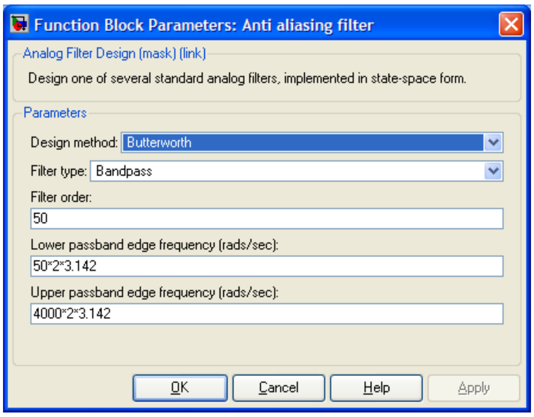

Figure 4.11 : Anti Aliasing Filter Block Parameter

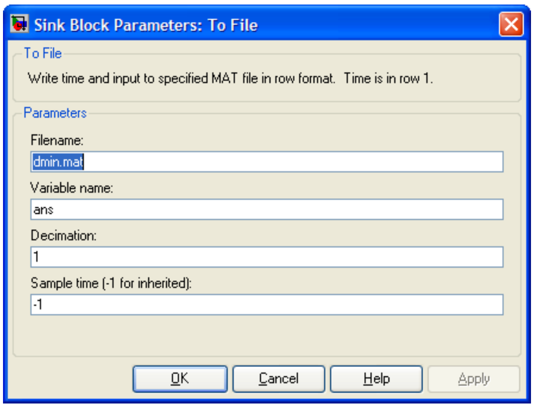

Figure 4.12 : To File Sink Block Parameter

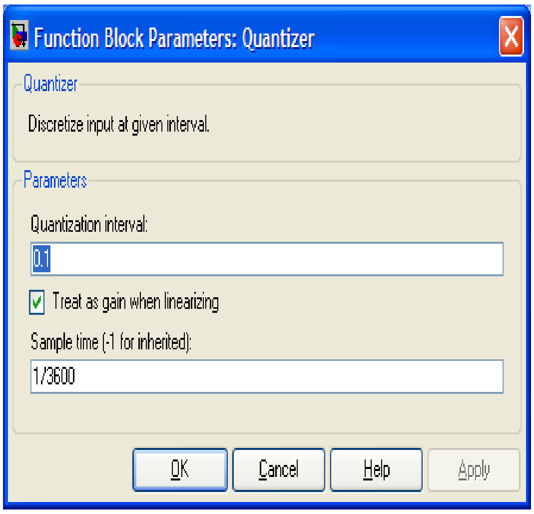

Figure 4.13 : Quantizer Block Parameter

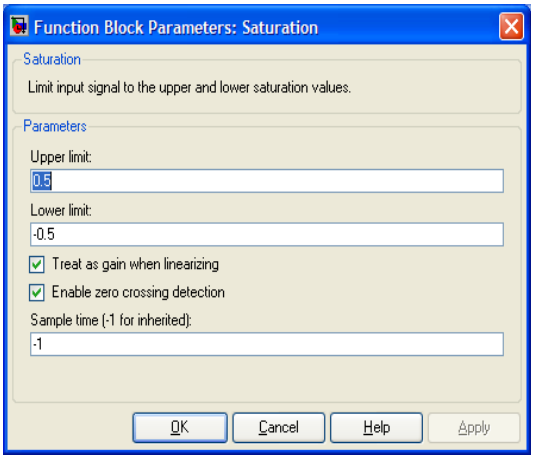

Figure 4.14 : Saturation Block Parameter

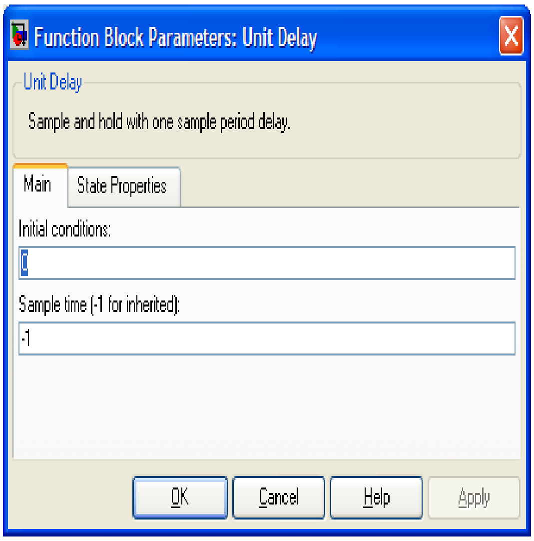

Figure 4.15 : Unit Delay Block Parameter

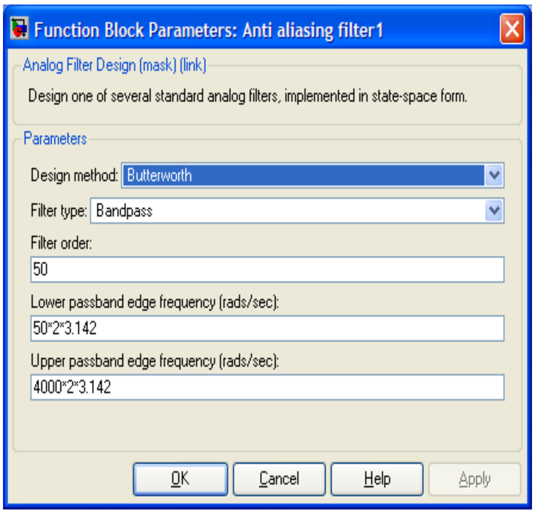

Figure 4.16 : Anti Aliasing filter Block Parameter

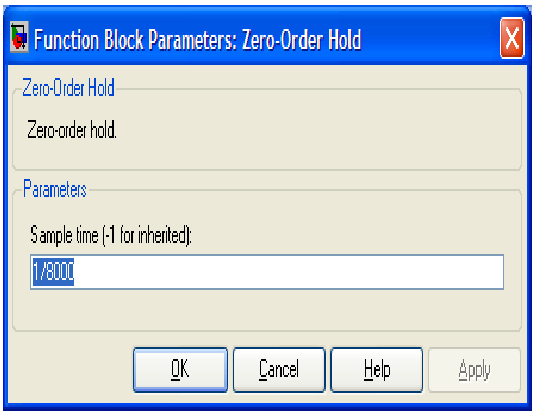

Figure 4.17 : Zero Order Hold Block Parameter

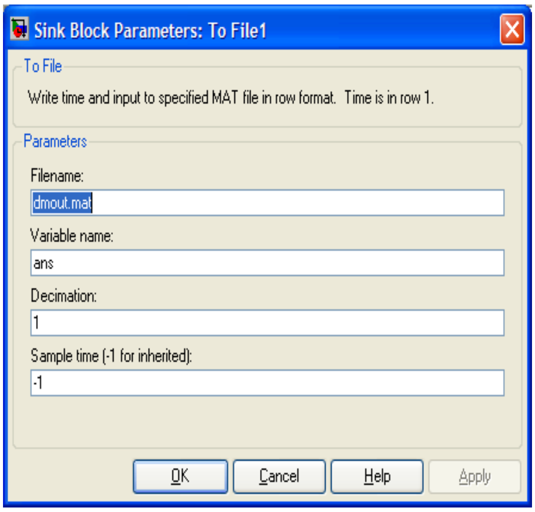

Figure 4.18 : To File 1 Block Parameter

5.0 RESULT / DISCUSSION

The result was obtained by using the SIMULINK in MATLAB environment and the waveform was display using the scope. The simulation in DM is measure at the input and the output of the system. From this, the SNR is calculated for the objective measurement. Beside that, the results also will include mean opinion score from different listeners to complete the subjective measurement.

5.1.1 WAVEFORM FOR THE INPUT DM

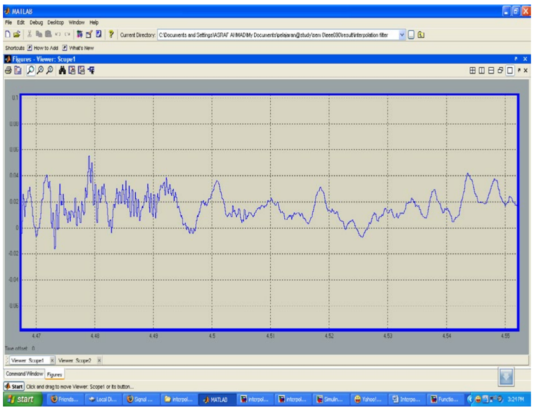

Figure 5.1 : Waveform for input DM design

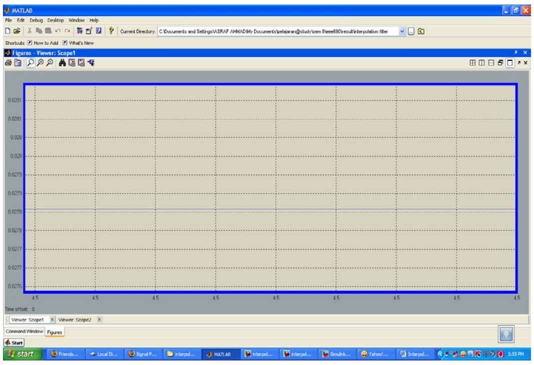

Figure 5.2 : The waveform value at point T=4.5

From the Figure 5.2, the value at point T=4.5 is 0.0278.

5.1.2 WAVEFORM FOR DESIGN A

The interpolation factor for this design is L=2.

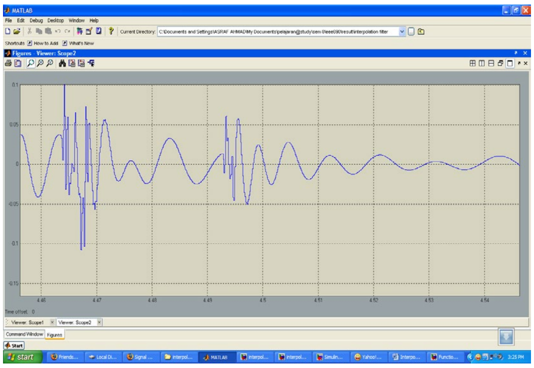

Figure 5.3 : Waveform for Design A

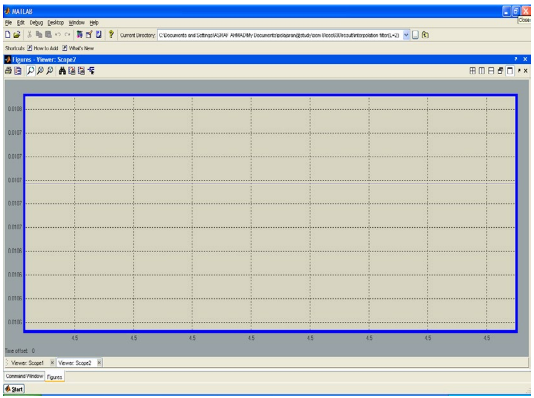

Figure 5.4 : The waveform value at point T=4.5

For Figure 5.4, the value at point T=4.5 is 0.0107.

5.1.3 WAVEFORM FOR DESIGN B

The interpolation factor for this design is L=2.

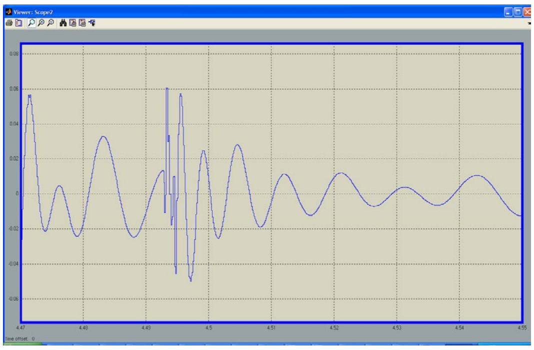

Figure 5.5 : Waveform for Design B

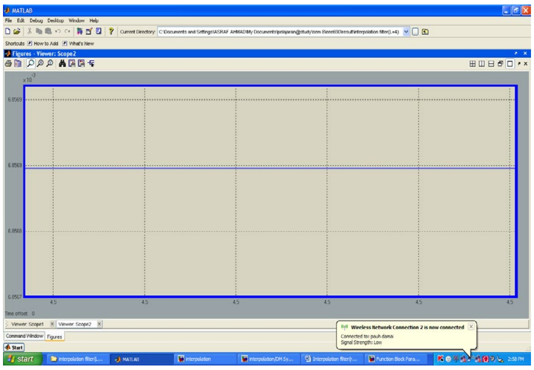

Figure 5.6 : The waveform value at point T=4.5

The value at point T=4.5 is 0.0068, as in Figure 5.6.

5.1.4 WAVEFORM FOR DESIGN C

The interpolation factor for this design is L=2.

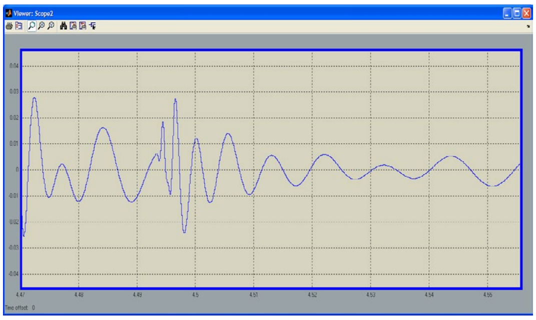

Figure 5.7 : Waveform for Design C

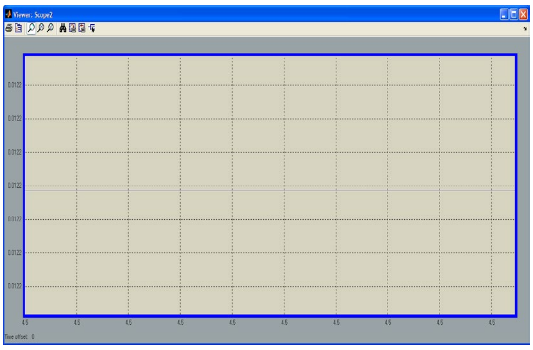

Figure 5.8 : The waveform value at point T=4.5

From the Figure 5.8, the value at point T=4.5 is 0.0122.

5.1.5 WAVEFORM FOR DESIGN D

The interpolation factor for this design is L=4.

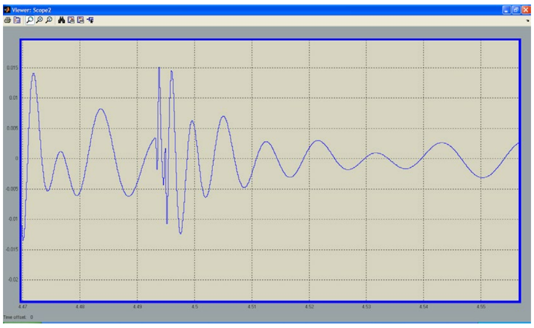

Figure 5.9 : Waveform for Design D

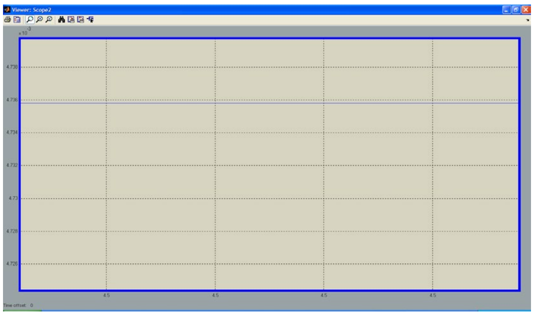

Figure 5.10 : The waveform value at point T=4.5

As the figure above, the value at point T=4.5 is 0.0047.

5.2 OBJECTIVE MEASUREMENT

Table 5.1 : SNR in dB for each design interpolation filter

DESIGN	SNR (dB)
A	4.22
B	2.44
C	5.02
D	1.61

Table 5.1 above, shows the result of SNR. From the table, Design C has the highest SNR in Delta Modulation system that was design. In general, higher signal to noise ratio is better and the signal is cleaner.

The result at the output T=4.5 are different for each design. Design C produce the best output at T=4.5 that is 0.0122. From this result, the noise signal in this design will be smaller than the Design A,B and D

The noise signal can be determined from the difference in the sample between the original signal and the reconstructed signal. From the result, the noise signals for Design A, Design B, Design C and Design D is 0.0171, 0.0210, 0.0162 and 0.0231, respectively. As a result, Design C is better than others design because

the value of noise signal is less in the transmission system.
From the noise signal value, the result for SNR value can easily determine because less value of noise signal will produce higher result in the SNR value.

5.3 SUBJECTIVE MEASUREMENT

Table 5.2: Corresponding quality for each design

DESIGN	CORRESPONDING QUALITY
A	2.31/5.00
B	2.72/5.00
C	2.84/5.00
D	2.94/5.00

The surveys for subjective measurement are taking from 32 individual listeners. From the table corresponding quality, the result show that the design D is better than Design A, B and C. The listeners rate highly on design D because from the survey, many of listeners said that the voice outputs from design D are more clearly.

Compare to the result in objective measurement, Design D is the worst value in SNR but in subjective measurement, Design D is the best. This happens because of human phenomenon, the only truly accurate judge of speech, and indeed the ultimate authority on the matter is the human ear.

6.0 CONCLUSION

As a conclusion, the objective of this project to design an interpolation filter in Delta Modulation technique for multirate digital signal processing was successfully achieved. There are 4 design of interpolation filter was propose in this project and performances of each design are compared to show the quality of the design.

There are loss signal in each design and Design C produce the lowest noise signal between other designs. The design is better because it is design using FIR interpolation filter with a small value of interpolation factor, L [2]. When the interpolation factor is small, that mean the ratio between the input and output is also small. So, the voice quality is better that other design.

FIR interpolation filter can conclude is a better filter design from simple interpolation design because it can easily designed to be linear phase. The linear-phase filters delay the input signal, but don't distort its phase. On the other hand, this FIR interpolation filter is simple to implement and also suited to multirate applications.

From the simulation result, it's clearly show that Design C for the interpolation filter has a better performance as compared to the design A,B and D in terms of SNR measurement.

7.0 FUTURE DEVELOPMENT

As a suggestion for future development, the voice that transmitted in the system can be better quality as the decimation filter at the input system is completed redesign. For this DM model, the decimation filter is simply design so that, the system is function through out the system. This is to ensure the speech quality hence reduce the bandwidth.

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