DSP IMPLEMENTATION OF CHANNEL ESTIMATION ALGORITHMS FOR OFDM SYSTEMS

A Thesis submitted in partial fulfillment of the Requirements for the degree of

Master of Technology

In

Electronics and Communication Engineering

Specialization: Communication and networks

By

N. VIJAYA RATNAM

Roll No. : 212EC5177

Department of Electronics and Communication Engineering

National Institute of Technology Rourkela

Rourkela, Odisha, 769 008, India

May 2014

DSP IMPLEMENTATION OF CHANNEL ESTIMATION ALGORITHMS FOR OFDM SYSTEMS

A Thesis submitted in partial fulfillment of the Requirements for the degree of

Master of Technology

In

Electronics and Communication Engineering

Specialization: Communication and networks

By

N.VIJAYA RATNAM

Roll No. : 212EC5177

Under the Guidance of

Prof. Sarat K. Patra

Department of Electronics and Communication Engineering

National Institute of Technology Rourkela

Rourkela, Odisha, 769 008, India

May 2014

Dedicated to…

My late grandfather

My parents and my friends

DEPT. OF ELECTRONICS AND COMMUNICATION ENGINEERING
NATIONAL INSTITUTE OF TECHNOLOGY, ROURKELA
ROURKELA – 769008, ODISHA, INDIA

Certificate

This is to certify that the work in the thesis entitled **DSP implementation of channel estimation algorithms for OFDM communication systems** by N.vijaya ratnam is a record of an original research work carried out by her during 2013 - 2014 under my supervision and guidance in partial fulfillment of the requirements for the award of the degree of Master of Technology in Electronics and Communication Engineering (Communication and networks), National Institute of Technology, Rourkela. Neither this thesis nor any part of it, to the best of my knowledge, has been submitted for any degree or diploma elsewhere.

Place: NIT Rourkela

Date: 23 May 2014

Dr. Sarat Kumar Patra

Professor

DEPT. OF ELECTRONICS AND COMMUNICATION ENGINEERING
NATIONAL INSTITUTE OF TECHNOLOGY, ROURKELA
ROURKELA – 769008, ODISHA, INDIA

Declaration

I certify that

a) The work contained in the thesis is original and has been done by myself under the general supervision of my supervisor.

b) The work has not been submitted to any other Institute for any degree or diploma.

c) I have followed the guidelines provided by the Institute in writing the thesis.

d) Whenever I have used materials (data, theoretical analysis, and text) from other sources, I have given due credit to them by citing them in the text of the thesis and giving their details in the references.

e) Whenever I have quoted written materials from other sources, I have put them under quotation marks and given due credit to the sources by citing them and giving required details in the references.

N.vijaya ratnam

23st May 2014

ACKNOWLEDGEMENTS

It is my immense pleasure to avail this opportunity to express my gratitude, regards and heartfelt respect to Prof. Sarat K. Patra, Department of Electronics and Communication Engineering, NIT Rourkela for his endless and valuable guidance prior to, during and beyond the tenure of the project work. His priceless advices have always lighted up my path whenever I have struck a dead end in my work. It has been a rewarding experience working under his supervision as he has always delivered the correct proportion of appreciation and criticism to help me excel in my field of research.

I would like to express my gratitude and respect to Prof. K. K. Mahapatra, Prof. S. Meher, Prof. P. Singh, Prof. S. Ari, Prof. S. Maiti, Prof. A.K. Sahoo, Prof. S. Hiremath, Prof. A. Swain, Prof. S. K. Das and Prof. S.K. Behera for their support, feedback and guidance throughout my M. Tech course duration. I would also like to thank all the faculty and staff of ECE department, NIT Rourkela for their support and help during the two years of my student life in the department.

I would like to make a special mention of the selfless support and guidance I received from my senior Pallab Maji and Prashanth, department of Electronics and Communication Engineering, NIT Rourkela during my project work. Also I would like to thank Saurabh Bansod and Baskar Jammu for making my hours of work in the laboratory enjoyable with their endless companionship and help as well; along with all other friends like Ragasudha,Karthik, Sandeep, Ravi, Saran Krishna, Venkatesh, Akhil, Bhargav, Naresh, and many more who made my life in NIT Rourkela a memorable experience all together.

Last but not the least; I would like to express my love, respect and gratitude to my parents, friends and my late grandfather, who have always supported me in every decision I have made, guided me in every turn of my life, believed in me and my potential and without whom I would have never been able to achieve whatsoever I could have till date.

VIJAYA RATNAM

Vijayaratam485@gmail.com

DSP IMPLEMENTATION OF CHANNEL ESTIMATION ALGORITHMS FOR OFDM SYSTEMS

ABSTRACT

Channel estimation has a lengthy and big history in OFDM communication systems. For this systems channel statistics need to track with simple techniques. Most of the OFDM channel estimation techniques also useful to MIMO-OFDM systems. Efficiency high data rate and performance of the wireless systems depends on estimation techniques. Generally present estimation techniques can be categorized into two characters. The first type is based on the pilot (known) data. Depending on pilot insertion divided into three type's Block-Type, Comb-Type Lattice –Type. Blind channel estimation techniques not require the pilot data, it requires previous channel estimated information. I.e. Selected know data portion to be enforced for the channel estimation. In this thesis, only the non-blind channel estimation techniques and stochastic gradient algorithms will be analyzed and estimated using Simulink. By increasing pilot density efficiently, we will track the channel variations, but it reduces the spectral efficiency. In wireless communication systems receiver design is an integral part in that channel estimation also plays a significant part. In general wireless communication systems know information is inserted at the sender and this know information is recovered at the receiver by using the different channel estimation techniques. In some cases it is possible to recover the transmitted information at the receiver by using differential modulation techniques and channel estimation is not necessitated in this Case but low data rate and wastage of SNR. In some other cases, the base station will complete

the estimation part and sends a pre-distorted signal but it makes system degradation. Pilot insertion is one of the main problem to estimate the channel on the receiver. Second one is tracking the channel with limited pilot data and estimation with less complexity. OFDM systems implementation and real-time estimation implementation using the Simulink is a challenging job.

The channel estimation based on the block - type, comb-type and lattice-type structure is studied. The Block type pilot structure is performed on every block of OFDM symbols and the Comb type pilot arrangement is performed on every OFDM symbol. Which are inserted particular data intervals. Comb-type and block type spectrums are verified and the bit error rate is compared. The objective of this thesis is to implementation channel estimation algorithm OFDM system implementation of the C6713 Digital Signal Processor (DSP) of Texas Instruments (TI). First, the basic channel LS and MMSE channel estimation techniques implemented and LMS, NLMS and RLS were implemented and tested using Simulink Next this model is an implementation on DSP C6713. Finally a comparison of the implemented estimator and compared BER values and mean square error estimated in LMS NLMS and RLS algorithms performed.

CONTENTS

ACKNOWLEDGEMENTS i
ABSTRACT iii
CONTENTS v
LIST OF FIGURES vii
NOMENCLATURE viii
ABBREVIATIONS ix
Chapter 1 1
DESIGN AND IMPLEMENTATION OF OFDM SYSTEM 1
1.1 Introduction 1
1.2 OFDM Implementation Using Simulink 2
1.2.1 System description 2
1.2.2 Simulink implementation 4
1.3 Advantages and Disadvantages 6
1.4 Applications 7
1.5 Motivation 8
1.6 OFDM Channel Model 8
1.7 Channel Estimation Techniques 9
1.8 Objective 10
1.9 Thesis Organization 11
Chapter 2 12
Channel Estimation in OFDM Systems 12
2.1 Pilot structures 12
2.2 Block-type 12
2.2.1 Least Square Estimator 14
2.2.2 Minimum Mean Square Estimation 15
2.3 Comb-Type 16
2.3.1 LS Estimator with 1D interpolation 17
2.3.2 Second Order Interpolation 19
2.3.3 FIR Interpolation (LPI) 19
2.3.4 Splice Cubic Interpolation (SCI) 20

2.3.5 Time Domain Interpolation (TDI) 20
2.4 Lattice-Type 21
CHAPTER 3 23
Adaptive Stochastic Gradient Algorithms Implementation 23
3.1 Introduction 23
3.2 LMS Algorithm Implementation In Simulink 24
3.2.1 Advantages 26
3.2.2 Applications 26
3.3 NLMS Algorithm In Simulink 27
3.4.1 Application 28
3.3 RLS Algorithm Using Simulink 28
Chapter 4 31
Hardware Implementation of OFDM Channel Estimation Techniques Using C6713DSK 31
4.1 Introduction 31
4.2 DSK Support Tools 32
4.2.1 Required Software and Hardware 32
4.2.2 DSK Board 32
4.4 TMS320C6713 Implementation 33
4.5 OFDM Hardware Implementation Using c6713DSK 34
4.6 Experimental setup and Simulink model 35
4.7 c6713DSK – Matlab Link Using the RTDX 36
4.7.1 Link uses RTDX 36
4.7.2 Using External Application 37
4.8 Channel Estimation Techniques, Implementation in C6713dsk 39
Chapter 5 42
Conclusion 42
5.1 Conclusions 42
5.2 Future work 42
References 43

LIST OF FIGURES

Figure 1. 1 general OFDM block diagram 4
Figure 1. 2 Simulink OFDM model 5
Figure 1. 3 Simulink OFDM spectrum 5
Figure 1. 4 Simulink OFDM constellation 6
Figure 2. 1 block -type pilot structure 13
Figure 2. 2 block LS estimation 15
Figure 2. 3 comb-type pilot structure 17
Figure 2. 4 linear interpolation 18
Figure 2. 5 FIR Interpolation 20
Figure 2. 6 Time domain interpolation 21
Figure 2. 7 Lattice-type 22
Figure 3. 1Simulink LMS algorithm model 25
Figure 3. 2 Simulink LMS channel estimation BER curve 26
Figure 3. 3 RLS performance curve 30
Figure 4. 1 TMS320C6713 board 33
Figure 4. 2 hardware setup 36
Figure 4. 3 code generation part 38
Figure 4. 4 GPD output 38
Figure 4. 5 setup diagram of interpolation channel estimation 39
Figure 4. 6 DSP bit error rate 40
Figure 4. 7 LMS, NLMS, RLS, setup diagram 41

NOMENCLATURE

Q^H	n-IFFT matrix
S_t	Coherence time
$f_{Doppler}$	Doppler frequency
K^{th}	N-subcarriers
$\bar{g}$	Channel impulse response
$\bar{n}$	AWGN channel noise
$\overline{H}$	Channel noise matrix
$\hat{H}_{mmse}$	Minimum mean square estimation matrix
σ_{max}	Maximum delay
s_f	Coherence Band Width
W^0	Weight vector
R_Y	Auto covariance matrix
R_{xy}	Cross covariance matrix
e(i)	Error matrix

ABBREVIATIONS

SISO-	:	Single input single output –orthogonal frequency division
OFDM	:	communication system
OFDM	:	Orthogonal frequency Division multiplexing
DAB	:	Digital Audio Broadcasting
IFFT	:	Inverse fast Fourier Transform
FFT	:	Fast Fourier Transform
ISI	:	Inter-symbol Interface
MIMO-	:	Multiple Input Multiple Output orthogonal Frequency Division
OFDM	:	Multiplexing
WLAN	:	Worldwide Interoperability For Microwave Access
ADC	:	Analog to Digital converter
DVB	:	Digital Video Broadcasting
DSP	:	Digital Signal Processing
BER	:	Bit Error Rate
DSK	:	Digital Signal Processor Starter Kit
GDP	:	General Purpose Display
CCS	:	Code Composer Studio
TI	:	Texas Instruments

SER	:	Symbol Error Rate
SNR	:	Signal to Noise Ratio
LMS	:	ithm
NLMS	:	Normalized Least Square Algorithm
RLS	:	Recursive least square error
GPD	:	General purpose display
RTDX	:	Real Time Data Exchange
LS	:	Least squares
MMSE	:	Minimum Mean Square Error
QAM	:	Quadrature Amplitude Modulation
USB	:	Universal Serial Bus
TDI	:	Time Domain Interpolation
SOI	:	Second Order Interpolation
LI	:	Liner Interpolation

Chapter 1

DESIGN AND IMPLEMENTATION OF OFDM SYSTEM

1.1 Introduction

Present wireless communication systems, application development based on the highest data rate communication, but it is not an easy task. It is potential to draw a high data rate by utilizing multi-carrier transmission technique which is employed in the OFDM systems. In OFDM systems transmitting total data in parallel by using a number of modulated sub-channels. The total usable bandwidth is split into a number of subcarriers and orthogonal to each other. OFDM offers a high data rate with complex receiver structure. The channel non linearity and distortions it is really hard to recover the transmitted information with simple receiver. Channel estimation algorithms has a tracking capability and estimate the channel so that we can recover the transmitted data and high data rate is possible by using the channel estimation techniques with receiver complexity. Complete OFDM introduced in 1971 before this large number of subcarrier oscillators needed because of this less efficiency and complexity increases. To get rid of the bank of oscillators to increase the efficiency of the system discrete.

The Fourier transform (DFT) is introduced in 1971 it is very easy and useful to go through the system and present FFT is used in OFDM and advantage in digital signal processing (DSP) technology made it an important component of telecommunications. In the 1990s, OFDM was exploited for wideband data communications over mobile radio FM channels, high-bit-rate digital sub-scribed lines (HDSL at 1.6 Mbps), asymmetric digital subscriber lines (ADSL up to 6Mbps), and very-high-speed digital subscriber lines (VDSL at 100 Mbps). Digital audio broadcasting started in 1987 which is a commercial use of OFDM system. In 1993 DVB is introduced and In 1995 DVB along with high-definition TV (HDTV) is started, in

1995 wireless local area network (WLAN) and Hiper LAN implementation started. OFDM has greater importance because of this, it is anticipated to turn the technology of choice in most wireless links [11].

1.2 OFDM Implementation Using Simulink

1.2.1 System description

First input binary data are taken in and performed the coding this channel coded bits are grouped together, and mapped to corresponding modulation constellation points. In this thesis 16-QAM rectangular modulation is performed (QPSK, QAM etc.). At this period, data. In frequency domain and the pilot are inserted here. A serial to parallel convertor is used. Data in frequency domain convert into time domain by using IFFT data are grouped together again, as per the number of required transmission subcarriers [5][9][8].

Let's take the data sequence $\{X(K)\}$ with a length of N by converting into the time domain $\{x(n)\}$ with the given following equation

$$x(n) = \text{IFFT}\{X(K)\} \qquad n = 0,1,2,\ldots..N-1$$

$$= \sum_{k=0}^{N-1} X(k) e^{j(2\pi kn/N)} \tag{1.1}$$

After IFFT guard time need to insert to prevent the ISI, which is larger then the delay spread. And add the cyclic prefix to the time domain signal .

The received signal is made by

$$y(n) = x(n) \otimes h(n) + w(n) \tag{1.2}$$

Where w(n) is additive white Gaussian noise h(n) is a channel impulse response

Received signal is given by Y(K) = FFT(y(n)) where k = 0,1,2,3,…..N-1.

$$= 1/N\sum_{n=0}^{N-1} y(n)e^{-j(2\pi kn/N)} \tag{1.3}$$

H(k) = FFT(h(n))

W(n) = FFT(w(n))

Estimated pilot data channel $H_P(k)$ at pilot location and the estimated data after FFT is given by

$$X_e = Y(k)/H_e(k) \tag{1.4}$$

Cyclic prefix (CP) is inserted in each block of data in keeping with the system specification and also the data multiplexed in a vary serial fashion At this time data are in time domain OFDM modulated and ready to be shipped. After transmission of OFDM signal in wireless channel. Once receive the signal at the recipient in time domain inserted CP is removed in receiver an FFT block is used to demodulate the OFDM signal. At this level, i.e. In frequency domain channel estimation is performed on complex received data pilots and complex data is dumped according to the complex constellation diagram and coded information is decoded and recover the original binary data. In this thesis we look at the precepts of an orthogonal division multiplexing (OFDM) systems. Since our aim is to investigate channel estimation methods for OFDM systems using Simulink, it is all important to acquire a firm understanding of OFDM systems in Simulink [1]. A pilot based channel estimation general block diagram show in below fig. 1. 1.

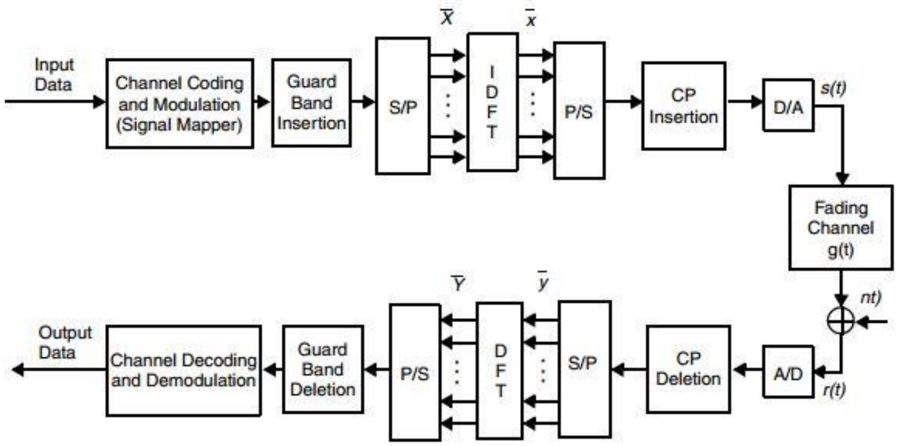

Figure 1. 1 general OFDM block diagram

1.2.2 Simulink implementation

The Simulink model of OFDM system is described in the figure below figure 1.2 using 16-QAM. The block included in this model are Bernoulli random binary integer generator, rectangular QAM modulator, multiport selector, matrix consultation for selecting the pilots, zero padded IFFT, Selector for adding a cyclic prefix, removing cyclic prefix, FFT, Remove zero padding, Remove pilots for estimation, rectangular QAM demodulator, bit error rate block, RTDX (channel output). Simulink OFDM spectrum and received constellation graphs show in figure 1.3 and figure 1.4.

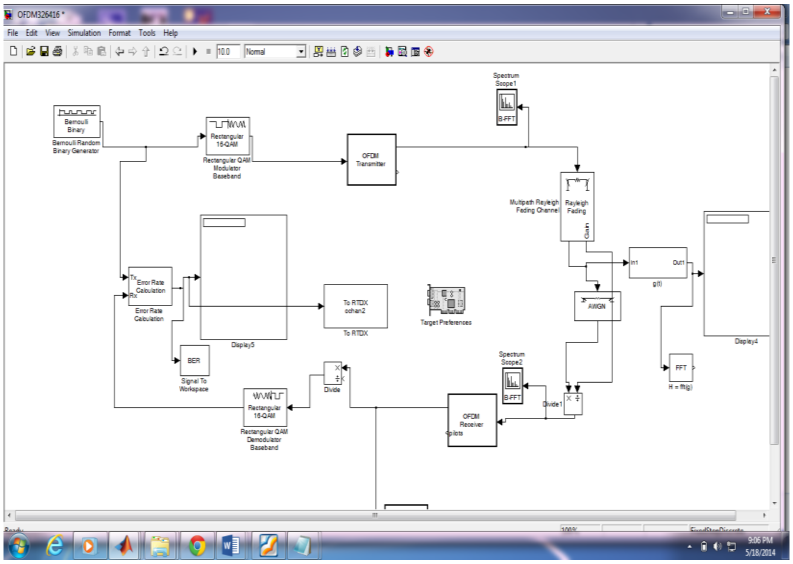

Figure 1. 2 Simulink OFDM model

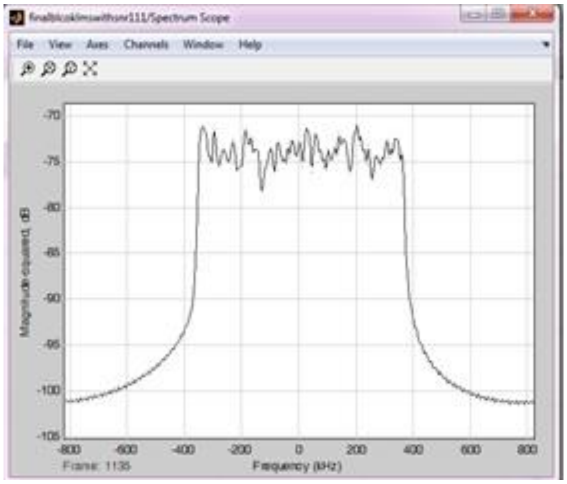

Figure 1. 3 Simulink OFDM spectrum

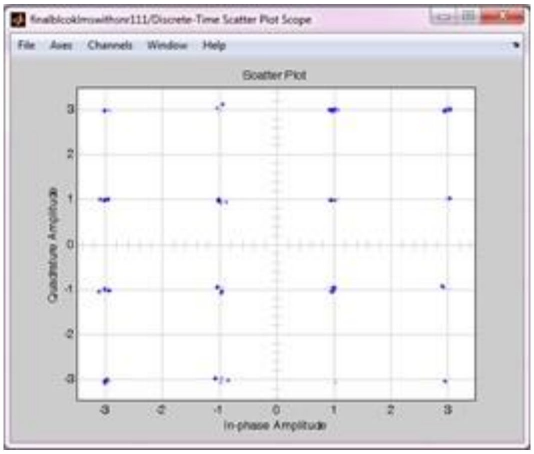

Figure 1. 4 Simulink OFDM constellation

1.3 Advantages and Disadvantages

Advantages

- Multipath distortion is vary less.
- High transmission bitrates.
- It delivers a high spectral efficiency.
- Can easily adapt to severe channel conditions without complex time-domain equalization
- Efficient implementation using FFT.
- Low sensitivity to time synchronization errors.
- Tuned sub-channel receiver filters are not essential.
- Facilitates single frequency networks.

Disadvantages

1. High synchronism accuracy
2. Complexity increase then the single carrier systems
3. Power and capacity loss due to guard interval
4. Considerable amount of bandwidth and power loss due to guard interval
5. Linear power amplifiers required
6. Multipath propagation must be avoided in other orthogonallity not be affected
7. High peak-to-average-power ratio (PAPR), requiring linear transmitter circuitry, which suffers from poor power efficiency
8. Highly Sensitive to Doppler shift.

1.4 Applications

- ADSL
- DAB
- HDTV
- HIPERLAN/2
- IEEE802.11g
- IEEE802.11a/n
- IEEE802.15.3a
- IEEE802.16d/e
- Broadband wireless access system IEEE802.16

- Wireless ATM Transmission systems
- International Mobile Telecommunications—Advanced Systems.

1.5 Motivation

To find the Performance of the any OFDM communication system. Perfect channel knowledge is required. It is not possible in real time and we require to get the channel characteristics and cognition of the channel by using channel estimation techniques depending on the classifications mainly two types of channel estimation techniques using blind type and non-blind type non –blind type channel estimation more efficient. Here we need to send the pilot information through the channel which is known to the transmitter and receiver. Hardware implementation in DSP processors more efficient than the other processors and also challenging task in DSP processors. Simulink inspires you to try things out. You can easily build models from a Simulink library and easy to edit the good examples. Simulink is a great tool to implement practical and it will solve the real time problems. Simulink is used to carry out the dynamic schemes. This is used to implement the models, easy to analysis and simulation. Simulink is used in educational foundations, research, research labs, this is employed to reduce the real time problems by using this SIMULINK library.

1.6 OFDM Channel Model

Consider an OFDM system with M subcarriers $x = [x_0, x_1 .... x_{N-1}]^T$ this are in frequency domain OFDM symbols. This symbol is converted into the time domain. By using the n-IFFT [5].

by $X = [X_0, X_1, ... X_{N-1}]^T$ $X = Q^H x$ where Q^H is a n-IFFT matrix.

$$Q_{m,n} = 1/\sqrt{N}(e^{-j2\Pi mn/N}) \qquad \text{for } m,n = 0,1,\ldots\ldots\ldots N-1. \qquad (1.5)$$

And added cyclic prefix(CP) length assume L and at the receiver received OFDM symbols remove the cyclic prefix at the receiver end is given by Y = [Y0,Y1,……………….$Y_{N-1}]^T$

$$Y = HX + Z \qquad (1.6)$$

H is a channel matrix and Z is the time-domain noise vector

1.7 Channel Estimation Techniques

Of OFDM systems or MIMO OFDM system channel estimation is an integral part. This communication system's efficiency depends on the different channel estimation techniques the main thing is here to track the channel characteristics which are applied on the receiver side. Here we are discussing only pilot added channel estimation techniques. Different type of channel estimation techniques listed below.

Block-type

Least square estimation

Minimum mean square estimation

Modified mean square estimation

Estimation with decision feed back

Estimation with decision feed back

Comb-type

Least square estimation with linear interpolation

FIR interpolation

Cubic splice interpolation

Second order interpolation

Time domain interpolation

ML estimator

PCMB estimator

Another pilot – pilot based estimators

LMS

NLMS

RLS

KALMAN

Lattice filters

2D – dimensional estimators

1.8 Objective

1. The main object of the work is to reduce the complexity of estimation algorithms and reduce the BER, increase the system efficiency without any complexity.
2. Study the channel estimation algorithms and understanding the different type of efficient channel estimation techniques

3. Try to find the new channel estimation algorithm and implemented in OFDM system and test the performance of the system and compare with the other channel estimation techniques.
4. Implement this new technique in hardware using DSP and mixed processors and try to find the efficiency and BER performance of the technique with real time.

1.9 Thesis Organization

The thesis has been organized into five chapters. This first chapter gives the introduction about the OFDM, advantages and disadvantage of OFDM and application. Different types of channel estimation techniques introduced in this first chapter and these are briefly summarized in the next following chapters. The object of the work flow and motivation of the channel estimation techniques is discussed in this chapter.

Chapter 2 The second chapter discuss the pilot based channel estimation techniques and hardware implementation hardware implementation. Introduction to the RTDX and GDP, channel estimation techniques, implemented in TMS320C6713 and compared with other techniques.

Chapter 3 The third chapter discuss about the gradient algorithms. LMS RLS AND NLSM

Its advantage and disadvantage and comparative study.

Chapter 4 The fourth chapter discuss about the channel estimation techniques, hardware implementation. Results and discussions.

Chapter 5 The fifth chapter presents the conclusion to the complete work and talks about the scope of future work to the research work that has been presented in the thesis.

Chapter 2

Channel Estimation in OFDM Systems

2.1 Pilot structures

Only restricted pilot data subcarriers are used for the primary channel estimation process. There are two ways to insert the pilots, these are in time and frequency domain. The pilot spacing is an important parameter generally this pilot are inserted in frequency domain and pilot spacing depends on the coherence frequency of the channel. Power allocation and modulation of the pilot data with regard to the factual data is an important parameter, if we increase pilot data power channel estimation accuracy increase. Generally, data, symbol power and pilot data power are equally considered. The performance of the system also depends on the pilot arrangement this pilot arrangement should efficiently track the channel variations this is the one of the important topics. Established along the pilot arrangement there are 3 forms of pilot arrangement are considered: block type, comb type, and lattice type this 3 arrangements discussed in next sections the estimation accuracy can be amended by increasing the pilot density. They present some disadvantages. Main disadvantage it decreases the spectral efficiency.

2.2 Block-type

In this block type arrangement show in below figure 2.1 group of pilot symbols are inserted in between the actual OFDM symbols with a pilot period of St With the aim of to track the channel characteristics pilot symbol period must positioned as regularly as the coherence time. The relationship between the coherence time and Doppler frequency given below equation (2.1)

$$s_t \leq 1/f_{doppler} \tag{2.1}$$

This case of pilot arrangement is not desirable for fast fading channel variation which is suited for frequency selective fading channels. That is, data burst transmission and slow channel variation this type of system is utilized [2].

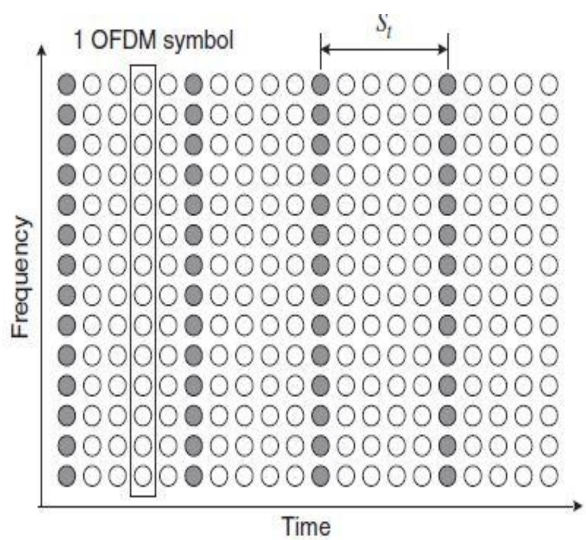

Figure 2. 1 block -type pilot structure

The block type channel estimation is supporting the LS, minimum mean square error, LMMSE. Figure 10. Below is depicted the construction of the block type pilot arrangement [1].

2.2.1 Least Square Estimator

The least square estimator is the basic technique for many channel estimation techniques pilot data symbols are used for better performance, given N pilot subcarriers assume that's orthogonal to each other. They are free of interference the pilot symbols for N subcarriers represented by diagonal matrix [2].

$$X = \begin{bmatrix} x[0] & 0 & . & . & 0 \\ 0 & & & & \\ & & & & \\ & & & & 0 \\ 0 & & & 0 & x[N-1] \end{bmatrix}$$

X (k) = pilot data at the K^{th} subcarrier where k = 0,1,2,……..N-1

Channel gain matrix H (k) at K^{th} subcarriers and the received pilot data at the receiver is given by my (k) represented as

$$Y = XH+Z \tag{2.2}$$

Channel vector H is given by H = [H (0), H (1), H (2),…… H [N-1]] ^{T} where Z is a noise vector is given by Z = [Z (0), Z (0),…..Z (N-1)]. Let $\hat{H}$ is channel estimate of H

LS Estimation of H_{lsp} (K) at each subcarrier is given by

$$H_{lsp}(K) = Y(K) / X(K) \tag{2.3}$$

To minimize the cost function H^ is given as $J(\hat{H}) = (\|Y - X\hat{H}\|)^2$.

To determine the mean square error of the H_{lsp} is given by

$$\text{MSE}_{lsp} = E\{(H-\hat{H}_{lsp})^T (H-\hat{H}_{lsp})\}. \qquad (2.4)$$

Simple least square estimation technique implemented in Simulink and Simulink BER curve by using block type show in figure 2.2 below this are helpful with low complexity mean square error is very high. This least square technique is implemented in DSP c6713DSK bit error rate is verified [2].

Least square estimation BER cure in Simulink

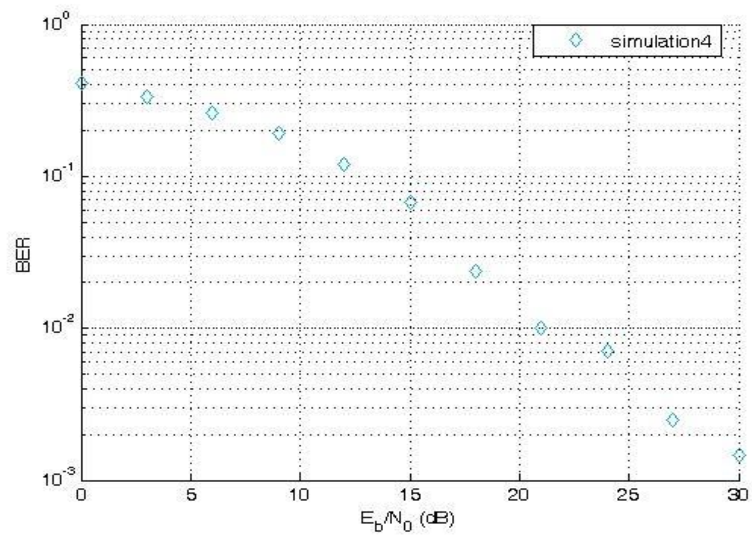

Figure 2. 2 block LS estimation

2.2.2 Minimum Mean Square Estimation

In MMSE estimation here we are calculating the second order statistics of the channel to minimize the MSE let $\bar{g} = [g_n]^T$ is sampled channel impulse response and $\bar{n} = [n_n]^T$ (0,…………N-1) is the AWGN channel noise matrix $\bar{H} = \text{fft}_N(\bar{g}) = F\bar{g}$ And $\bar{N} = F\bar{n}$ and corresponding auto covariance matrix are $R_{gg,,}R_{HH}$ and R_{YY} and corresponding cross covariance matrix is given by R_{gY}, It is derived that.

$$R_{HH} = E\{\overline{H}\ \overline{H}^H\} = F\, R_{gg} F^H \quad (2.5)$$

$$R_{gY} = E\{\overline{g}\ \overline{Y}^H\} = R_{gg,}\, F^H\, X^H \quad (2.6)$$

$$R_{YY} = E\{\overline{Y}\overline{Y}^H\} = XFR_{gg}F^H\, X^H + \sigma_N^2\, I_N \quad (2.7)$$

$\hat{g}_{mmse} = R_{gY}{}^{-1}\, Y^{HH}$ final MMSE estimation is given by Equation 8

$$\hat{H}_{mmse} = F\,\hat{g}_{mmse} \quad (2.8)$$

Minimum mean square error greater performance, and so the least square estimation, but it increases the complexity. Used in low SNR circumstances LMMSE are widely applied to cut complexity [3].

2.3 Comb-Type

For this kind of arrangement, the pilot data symbol is regularly inserted for each block of data symbols. During this phase of pilot arrangement pilot symbols are put in as frequently as coherent bandwidth in frequency axis to follow the channel characteristics [2]. This coherence bandwidth is reciprocally to maximum delay spread and the relationship between two are given below equation (2.9)

$$s_f \leq 1/\sigma_{\max} \quad (2.9)$$

Which are suitable for fast fading channels comb-type pilot structure is shown in figure 2.3. The job here is to estimate the channel conditions at the data subcarriers. In comb type channel estimation the solution includes the LS estimator with interpolation show in below sections

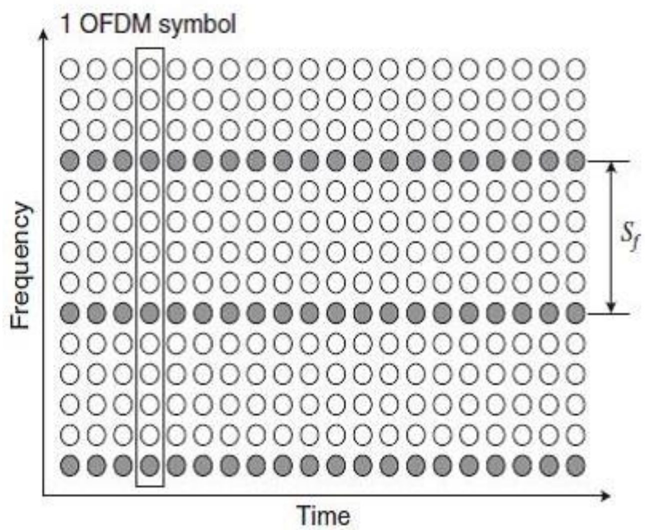

Figure 2. 3 comb-type pilot structure

2.3.1 LS Estimator with 1D interpolation

Pilot data symbols are inserted at a regularly Interval in actual block of data i. In specified location here the channel estimation is done by using the interpolating the pilot data to the nearest pilot data channel estimation is done by using the pilot interpolated data [1].

Linear interpolation (LI)

In this LI estimation first two nearest data points are interpolated this pilot interpolated data points used estimate the channel this very simple method. This linear interpolation implemented in Simulink

$$\hat{H}_d(m+l) = \left(1 - \frac{1}{L}\right)\hat{H}_p(m) + \frac{1}{L}\hat{H}_p(m+L)$$

$$0 \le l < L \tag{2.10}$$

Here the comb-type channel estimation techniques in OFDM system using Simulink show in figure 2.4. Below SNR vs BER

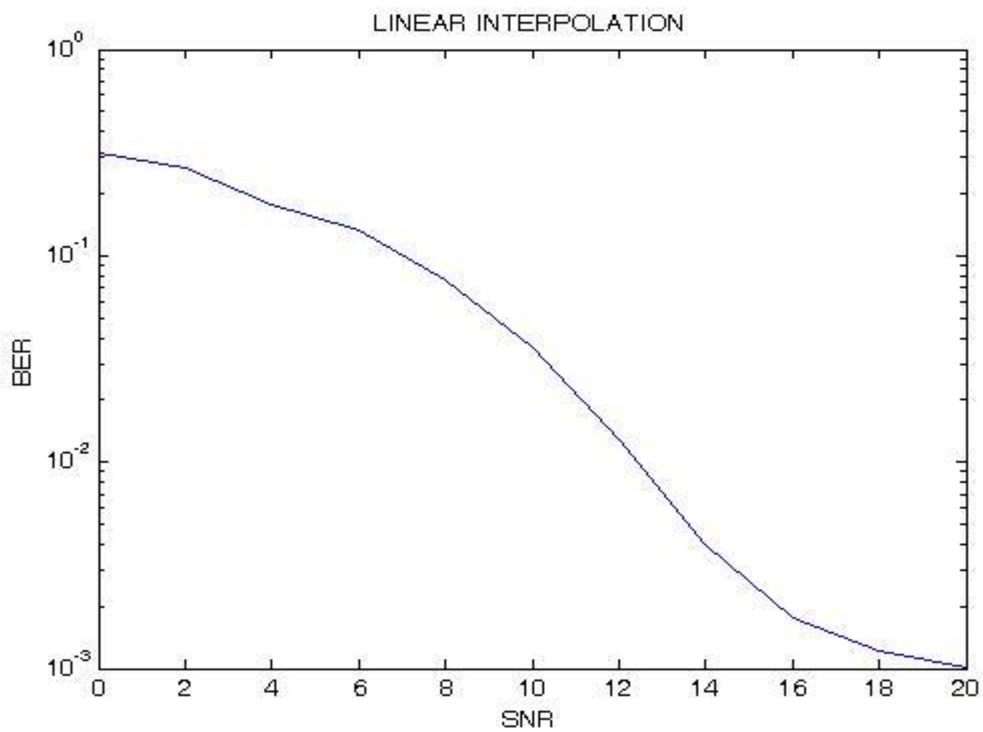

Figure 2. 4 linear interpolation

2.3.2 Second Order Interpolation

Here we used three pilot carriers to estimate the second order Interpolation. This method performs better than the least square Method Estimation based on a weighted linear combination of the three adjacent pilot's implementation second order interpolation show in below [4].

$$
\begin{aligned}
H_e(k) &= H_e(mL+l) \\
&= c_1 H_p(m-1) + c_0 H_p(m) + c_{-1} H_p(m+1)
\end{aligned}
$$

$$
\text{where} \begin{cases} c_1 = \dfrac{\alpha(\alpha-1)}{2}, \\ c_0 = -(\alpha-1)(\alpha+1), \ \alpha = \dfrac{l}{N} \\ c_{-1} = \dfrac{\alpha(\alpha+1)}{2}. \end{cases} \tag{2.11}
$$

2.3.3 FIR Interpolation (LPI)

This method first it is inserting the zeros into the given interpolation location after that it performs the interpolation mean square error is minimized with this method. Linear and FIR interpolation using Simulink BER curve show in figure 2.5 below

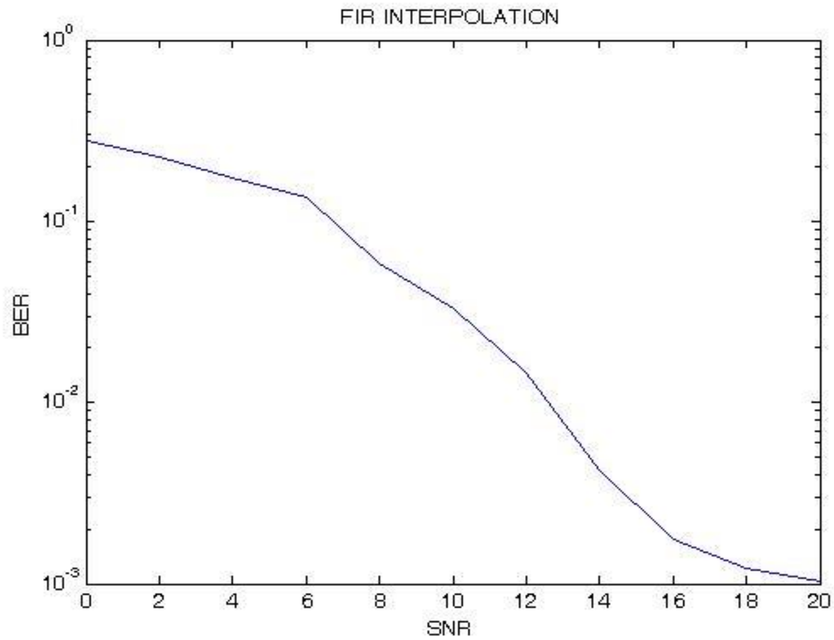

Figure 2. 5 FIR Interpolation

2.3.4 Splice Cubic Interpolation (SCI)

The method produces a smooth and continuous polynomial fitted to given data points

2.3.5 Time Domain Interpolation (TDI)

After finding the H_{lsp} first insert the zeros into original vector i.e. Zero padding and convert into frequency domain into time domain and perform the piecewise linear interpolation and converted back into frequency domain this is practiced in middle and low signal to noise ratio environment[1].

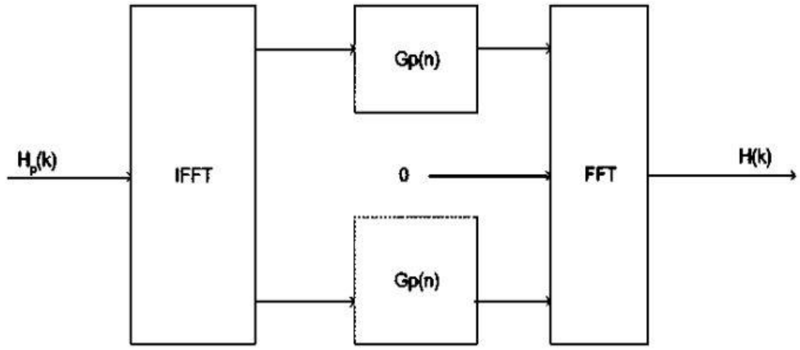

Figure 2. 6 Time domain interpolation

In this channel estimation technique first we need to take the least square estimation on the received pilot data and perform the 1-D interpolation send this data on to the actual data. Here in this type channel estimation technique 2 simple techniques implemented using Simulink and hardware implementation done with the C6713DSK DSP kit.

2.4 Lattice-Type

This is also called as two dimensional pilot arrangement pilot data are introduced both the time and frequency axes within certain periods. In this type of pilot arrangement pilot symbols are put in as frequently as both time and frequency axis to track the channel characteristics. This arrangement must satisfy the given equation 3 and pilot arrangement show in figure 2.7

$$s_t \leq 1/f_{doppler} \text{ and } s_f \leq 1/\sigma_{\max} \tag{2.13}$$

In the above equation shown in St, SF means the periods of pilot symbols in both axis correspondingly [1] [9]. 2D estimators structure from a huge computational complexity, channel estimation based on the 2D least squares (LS) and 2D normalized least squares (NLS) are suggested [1] [7].

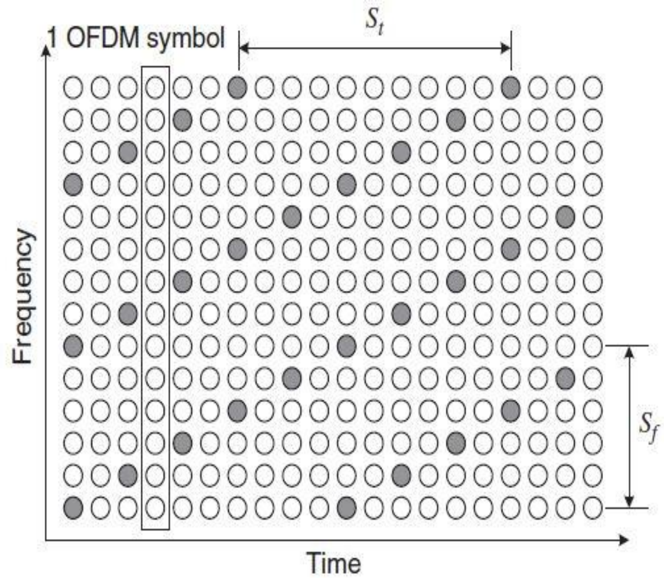

Figure 2. 7 Lattice-type

CHAPTER 3

Adaptive Stochastic Gradient Algorithms Implementation

3.1 Introduction

In previous chapter3 discussed about the mean square channel estimator's one of the greatest difficulties is that essential to estimate the covariance and cross covariance matrix of the duct, which are not much available in nature, such a situation I need to cover the signal statistics and need to calculate the signal statistics. Gradient algorithms abilities for learning and tracking are the principal causes behind the broad role of this method in this chapter4, this learning mechanism and tracking mechanism is possible to achieve by iterative schemes. One of the methods is steepest-descent methods and fundamental of most adaptive filtering techniques. The steepest - descent algorithm is given consider random vector u with $R_u = Eu^*u > 0$ and random variable d with variance σ^2 and w^0 is a weight vector [3].

$W^{0\,=}$ this is the solution to the least square estimation problem

$$\min_{w} E|d - uw|^2 \tag{3.1}$$

And recursively as follows begin with starting guess w_{-1} is generally 0 for our channel estimation case and any positive step size is very small generally we considered and iterations i≥0.

$$w_i = w_{i-1} + \mu[R_{du} - R_u w_{i-1}] \tag{3.2}$$

$$w_i \rightarrow w^0 \quad \text{and} \quad i \rightarrow \infty$$

3.2 LMS Algorithm Implementation In Simulink

Consider a random variables d with realizations {d (0), d (1),……..}, And a random vector u with realizations {u1, u2,………} the weight vector where u is a row vector w^o is given by

$$\min_{w} E|d - uw|^2$$

$$w_i = w_{i-1} + \mu u_i^* \left[d(i) - u_i w_{i-1}\right], \quad i \geq 0, \quad w_{-1} = \text{initial guess}$$

(3.4)

Generally, initial guess is zero and μ is very small which a positive value is

$$w^0 = R_u^{-1} R_{du} \tag{3.5}$$

The error is then used to regulate the filter coefficients form w_{i-1} to w_i in keeping with. The error signal can assume little values. The error signal is given as the difference between actual signal and the desired signal [2].

$$\text{M.m.s.e} = \sigma^2 - R_{ud} R_u^{-1} R_{du} \tag{3.6}$$

The error is then used to adjust the filter coefficients from w_{i-1} to w_i according to In steady-state, the error signal will assume small values

LMS channel estimation in OFDM system using the Simulink show figure below. Received pilot data are given to the LMS filter input and transmitter pilot data is given to LMS filter desired signal. The step size is very small value selected. Let's consider the $\{d_i, u_i\}$, are the input and output of the OFDM system.

At every time instant I, the measured output of the channel, d (I), is compared with the yield of the adaptive filter, =, and an error signal, we (I) = d (I) -, is generated.or signal, e(i) = d(i)- $u_i w_{i-1}$, is generated. The error is then utilized to correct the filter coefficients.

In case of computational cost LMS need 2M real additions and 2M+1 real multiplications per iteration is required for real valued data, for complex valued data 8M real addictions and 8M+2 real multiplication require . Weight adjustment is an important, weight adjustment is as minimum as possible get the desire signal output of adaptive filters. Adaptive filter efficiency depends on the several iteration this iteration as many as possible generally considered. LMS algorithm can work stationary or non-stationary. In Simulink, it is very simple to make an LMS channel estimation model using Simulink show in figure 3.1 below. Here step size is taken as 0.1.figure 3.2 shows the BER curve.

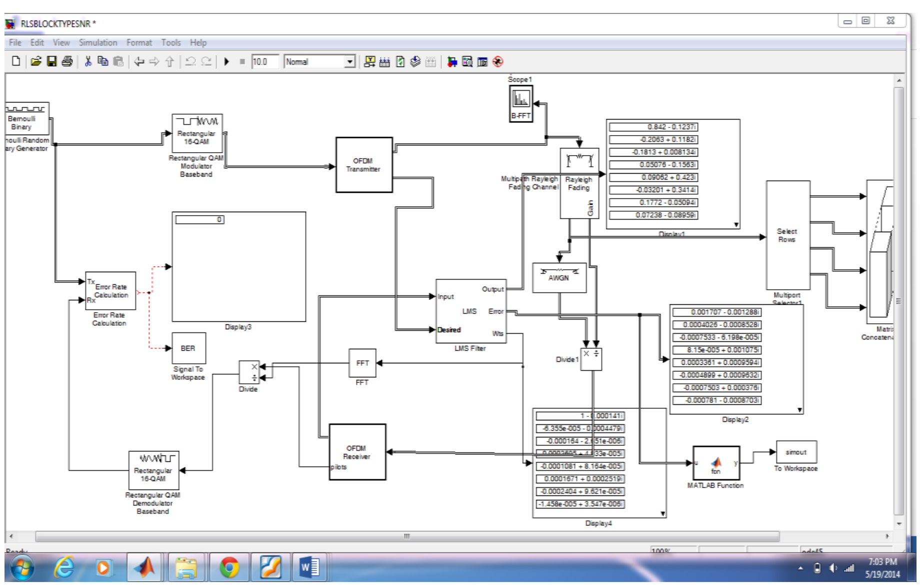

Figure 3. 1Simulink LMS algorithm model

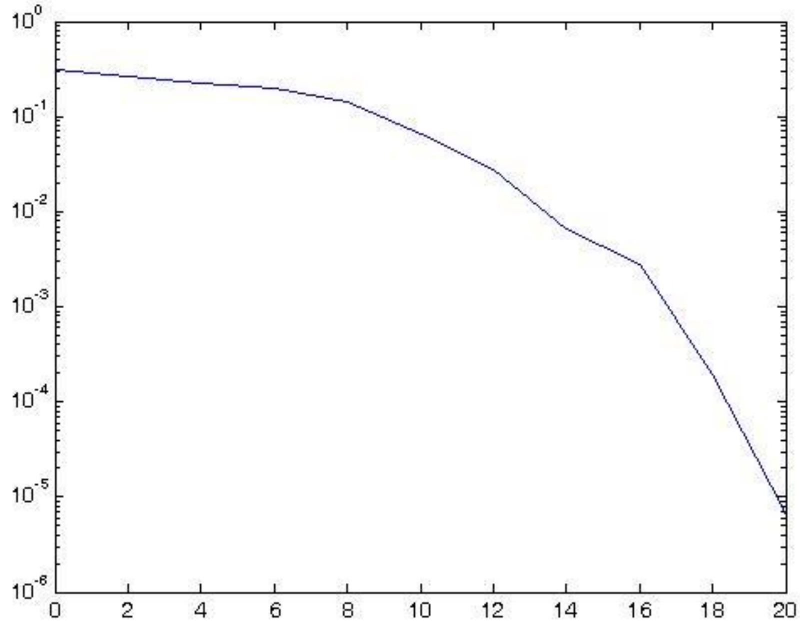

Figure 3. 2 Simulink LMS channel estimation BER curve

3.2.1 Advantages

1. Simplicity
2. Robustness
3. Low computational complexity

3.2.2 Applications

- ✓ Adaptive channel estimation
- ✓ Adaptive channel equalization
- ✓ Decision feedback equalization

3.3 NLMS Algorithm In Simulink

Consider a random variables d with realizations {d(0),d(1),,……..}, And a random vector u with realizations {u1, u2,………} the weight vector where u is a row vector w^o is given by

$$\min_{w} E|d - uw|^2 \tag{3.7}$$

Approximated NLSM is given by

$$w_i = w_{i-1} + u / \varepsilon + \|u_i\|^2 u_i^{*}[d(i) - u_i w_{i-1}], i \geq 0, w_{-1} = initial, guess(0) \tag{3.8}$$

ε is a small parameter (positive)

the newton's regularization ε(i) an μ(i) are constant

$$w_i = w_{i-1} + \mu[\varepsilon I + R_u]^{-1}[R_{du} - R_u w_{i-1}] \tag{3.9}$$

NLMS recursion

$$wi = w_{i-1} + \mu / \varepsilon + \|u_i\|^2 u_i^{*}[d(i) - u_i w_{i-1}], i \geq 0, \tag{3.10}$$

In case of the NLMS computational cost it requires the 3M real additions, 3M+1 real multiplications and one real division require in our algorithm case complex valued case for this 10M real addition, 10M+2 real multiplications and one real division [3].

NLMS has a computational complexity, higher than the LMS, but the advantage is that converges faster than LMS. In normal case LMS creates a problem LMS require the large information (pilot) to estimate the channel. μ is normalized in such a way that energy of data vector.

3.4.1 Application

1. Echo cancellation and noise reduction

NLMS OFDM channel estimation using Simulink and it's SNR vs BER curve show in the figure 3.2 same as the LMS algorithm with the varying step-size.

3.3 RLS Algorithm Using Simulink

RLS adaptive filter is an algorithm which recursively finds the filter coefficients that minimize least squares cost function RLS has extremely fast convergence but computational complexity increases and good tracking performance [3].

Consider a random variables d with realizations {d(0),d(1),,……..}, And a random vector u with realizations {u1, u2,………} the weight vector where u is a row vector w^o is given by

$$\min_{w} E|d - uw|^2$$

Regularization newton's method

$$w_i = w_{i-1} + \mu(i)[\varepsilon(i)I + Ru]^{-1}[Rdu - R_u w_{i-1}] \quad (3.11)$$

Replace $$R_{du} - R_u w_{i-1} \quad (3.12)$$

Approximation

$$ui * [d_i - u_i w_{i-1}] \quad (3.13)$$

For better estimation R_u

$$\hat{R}_u = 1/i+1\sum_{j=0}^{i}\lambda^{i-j}u_j^*u_j \tag{3.14}$$

$\lambda = 1$

And so the above equation becomes

$$\hat{R}_u = 1/i+1\sum_{j=0}^{i}u_j^*u_j \tag{3.15}$$

u_i most recent regression is require to update the p_{i-1} to p_i

Iteratively through recursion

$$P_i = \lambda^{-1}\left[P_{i-1} - \frac{\lambda^{-1}P_{i-1}u_i^*u_iP_{i-1}}{1+\lambda^{-1}u_iP_{i-1}u_i^*}\right]$$
$$w_i = w_{i-1} + P_i\, u_i^*[d(i) - u_iw_{i-1}], \quad i \geq 0$$
$$P_{-1} = \epsilon^{-1}\mathrm{I} \text{ and where } 0 \ll \lambda \leq 1. \tag{3.16}$$

The RLS algorithm is associate order of magnitude costlier than LMS-type algorithms, requiring O(M2) vs.O(M) operations per iteration. However, RLS converges considerably quicker than LMS t's familiar that the RLS filter converges quicker than the LMS filter generally, however, that if you are following time varied parameters the LMS algorithm perform higher. LMS filter is sort of a purpose estimate, however the RLS uses additional data. RLS has fast convergence compare to other algorithms and full tracking capability. When it comes to RLS channel with $\varepsilon = 0.995$ and variance = 1 the algorithm producing high computational complexity and corresponding response curve is drawn between BER and SNR is as shown in the figure [3.3].

The performance of the RLS algorithm depending on the parameter ε for number of observations and number of iterations in RLS the mean value keep on decreasing and become the constant theoretically [3].

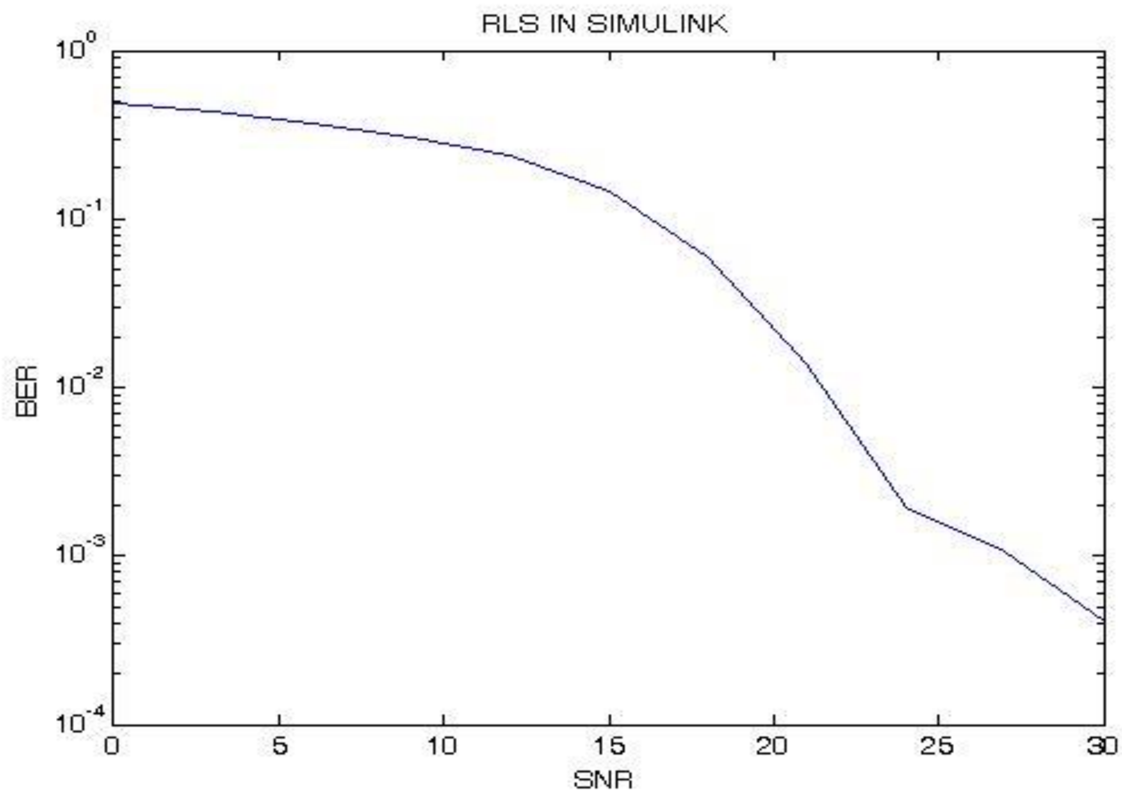

Figure 3. 3 RLS performance curve

Chapter 4

Hardware Implementation of OFDM Channel Estimation Techniques Using C6713DSK

4.1 Introduction

Real time digital signal processing give a guaranteed of delivery of data in certain time keep pace with some external events in non-real time case does not have the such time constraints.DSP have cost effective compare to the compare to the other processor depending on the application.DSP are less effected by the environmental changes this are easy to use and flexible general DSP's are more useful for audio signals which is 0 to 90khz general DSP consist of ADC which is used to capture the input signal this signal processed and converted to DAC converter Processing of analog signals can be done in either in analog or digital domain. In analog communication response to various physical phenomena in analog manner i.e. in continuous time and amplitude.in order to convert the analog signal to digital signal digitizing is used perform via an analog to digital converter. The reason behind the using of DSP'S are those are used to allow the programmability DSP C6713DSK used many application by changing the simple code.DSP'S are more stable and tolerant output then the analog communication system. Present world DSP's play major role in the 3G wireless, cable modems, and DSL modems. The processing digital signals can be implemented in different domains like DSP's and VLSI circuit, microprocessor [13].

4.2 DSK Support Tools

4.2.1 Required Software and Hardware

DSK6713 starter kit is need for code generation and to test the data at input and output are function generator, microphone, cables with audio jacks and oscilloscope USB cable to connect the DSP to PC's. Software is needed to execute the generated code called code composer studio (CCS). In this CCS application consist of compiler, assembler, linker simulator, and debugger utilities [13].

4.2.2 DSK Board

16MB-SDRM and 256KB flash memory Input and output MIC IN, LINE IN are input ports MICOUT, LINEOUT output ports The DSK features the TMS320C6713 DSP, a 225 MHz device delivering up to 1800 million instructions per second c6713 internal memory of 256KB c6713 perform the both floating point and fixed point operations, JTAG interface through USB figure 4.1 below shows the TMS320C6713 starter kit and it's applications [13].

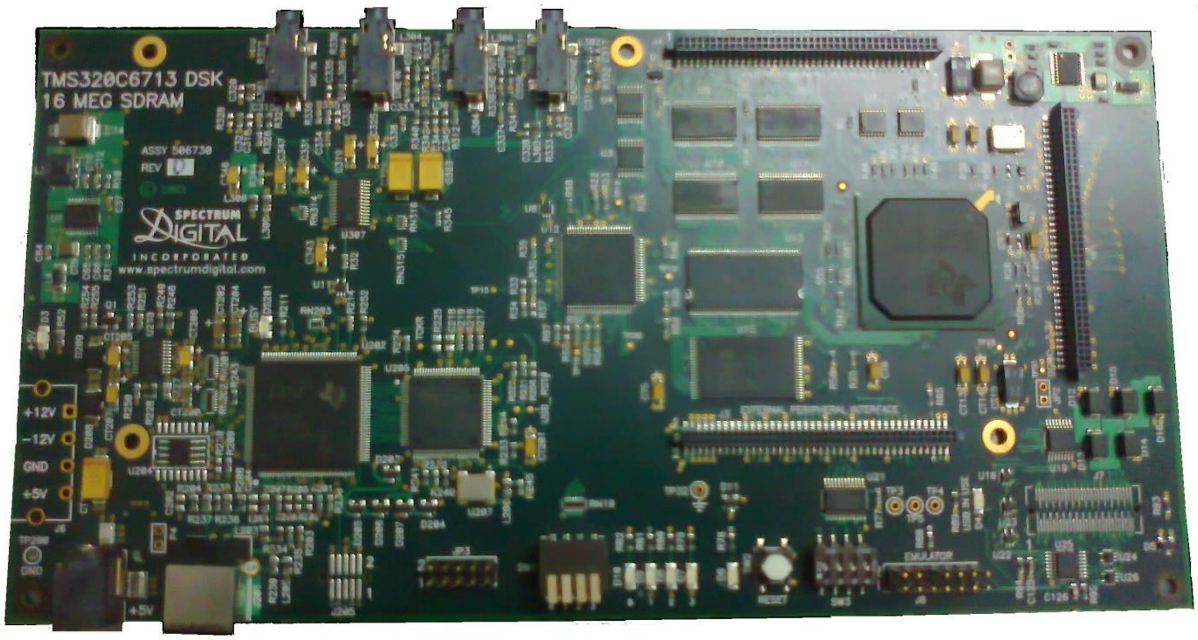

Figure 4. 1 TMS320C6713 board

4.3 Applications

- Communication
- Image processing
- Controls to speech
- Networking
- Instrumentation

4.4 TMS320C6713 Implementation

Matlab 2007a and CCS 3.3 are the best combination to implement DSP RTDX and using GDP. In Matlab2007a and CCS 3.3 and older versions there is a direct link to generate the code and download into CCS but earlier versions we need to link with latest CCS versions by using "xmakefilesetup", this is a lengthy procedure and we need to specify the hardware target in latest

CCS versions. CCS3 and below versions do not support for windows7 users better to exit switch to CCS4/CCS5 with MATLAB 2011a are the best shapes [13].

DSP's are super-efficient compared to the PC'S executing the signal processing algorithms, hence they are of great use in the industry when it adds up to the simple display PC'S much better than DSP'S For example, you want to do any complex model it will execute 10 times quicker than the PC'S but doesn't supply us any form of visual display unit but there is even a chance of making the output using RTDX and GDP. Information exchange between the DSP and computer two way communication channel is methodology this setup is implemented in this thesis [13].

4.5 OFDM Hardware Implementation Using c6713DSK

As shown in the figure 1.2 chapter1 OFDM implementation using Simulink models with hardware implementation setting and running model show. This OFDM transceiver model with 16-QAM rectangular modulation, 56 symbols and 8 pilot carriers are in frequency domain. 64 – point IFFT convert the frequency domain to time domain last 16 bits of data is added to the beginning of the data to make the cyclic prefix and serial to parallel conversion is done. At the receiver remove the cyclic prefix and perform the FFT with same 64-point and pilot data is separated here to estimate the channel, data symbols are sent to the error rate calculatorfigure1. 2 chapter 1. Select the target preference and select the settings accordingly. Select the "to RTDX " to produce the desired response connects this block to get DSP output RTDX is generally applied to transpose the data between the aim to the Matlab by using JTAG emulator, my job is to count on the BER. RTDX is connect to error rate calculator and build the model by changing the solver settings accordingly. Plug in the PC to the c6713DSK using USB cable, open the Matlab and to determine the CCS is installed properly or not for this in Matlab command window type "ccsboardinfo" it returns information "board type " and "processor name". To follow out any model proper hardware configuration selection is very important, This model implemented with windows Xp Matlab7 and CCS3.3 this is the best combination another one combination is windows 7, Matlab2011a with CCS4/CCS5.Ones done all these settings you real-time workshop converts Simulink models into ASIC C/C++ code that can compile using CCS. Link for CCS is used to invoke the code building process from within CCS to build an executable. This code can then be downloaded to the target DSP from where it flows. The data on the target is accessible in CCS or in Matlab via Link for CCS or via Real-Time Data Transfer (RTDX). This thesis primarily uses RTDX for accessing information on the target DSP. Two ways to control the DSP output discussed in below section with OFDM implementation [13].

4.6 Experimental setup and Simulink model

The below figure 4.2 indicate that the experimental setup and Simulink model of OFDM system with channel estimation techniques. The hardware setup required the PC's and c6713DSK KIT and connecting wires. Power on the DSP kit with the adapter cable, connect the PC's to the TMS320C6713 hardware kit with the USB cable. Go over the connector on the PC. And check also in MATLAB by using the function "ccboardinfo" once you connect the hardware go to Simulink and run the model select the configurations according to the application and build the model and generate the code to target hardware it will automatically generate the code and automatically. You can free to see the code in CCS and see the output by using RTDX [12].

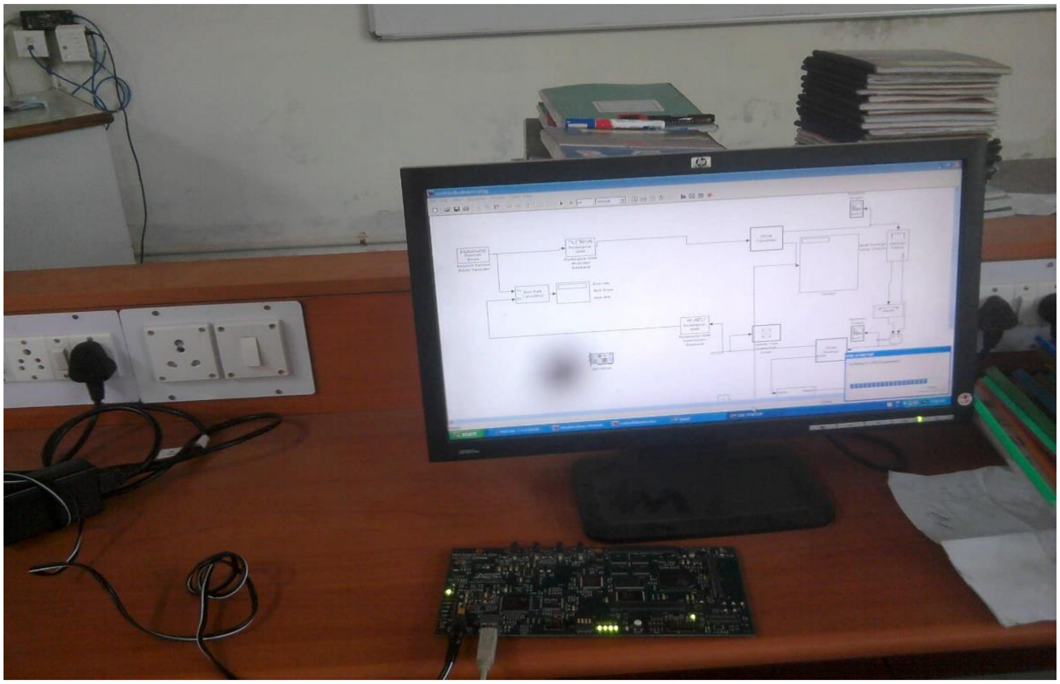

Figure 4. 2 hardware setup

4.7 c6713DSK – Matlab Link Using the RTDX

Two ways to exhibit the output one way is by utilizing the DSP MATLAB code and another one is using GDP here the simple program for open the RTDX given below. Two types RTDX channel is available in Simulink one is ‘to RTDX’ and ‘from RTDX’ one is used for sending/write from PC’s to DSP’S and another channel is used receive/read the data from the DSP. Here I used ‘to RTDX’ to get the DSP output.

4.7.1 Link uses RTDX

```
cc = ccsdsp;

open(cc.rtdx,'ochan2','r')

%od = zeros(1.1000);

for k = 1:10000
```

```
%od(1:end-1) = od(2:end);

%od(end) = readmsg(cc.rtdx,'ochn1','double',1)

od = readmsg(cc.rtdx,'ochan2','double',1);

plot(od);

drawnow;

end
```

4.7.2 Using External Application

This application is used to get the numerical results using little window show in figure 4.4 below with DSP output. GPD used to add the channel to provide interface this application available in the CCS Folder itself here tested results show in figure 4.3. Below with OFDM resultant received data bits and errors in DSP6713. And also code generation part is also shown in figure 4. 3 Below.

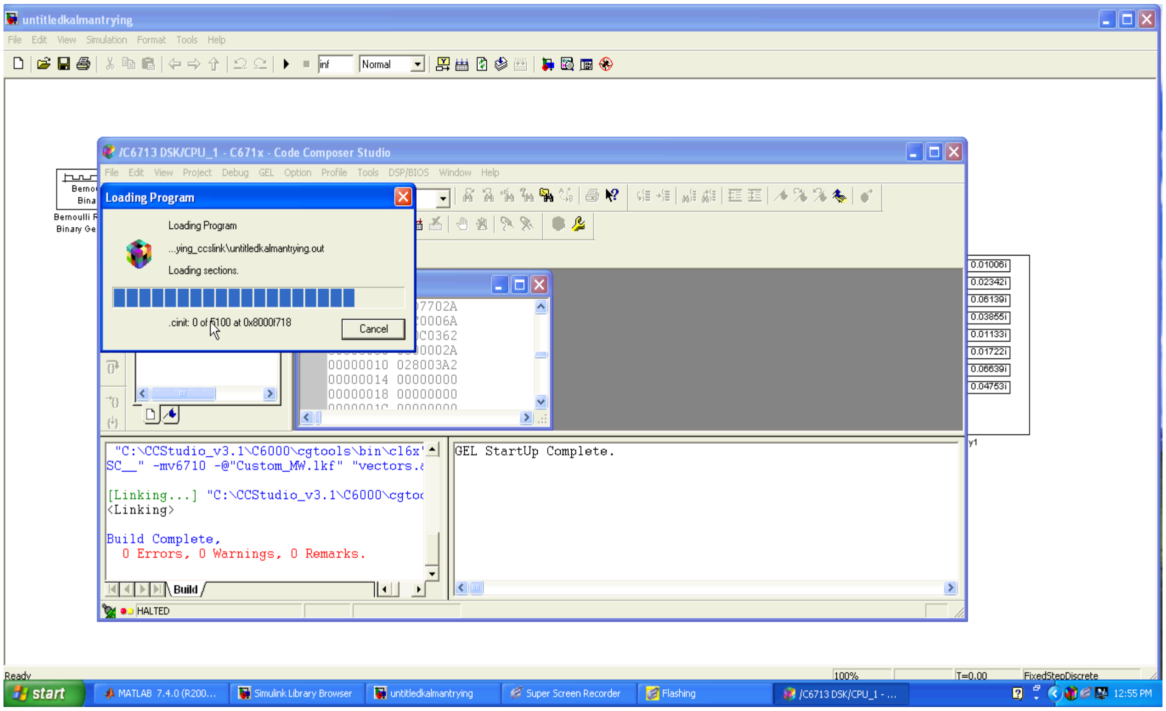

Figure 4. 3 code generation part

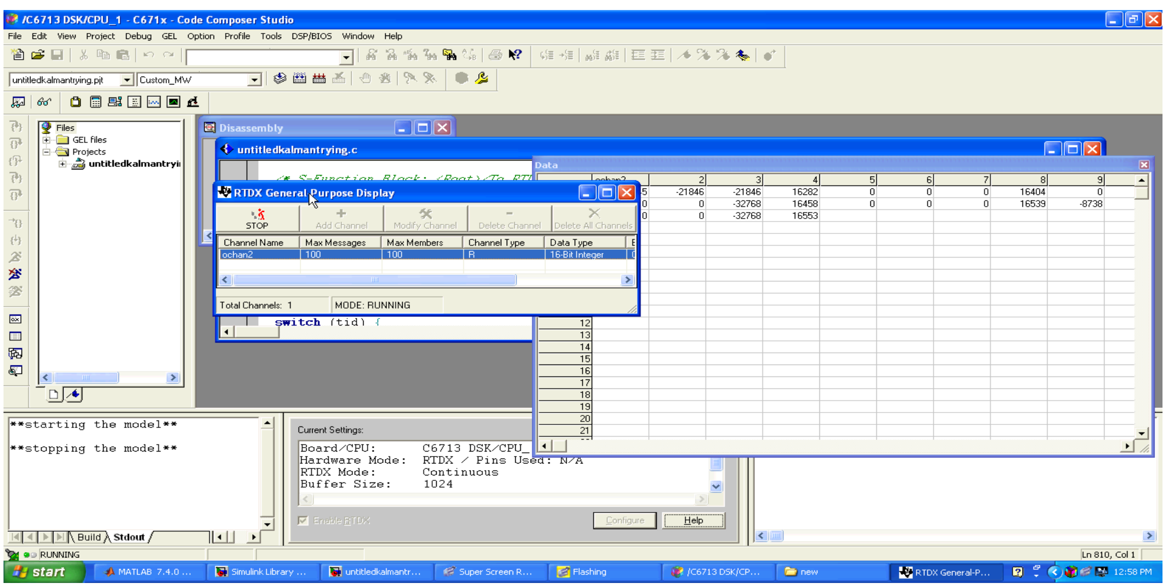

Figure 4. 4 GPD output

4.8 Channel Estimation Techniques, Implementation in C6713dsk

In this thesis block-type and come-type channel estimation techniques are discussed. Basic and fundamental channel estimation technique least square estimation using block-type and comb-type are discussed and implemented in c6713DSK with the help of RTDX and GDP bit error rate is computed. Implemented and efficiency of the system estimated. Block type channel estimation techniques are complexity increases and these are suitable for slow type channel variations. Below figure 4.5. DSP hardware implementation and it' bit error rate curve [13].

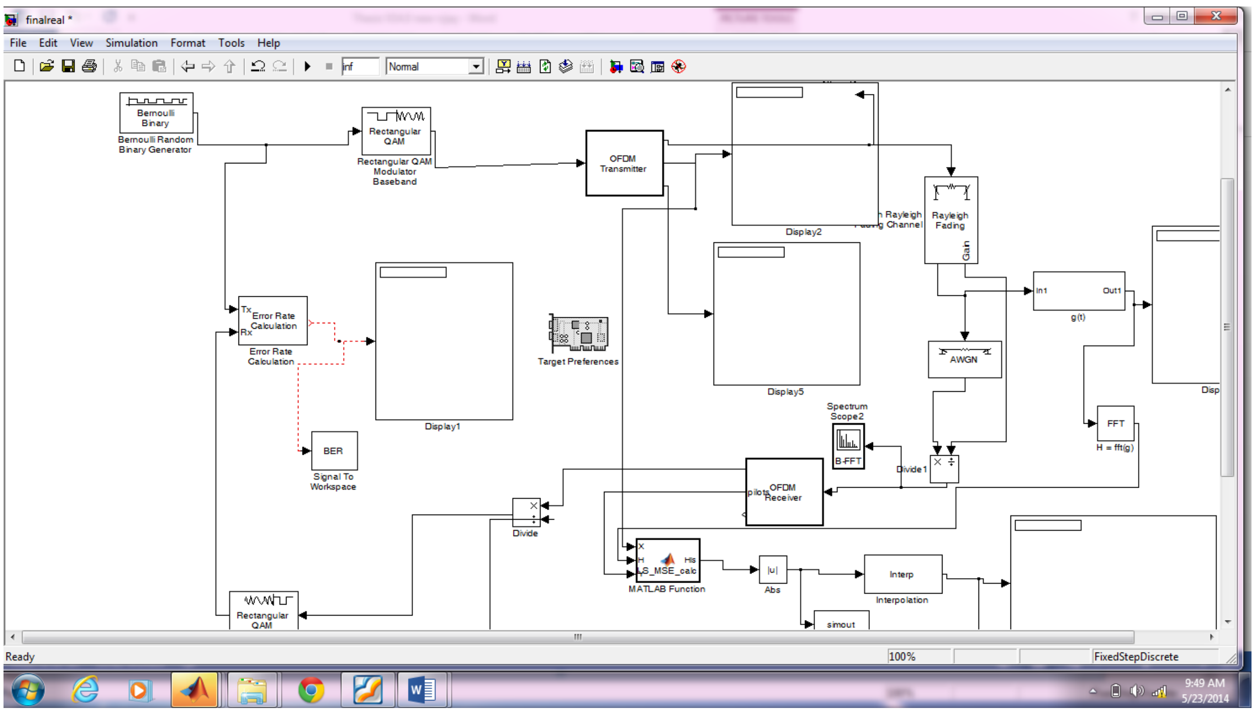

Figure 4. 5 setup diagram of interpolation channel estimation

Hardware implementation is done using c6713DSk kit with the help of external application like i mentioned in this chapter and below figure 4.6 shown below

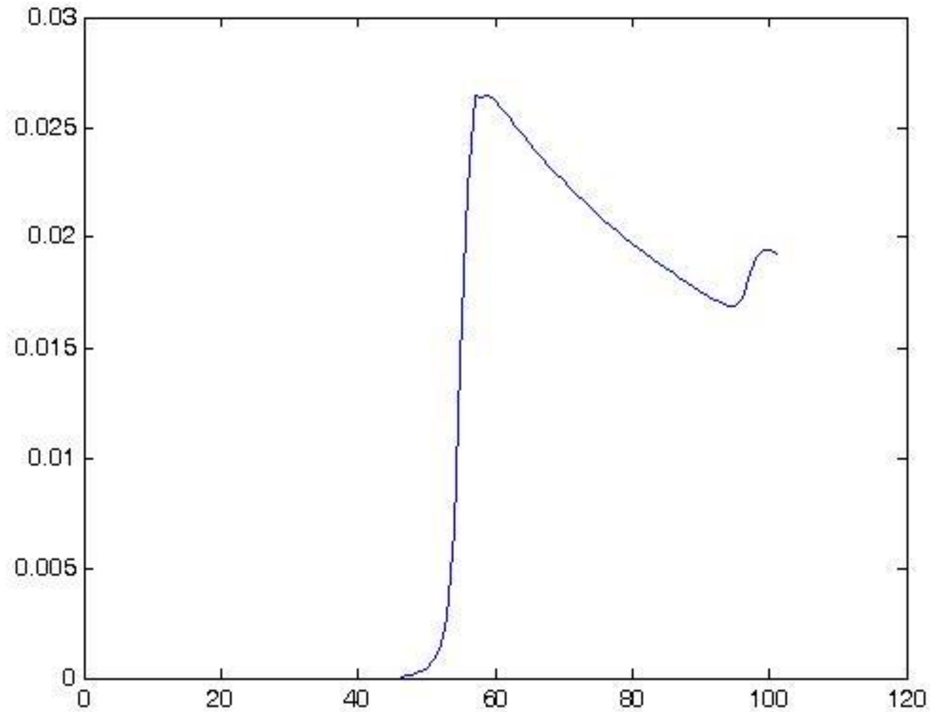

Figure 4. 6 DSP bit error rate

Presents the results for SNR=20dB and for SNR=10 dB. In both cases LMS, NLMS and RMS has performance that compares very favorably with the other two algorithms but with a very appealing computational complexity. In Fig. 4.7 we also plot the performance of the adaptive algorithms using less step size. We present the bit error rate (BER) of the RLS, LMS NLMS scheme with perfect channel knowledge, for different values of the SNR. We observe an indistinguishable performance of the adaptive schemes as compared to the one with perfect channel knowledge.

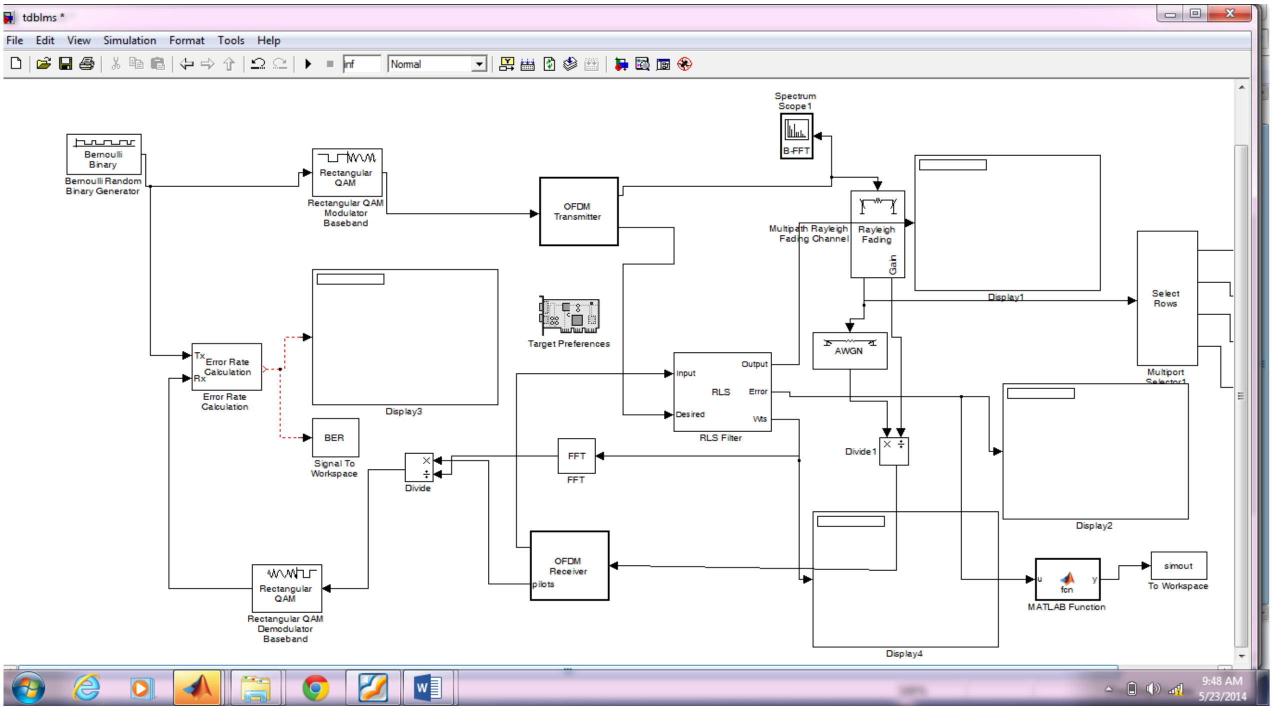

Figure 4. 7 LMS, NLMS, RLS, setup diagram

Chapter 5
Conclusion

5.1 Conclusions

In this thesis, a performance comparison between the different channel techniques based on the pilot arrangement is investigated. Comb-type channel estimation techniques are implemented in Simulink and compared with different interpolation techniques. Complexity increases in comb-type and block-type algorithms for that we estimated the other channel estimation techniques in a simple way to reduce the complexity. LMS, NLMS, and RLS are the Compared with the other channel estimation techniques and also compared LMS, NLMS, and RLS algorithms with different step-sizes using the SIMULINK models. Compared with the three algorithms RLS has high performance RLS algorithms has a high perform. And also implemented this algorithms in the real time implementation using the TMS320C6713 DSP by transferring Simulink model to the DSP c6713DSK kit with the help of CCS and verified the output with the help of the RTDX. This thesis focuses on channel estimation with different interpolation approaches and adaptive algorithms OFDM system.

5.2 Future work

The future works that can be conducted on the technique are deduced from the drawbacks of the techniques. Two dimensional estimations techniques more efficient then the above techniques but complexity increase.

References

[1] M. K. Ozdemir and H. Arslan, "Channel Estimation for Wireless OFDM Systems," IEEE Communications Surveys & Tutorials, vol. 9, no. 2, pp. 18-48, 2nd Quarter 2007.

[2] Henrik Schulze, Christian Luders, "Theory and Applications of OFDM and CDMA Wideband Wireless Communications", John Wiley & Sons Ltd., chap 4, pp 145-160, 2005

[3] Ali H. Sayed, Adaptive Filters, Wiley, NJ, 2008[13] (ISBN 978-0-470-25388-5).

[4] Van de Beek, J.J., Edfors, O., Sandell, M. et al. (July 1995) On channel estimation in OFDM systems. IEEE VTC'95, vol. 2, pp. 815–819.

[5] Hsieh, M. and Wei, C. (1998) Channel estimation for OFDM systems based on comb-type pilot arrangement in frequency selective fading channels. IEEE Trans. Consumer Electron., 44(1), 217–228.

[6] Tufvesson, F. and Maseng, T. (May 1997) Pilot assisted channel estimation forOFDMin mobile cellular systems. IEEE VTC'97, vol. 3, pp. 1639–1643.

[7] Fernandez-Getino Garcia, M.J., Paez-Borrallo, J.M., and Zazo, S. (May 2001) DFT-based channel estimation in 2D-pilot-symbol-aided OFDM wireless systems. IEEE VTC'01, vol. 2, pp. 810–814.

[8] Hu, D. and Yang, L. (Dec. 2003) Time-varying channel estimation based on pilot tones in OFDM systems. IEEE Int. Conf. Neural Networks & Signal Processing, vol. 1, pp. 700–703.

[9] Li, Y., Cimini, L. Jr, and Sollenberger, N.R. (1998) Robust channel estimation for OFDM systems with rapid dispersive fading channels. IEEE Trans. on Commun., 46(7), 902–915.

[10] Tang, Z., Cannizzaro, R.C., Leus, G., and Banelli, P. (2007) Pilot-assisted time-varying channel estimation for OFDM systems. IEEE Tran. Signal Processing, 55(5), 2226–2238.

[11] Rohling, H. and Grunheid, R. (May 1997) Performance comparison of different multiple access scheme for the downlink of an OFDM communication system. IEEE VTC'97, vol. 3, pp. 1365–1369.

[12] Bertrand Muquet, *Member, Cyclic Prefixing or Zero Padding for Wireless Multicarrier Transmissions? IEEE,* Zhengdao Wang, *Student Member, IEEE*, Georgios B. Giannakis,*Fellow, IEEE*, Marc de Courville, *Member, IEEE*, and Pierre Duhamel, *Fellow, IEEE.*

[13] Chassaing, Rulph. *Digital Signal Processing and Applications with the C6713 and C6416 DSK*. Vol. 16. John Wiley & Sons, 2004.

[14] Yuping Zhao, Aiping Huang, “A novel channel estimation method for OFDM mobile communication systems based on pilot signals and transform-domain processing ,” IEEE VTC , Vol. 3, May 1997.

[15] Simeone, Osvaldo, Yeheskel Bar-Ness, and Umberto Spagnolini. "Pilot-based channel estimation for OFDM systems by tracking the delay-subspace."Wireless Communications, IEEE Transactions on 3.1 (2004): 315-325.

[16] Colieri, Sinem, et al. "A study of channel estimation in OFDM systems."Vehicular Technology Conference, 2002. Proceedings. VTC 2002-Fall. 2002 IEEE 56th. Vol. 2. IEEE, 2002.

[17] Yang, Baoguo, et al. "Channel estimation for OFDM transmission in multipath fading channels based on parametric channel modeling." *Communications, IEEE Transactions on* 49.3 (2001): 467-479.

[18] M. Morelli and U. Mengali, “A comparison of pilot-aided channel estimation methods for OFDM systems,” *IEEE Transactions on Signal Processing*, vol. 49, no. 12, pp. 3065-3073, Dec. 2001.

Online Resources:

1. www.ti.com – Official Website of Texas Instruments
2. www.wikipedia.org
3. *http://www.mathworks.in/products/simulink/*
4. www.google.com – Search Engine for data and images