Quantization Noise Shaping for LTE Fronthaul Downlink

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Popular science summary

The communication networks have been developed in the years to meet the requirements of the modern life. One of those developments is the appearance of the so-called Centralized-Radio Access Network (C-RAN) in Long Term Evolution (LTE) and LTE-Advanced. C-RAN has major differences compared to the legacy RAN. In C-RAN, no changes happen on the Remote Radio Units (RRUs) but the Baseband Units (BBUs) are separated from the base station. BBUs in C-RAN have been gathered in a BBU pool away from the base station. The Common Public Radio Interface (CPRI) is used to connect the BBU and the RRU using fiber links. This connection which includes BBU, RRU and CPRI links is called the fronthaul.

The CPRI protocol is one of the best methods to be used in the fronthaul. On the other hand, CPRI has limitations in terms of higher data rates and it is known that the upcoming generations promise with much higher data rates. First thing could come to mind is to replace CPRI links or use an other protocol but this idea costs time, effort and money. This guides the researchers to try to apply the simple compression algorithms on the baseband signal before entering the CPRI interface. The compression has proofed that it is a high-efficient and low-cost method in many applications such as audio and video signal processing.

The compression algorithm which has been followed here is called noise shaping which follows the quantization. This algorithm is applied on different LTE Downlink (DL) bandwidths. To apply the compression, the LTE DL baseband signal should be analyzed and the first thing should be known about it that it is an OFDM signal. For all bandwidths, the OFDM DL signal is divided into frames and each frame has a number of subframes. The structure of each subframe is that each subframe contains 7 symbols which are separated using the Cyclic Prefix (CP).

CP is a redundancy added to the signal to avoid the Inter-Symbol Interference (ISI). The redundancy is increasing the length of the baseband signal at BBU and it plays no rule while transmitting over fiber links so the first thing to be done is to remove CP to increase the compression rate. In this stage, the baseband signal which is represented by 15 bits for each sample is ready to apply the compression algorithm on it. This starts with applying the quantization and then noise shaping.

Quantization is a process of mapping a continues set of values to a discrete set. There are two main characteristics of quantization they are quantization levels and quantization regions. The quantization levels depend on the number of bits used to represent the baseband signal and the quantization regions depends on the difference between the maximum and minimum values of the signal. There are two types of quantization, uniform quantization and non-uniform quantization. In uniform quantization, the quantization regions are uniform rather its non-uniform for the non-uniform quantization. Here, uniform quantization is used because its simpler to implement. After the signal is uniform quantized, the difference between the resulted output signal and input baseband signal gives the quantization noise. This results in a compression up to 7 bits for the uniform quantization regions. This results could be improved if the quantization noise is shaped.

One of the most common methods which helps to shape the noise is the Noise shaping method. This method is originally used to shape the noise after compression for the audio signals and later on it is used in many other applications. The quantization noise shaping (QNS) process is done based on Noise shaping method which depends on the feedback filtering for the noise. The basic Noise shaping method with single feedback loop is shown in the figure below.

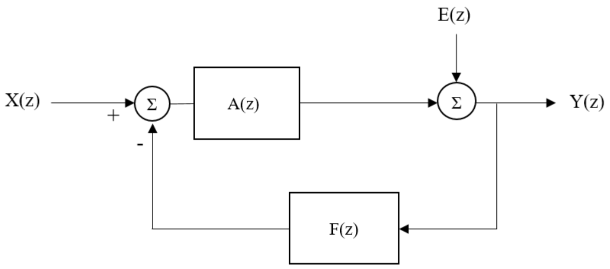

Figure 1: Noise shaping method

The quantization noise shaping is done by increasing the quantization noise at the edges (guard bands of the OFDM symbol) and decreasing the quantization noise in the middle (in band). This process results in one more bit reduction (6 bits) compared to quantization. This quantization noise shaping needs further improvements in shaping and for this the concept of oversampling is used.

The oversampling is a process to increase the number of samples by a certain factor for the generated signal, which results in a higher sampling rate. In this work, a factor of 2 is used to increase the number of samples. Oversampling will not affect the power of the signal but it will spread the quantization noise in a larger frequency band. Before doing the quantization noise shaping process, the oversampling is done for the whole baseband signal. Then this over sampled signal is quantized and noise shaped by repeating the whole process which we done before for quantization noise shaping. After this, the noise shaped and over sampled signal to be filtered by a low pass filter and down sampled it to get the original sampling rate of the signal. The use of oversampling gives a better shaping and a same bit reduction compared to QNS of the signal without oversampling. The final design which covers all previous steps is shown below.

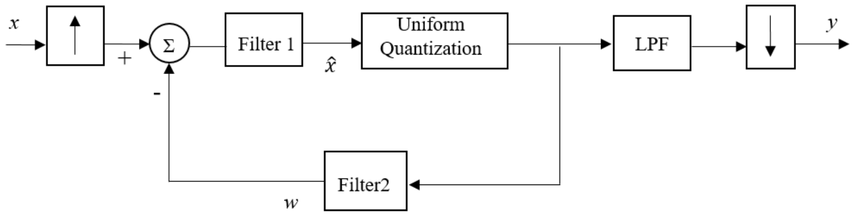

Figure 2: Oversampling and Quantization noise shaping where x denotes the input signal and y denotes the output

In this thesis, the noise shaping is done for three different DL LTE bandwidths: 5, 10 and 20 MHz. The output of the quantized noise shaped signal after oversampling is obtained and is shown in the figure below for 10 MHz bandwidth.

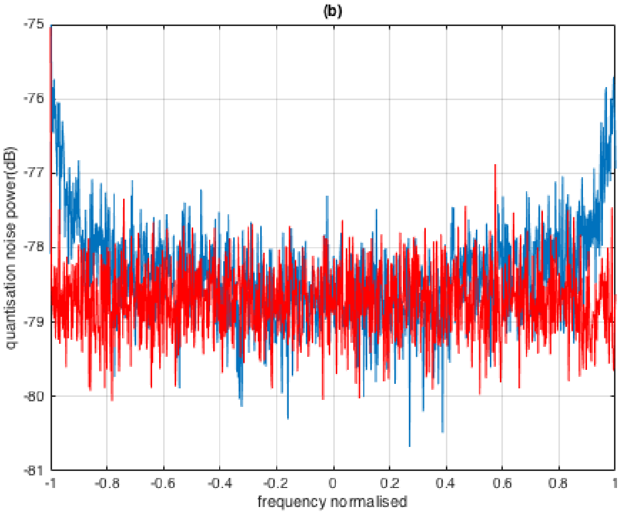

Figure 3: Quantization noise shaping after Oversampling for 10 MHz

The Error Vector Magnitude (EVM) is taken to evaluate the results step by step and it has been proved that all values are under EVM maximum value which gives a good impression about applying noise shaping to compress data before entering CPRI interface. The compression rate which has been reached is around 2.5 (i.e. each sample could be represented by 6 bits instead of 15). This may allow the established links to transport much higher data rates with a simple and cost efficient method.

Abstract

The modern mobile networks such as the Cloud Radio Network Access (C-RAN) are developing to deal with upcoming requirements in the current generation 4th Generation (4G) and the upcoming generation i.e. the 5th Generation (5G). One of the challenges in the modern communication systems is the required high data traffic specially at the fronthaul. Fronthaul in Long Term Evolution (LTE) contains the Baseband Unit (BBU) and Remote Radio Unit (RRU). These two units are connected using a protocol called by Common Public Radio Interface (CPRI). The CPRI is containing a set of fiber links used in the fronthaul to transport data between BBU and RRU in both Uplink (UL) and Downlink (DL). The fiber optic cables have a limited amount of data to transport. This leads to a limitation of transmitting higher data rates in both DL and UL directions. To solve this problem, a compression algorithm is needed to compress the data before transmission.

In this thesis a compression algorithm is introduced to compress the complex baseband LTE DL signal for different bandwidths at the BBU. The concept of quantization is used to obtain the compression. The uniform quantization is used to reduce the number of bits. To improve the results of quantization, the concept of Quantization Noise Shaping (QNS) is introduced. The QNS system is based on Noise shaping method. In this, the feedback filtering system is used to shape the quantization noise. This gives a one more bit reduction compared to quantization. Furthermore, we used the oversampling concept by oversample the complex baseband signals to get better results. This gives better shaping and same bit reductions.

The simulation is done using MATLAB LTE System Toolbox which offers LTE test complex baseband signals and built in functions. The results are evaluated according to 3rd Generation Partnership Project (3GPP) standardization by measuring the Error Vector Magnitude (EVM) and Signal to Quantization Noise Ratio (SQNR).

Acknowledgments

First and foremost, we offer our sincere gratitude to our supervisor, Stefan Höst for giving us this opportunity to work on this thesis. He supported us throughout the thesis with patience and knowledge. We would particularly like to thank him for giving us the full freedom to work in our way and without him this thesis would not be completed or written on time.

His guidance and advice taken us in right way and the knowledge which we got from him is unforgettable. Another thanks to our assistant supervisor Yezi Huang, who helped us in the beginning phases and her ideas helps immense characterization of the work. We wish to thank our examiner Maria Kihl for her thorough and detailed review of our project.

We would like to thank all our teachers in EIT for giving us the knowledge to work in this field. Their appreciation helps us to work well in this project.

Another important thanks to our families and friends, who gave their hands in our need time and understood our work, cheered us in all the situations. We want this help in our whole life to grow tall in the future.

Preface

This thesis work is done by Alaa and Veereswari, with assistance from supervisors Stefan Höst and Yezi Huang, Electrical and Information Technology (EIT), Lund University. Both the authors work in equal spaces in all the situations and both played active role when working in the coding and documentation. During the initial stages of the project, with the help of supervisors, both designed the system. The work was divided when it comes to documentation. The Chapters 1,2 done by Alaa and the Chapters 3,4 done by Veereswari, by that all the remaining chapters are done in group together.

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List of Acronyms

3G	3rd Generation
3GPP	3rd Generation Partnership Project
4G	4th Generation
5G	5th Generation
ADC	Analog to Digital Converter
BBU	Base Band Unit
BW	Bandwidth
C-BBU	Centralized - Base Band Unit
CP	Cyclic Prefix
CPRI	Common Public Radio Intefrace
C-RAN	Cloud- Radio Access Network
DAC	Digital to Analog Converter
DC	Direct Current
DL	Downlink
DRoF	Digital Radio over Fiber
ETSI	European Telecommunications Standards Institute
EVM	Error Vector Magnitude
FDD	Frequency Division Duplex
FFT	Fast Fourier Transform
IDFT	Inverse Discrete Fourier Transform
ISI	Intersymbol Interference
LPC	Linear Predictive Coding
LTE	Long Term Evolution
LTE-A	Long Term Evolution - Advanced
MPLS	Multi Protocol Label Switching
NDLRB	Number of Down Link Resource Blocks
OBSAI	Open Base Station Architecture Initiative

OFDM	Orthogonal Frequency Division Multiplexing
OFDMA	Orthogonal Frequency Division Multiplexing Access
ORI	Open Radio Interface
OTN	Optical Transport Network
PHY	Physical Layer
PSD	Power Spectral Density
QAM	Quadrature Amplitude Modulation
QNS	Quantization Noise Shaping
QPSK	Quadrature Phase Shift Keying
RAN	Radio Access Network
RB	Resource Block
RE	Radio Equipment
REC	Radio Equipment Control
RRU	Remote Radio Unit
SC-FDMA	Single Carrier Frequency Division Multiplex Access
SQNR	Signal to Quantization Noise Ratio
TDD	Time Division Duplex
TM	Test Model
TMN	Test Model Number
UE	User Equipment
UL	Uplink
UMTS	Universal Mobile Telecommunications System
UTRA FDD	Universal Terrestrial Radio Access Frequency Division Duplex
V-BBU	Virtual- Base Band Unit
WLAN	Wireless Local Area Network
WiMAX	Worldwide interoperability for Microwave Access

Chapter 1

Introduction

LTE, the growing modern technology, offers unprecedented data rates, high capacity and short latency. Advanced modulation techniques are used for transmission in LTE such as Orthogonal Frequency Division Multiple Access (OFDMA) and Single Carrier Frequency Division Multiple Access SC-FDMA. LTE applies OFDMA in Downlink (DL) transmission and SC-FDMA in Uplink (UL) transmission.

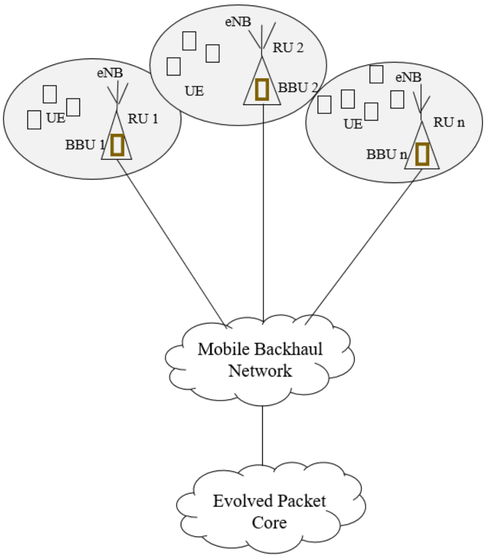

Figure 1.1: Legacy Radio Access Network (RAN)

The legacy Radio Access Networks (RAN) consists of separated Radio Units (RU) and separated Base Band Units (BBU) which are distributed among the region of the mobile network to provide the connection. RU and BBU are interfaced using CPRI links. BBUs are connected to the backhaul through fiber optic links. In the modern mobile networks, a huge growth in the data traffic is occurred. This can not be achieved in the legacy RAN without installing small cells or develop the legacy RAN. This has been achieved by creating a network which can deal with the increment of the data traffic and to cover more areas in a simple and cost efficient way.

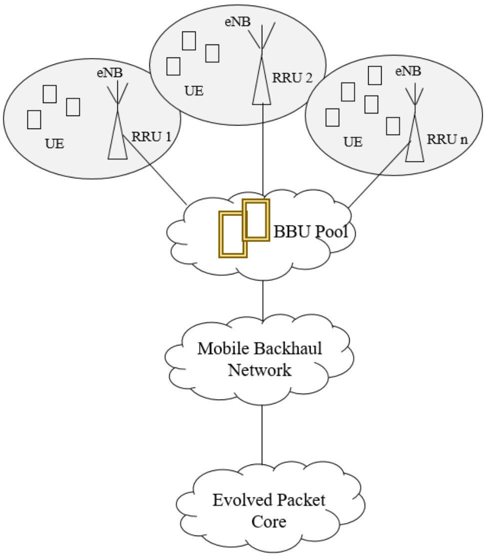

Figure 1.2: Centralized Radio Access Network (C-RAN)

A centralised BBU (C-BBU or BBU hostelling) is introduced in the modern networks to be replaced with all individual BBUs in the legacy RAN. The C-BBU is installed to interface with the RRUs using CPRI interface. BBUs can be centralised at macro cell sites to control the spectrum while reducing the power consumptions. This clean, centralised processing and real time technology is also called Cloud Radio Access Network (C-RAN).

C-RAN is implemented using servers and switches as C-BBU and virtual BBU (V-BBU). It uses several cores to meet the standard requirements of the wireless networks (i.e. 2G, 3G, 4G and 5G). The usage of C-RAN provides benefits such as improving the network security, ability to self optimization, and the operational costs can be controlled.

The fronthaul, which is a new network segment, includes the C-BBU and the distributed RRUs in the C-RAN. The internal Interface which is connecting BBU and RRU in both Legacy RAN and C-RAN is called CPRI (Common Public Radio Interface), OBSAI (Open Base Station Architecture Initiative) and ETSI ORI (European Telecommunications Standards Institute, Open Radio Interface) specifications. Both CPRI and OBSAI solutions are implementing digital radio over fiber (DRoF) so that the sampled, quantized and coded signal will be transmitted to the BBU pool. The ORI is introduced by operators like SK Telecom, KDDI and vendors like ZTE and Samsung. The specification of ORI is based on CPRI specification but it is designed for specific but unclear options.

Nowadays the most used by C-RAN vendors is CPRI since the mapping methods of it is more efficient. In C-RAN, the CPRI interface can be used to fit the co-locating of BBU depending on the number of cell sites in a common location (BBU hostelling). The fronthaul has some requirements regarding radio site configurations, bitrate per antenna site and latency. CPRI is transmitting a constant bit rate data over assigned channel using a serial interface.

The balance between the huge data traffic and the low latency between pooled BBUs and RRUs is the main aim for the network operators. They started by building a direct fiber connectivity (also known as dark fiber). This method is good but it is not practical for the future applications of the small cells. According to network infrastructure technologies such as Optical Transport Network (OTN) and Multi-Protocol Label Switching (MPLS), the implemented fiber system is difficult to manage and troubleshoot problems occurred in C-RAN. This is one of the challenges in C-RAN.

To solve this problem, an upcoming new fiber system is available in the market. It is called the active fiber monitoring which allows the operators to control the fiber during provisioning and while in service. This requires to replace OTN and MPLS with new technologies to meet the new fiber system requirements. This costly solution can be added to other solutions which deal with the data before entering the CPRI links. Compression of the data by using Quantization Noise Shaping (QNS) or by using Linear Predictive Coding (LPC) could help the infrastructure technologies to meet the upcoming requirements and may help to not replace the fiber system. In this report, the compression of data using QNS is introduced.

1.1 Problem Definition

The physical infrastructure between C-BBU and RRU causes a big challenge in the C-RAN. The problem is that the infrastructure technologies which are used in the legacy RAN can not be used in the upcoming generations due to the huge data traffic requirements. The usage of the old infrastructure technologies in C-RAN leads to a limitation in terms of adding more antennas or using higher bandwidths.

The available solutions provided by the fiber optic producers are cost efficient and complex. In this thesis, a simpler solution based on the compression will be implemented. Initially, the compression method was introduced to be used in the audio signal processing. Audio compression enables to distribute the signal without using a big amount of media storage or transmission bandwidth. The aim of compression is to reduce the data transmitted. Audio compression and decompression can be achieved by many types of algorithms such as quantization [16].

In LTE DL, the OFDMA is applied to offer a transmission form which includes a high number of subcarriers with a low data rate modulation. This high bandwidth modulation format requires a compression algorithm before transmitting the data from the BBU to RRU. In the typical LTE DL, the data at the BBU is compressed using a 15-bits quantizer where the real samples and the imaginary samples are quantized individually. The quantization can be tested for less number of bits also and gives minimum Error Vector Magnitude (EVM) under the 3GPP standardization. To get furthermore compression the QNS method is introduced. QNS is a low-complexity compression method for LTE to meet the upcoming requirements in the next generation of communication. This choice should deal with the biggest challenge in the new networks which is the huge data traffic.

This thesis is introducing a method to compress the data in the LTE DL transmissions at the BBU. This method of compression is based on the quantization and the noise shaping. The quantization results in a certain compression but gives quantization noise. This noise should be shaped to get further compression for the data by applying a good filtering system. The simulation is using the MATLAB LTE System Toolbox for different bandwidths and lower number of bits.

The compression algorithms help the C-RAN to deal with the limitations due to the usage of CPRI such as the number of fiber links and the infrastructure technologies. The compression should be done without affecting the 3GPP standardization.

1.2 Related work

Initially, the compression of data using the process of shape the quantization noise is done for the audio signals. Many old researches were focusing on use this method in other applications excluding the LTE signals such as *Least Squares Theory and Design Of Optimal Noise Shaping Filters* [13]. This paper is discussing the ability to use quantization to minimize the word length of the audio streams and how to shape the distortion results in from quantization using a noise shaping feedback filtering system. The method used is called Dithered requantization with error feedback filter and requantization error as shown in the Figure 1.3.

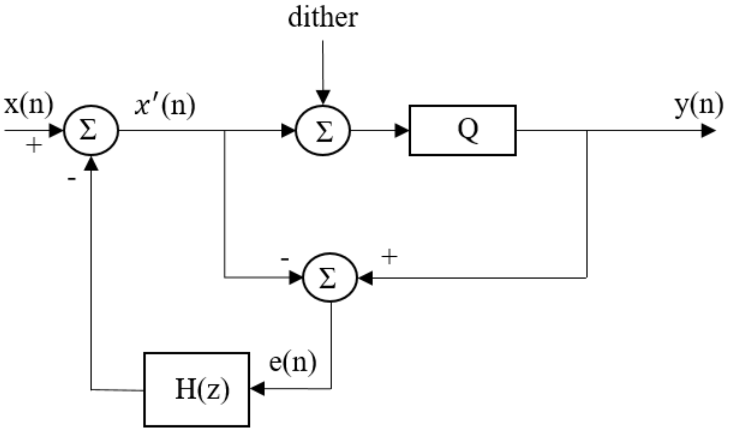

Figure 1.3: Dithered requantization with error feedback method

In this thesis work, the compressing method used is based on Sigma Delta modulation i.e. a method which could be used in compressing the audio signals. The IEEE has published *Cascaded Noise Shaping For Oversampling A/D and D/A Conversion* [12] was our reference to understand the Sigma-Delta modulation and how it could be used to meet the aim of this project. This concept is depending on designing a feedback filtering which is used in A/D and D/A converters, includes noise shaping method. The Figure 1.4 shows the basic Sigma- Delta modulation block diagram.

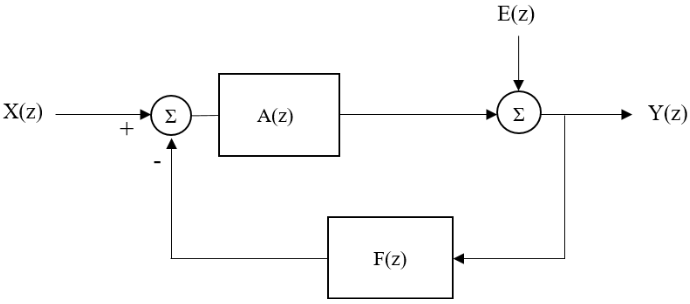

Figure 1.4: Sigma-Delta modulator

The compression of data is achieved in many previous researches. In 2012, Alcatel-Lucent introduced a compression algorithm published under the title *CPRI compression transport for LTE and LTE-A signal in C-RAN* [14]. The algorithm is based on the characteristics of LTE DL data. The first step is to remove redundancies in the spectral domain, then the block scaling combined with a non linear quantizer which is designed in Alcatel laboratories. The idea of this paper is to design a system to reduce the quantization error without shaping. The idea result achieved by this paper is compression is 11 bits but the compression could be done until 6 bits if the User Equipment (UE) and RRU are designed carefully.

In 2013, a research with title *Time-Domain Compression of Complex-Baseband LTE Signals for Cloud Radio Access Networks* [8] has been published. This paper discusses a method of compressing the baseband LTE signal using a non-linear Gaussian optimized quantizer. The noise occurred from the quantization is shaped using filters and the simulation is done using the LTE Link Level Simulators developed at the Vienna University of Technology. The results achieved in this paper is a 5 bits reduction.

Furthermore, other methods based on compression are developed to solve the problem of upcoming huge data rates such as the Linear Predictive Coding (LPC) [15][18].

1.3 Methodology

The methodology used in this thesis is analysed in this section. This depends on studying the theoretical background of the LTE fronthaul and its limitations and analysing the previous works. The investigation is done to solve the problem as described below:

1. Study the behaviour of LTE downlink signals and OFDM modulation format in order to prepare the signal for our system. The MATLAB LTE system

Toolbox is used to generate the downlink signal and analyse it.

2. Deep theoretical Study about relative work in this field and our study started with a study of the Quantization types and Designing a suitable Quantizer to meet the full system. The quantizers types described in this project are the uniform quantizers, non-uniform quantizers and the uniform quantizers followed with Huffman coding. The uniform quantizer has chosen in our design because of it's simplicity.
3. The second step is to study the Sigma-Delta modulation and derive the filtering system to proceed with the shaping process. QNS is the heart of our system so a deep study has been done for the type of the modulators and for the types of filters in order to meet our requirements. The Single-Quantizer Sigma-Delta modulator is finally chosen with a feedback filtering system. The Chebychev type 2 filters are used with certain cut off frequencies and suitable order.
4. The Oversampling is used to improve the quantization noise shaping to meet our aim of the project. It is done by oversample the original signal with factor 2. This gives a higher number of samples in the same bandwidth so the QNS will be more flexible and it should give better results.

1.4 Thesis Outline

This thesis consists of seven chapters. Chapter 1 presents a brief introduction about the project, problem definition, aim of project, methodology and the thesis outline. Remaining chapters are described below:

- Chapter 2 describes the LTE downlink fronthaul and usage of CPRI protocol. LTE downlink baseband signal is analysed. MATLAB LTE Toolbox is also discussed.
- Chapter 3 analysis the characteristics of quantization and its types. Here we decided the quantization method to be used in the project.
- Chapter 4 gives details about the quantization noise, quantization noise shaping and oversampling.
- Chapter 5 describes the final system implementation which is carried on regular steps. Block diagram for each step is also added in this chapter.
- Chapter 6 discusses the final results obtained by simulation.
- Finally, Chapter 7 concludes the thesis work and discusses future work proposals in this field.

Chapter 2

Overview of LTE Downlink Fronthaul

The main aim in this thesis work is to understand the fronthaul and the signal behaviour in LTE system. In this chapter, the fronthaul characteristics, the CPRI specifications and the LTE DL baseband signal at the BBU will be analysed. Moreover, the signal generation using Matlab LTE System Toolbox is discussed in this chapter.

2.1 Definition of LTE fronthaul and CPRI protocol

The Fronthaul in mobile networking can be defined as the connection between the BBU and the RRU. The communication protocol running over the fiber between the BBU and the RRU is the CPRI protocol. These links should be fast in order to synchronize the transmissions across the network. Fronthaul is a one of the network elements that makes LTE-A networks a reality.

2.1.1 CPRI Interface specifications

Fiber and CPRI together allow mobile network operators to deliver better quality and faster service to all mobile device users. CPRI was established in 2003 by base station vendors, such as Ericsson, Nokia Siemens Networks, Alcatel Lucent, and Huawei Technologies [1]. The aim behind using CPRI is to define a specification that standardizes the protocol interface between BBU and RRU.

Furthermore, the incoming uplink signal is digitized at the RRU. The digitized data is then transported using the CPRI protocol links. The same process is used when traffic is entered via the IP Backhaul i.e. downlink data. Once the RRU receives this data, it then converts it to analog, amplifying and radiating it over the air to the User Equipment (UE). The Figure 2.1 describes the CPRI transport concept and the fronthaul contents in the LTE system.

2.1.2 CPRI versions and line bit rate options

The most recent CPRI specification is version 7.0 which has been released in 2015. The new 7.0 release adds 24G line-rate to the previously released 10G LTE-Advanced [2].

All versions of CPRI interface specification will have the following scope:

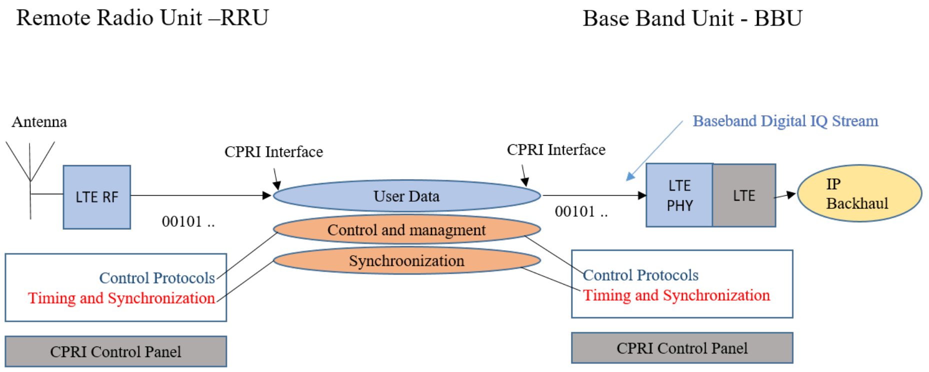

Figure 2.1: CPRI transport concept in LTE systems

1. A digitized base station interface between 'Radio Equipment Control' (REC) and Radio Equipment (RE).
2. The specification covers layer 1 and layer 2.
3. Physical layer (layer 1) supports an electrical interface and optical interface.
4. Layer 2 shall support flexibility and scalability.
5. The Specification shall comply with 3GPP Universal Terrestrial Radio Access Frequency Division Duplex (UTRA FDD) release 5.

CPRI recognizes many interface rates/options. The latest version has the options which is shown in Table 2.1.

Table 2.1: CPRI data rates options for version 7.0

CPRI line rate option	Data rate (Mbps)
1	614.4
2	1228.8
3	2457.6
4	3072.0
5	4915.2
6	6144.0
7	9830.4
7A	8110.08
8	10137.6
9	12165.12
10	24330.24

All CPRI line bit rates have been chosen in such a way that the basic UMTS chip rate of 3.84 Mbit/s can be recovered in an efficient way.

2.2 Overview of LTE DL baseband signal

LTE is designed to transmit packets of data efficiently using OFDM modulation. This section explains how LTE System Toolbox is used to generate LTE DL baseband signal and how OFDM is applied. It shows also a brief analyzing of the structure and the organization of the data in DL transmission.

2.2.1 OFDM in LTE Downlink advantages and modulation format

The aim behind using OFDM is that it offers a transmission form that uses a huge number of nearby spaced carriers which are modulated with a low data rate. It would come into mind that the carriers may interfere but making the signals orthogonal leads to no interference. OFDM is used in LTE to provide a multiple access scheme using OFDMA in the downlink and SCFDMA in the uplink. It is also used in many other systems from Wireless Local Area Network (WLAN), Worldwide interoperability for Microwave Access (WiMAX) to broadcast technologies.

Advantages of OFDM

OFDM is an efficient bandwidth technique. It has many advantages including its robustness to fading and interference. As mentioned before, the implementation of OFDM is different in the downlink and the uplink because of the different requirements and equipment's for the two directions. In brief words, OFDM is a format which carries high data rates that is why it is a good choice for LTE system.

Modulation format

The available modulation types for OFDM LTE downlink signal are Quadrature Phase Shift Keying (QPSK) with 2 bits per symbol or Quadrature Amplitude Modulation (QAM) i.e. 16QAM with 4 bits per symbol, 64QAM with 6 bits per symbol and 256QAM with 8 bits per symbol.

The choice of modulation format depends on the dominating conditions i.e. signal to noise ratio (SNR). The QPSK do not require a large SNR because it has fewer signal points which is more robust for the same power. On the other hand using of higher order modulation forms is used only when there is a sufficient SNR.

2.2.2 OFDM frame structure

In LTE standards the Frequency Division Duplex (FDD) and Time Division Duplex (TDD) are used. In FDD, both UL and DL frames are 10ms and can be separated by either time or frequency but multiplexed in frequency domain. For TDD both UL and DL frames are multiplexed in time domain and transmitted on the same frequency. In this thesis, the FDD frame structure is used and shown in the Figure 2.2.

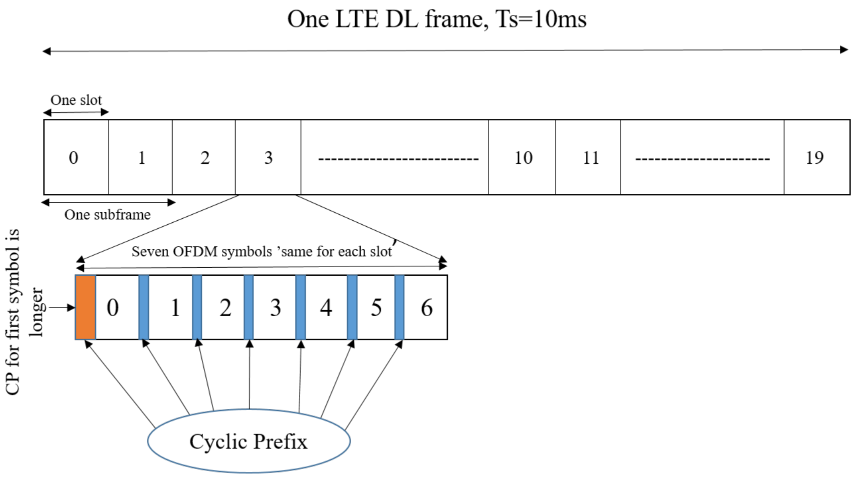

Figure 2.2: OFDM FDD frame structure

2.2.3 OFDM characteristics for different LTE bandwidths

A main key parameter related to the use of OFDM in LTE is the bandwidth. The choice of the bandwidths may affect the number of subcarriers and the symbol length that could be used in the OFDM signal. According to 3GPP standardization, the available bandwidths to be used in LTE are 1.4 MHz, 3 MHz, 5 MHz, 10 MHz, 15 MHz and 20 MHz. In LTE downlink, the OFDM subcarrier space is divided into resource blocks. This helps to divide the data and transmit it across standard numbers of subcarriers. Each resource block contains 12 subcarriers and each signal bandwidth has a certain number of resource blocks. The Table 2.2 specifies the number of subcarriers and resource blocks (RBs) for different LTE bandwidths

Table 2.2: Number of subcarriers and RBs for LTE DL frequencies

BW	RB	NFFT size	In-Band	Guard bands
1.4 MHz	6	128	73	55
5 MHz	25	512	301	211
10 MHz	50	1024	601	423
15 MHz	75	1536	901	635
20 MHz	100	2048	1201	847

Resource blocks for 1.4 MHz

Each LTE DL channel frequency has a different number of RBs. The RBs are containing one subcarrier called the Direct Current (DC) subcarrier which has no information and used as a reference. This DC subcarrier will not be transmitted but will be counted in the number of subcarriers. The Figure 2.3 shows the RBs in LTE DL signal for 1.4 MHZ.

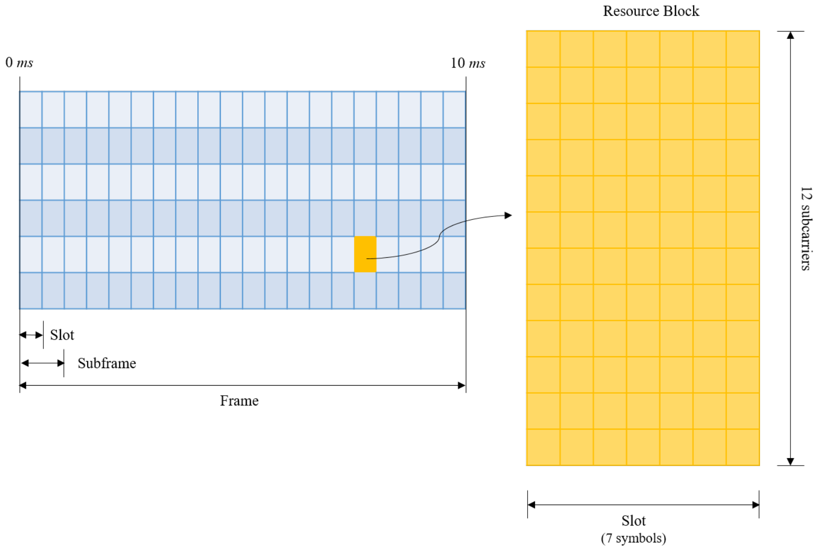

Figure 2.3: Resource blocks in LTE for 1.4 MHz

A resource block can be allocated to one user. Each RB is 180 kHz in frequency domain and one slot in time domain. In frequency domain, the size of the RBs is 12*15 kHz or 24*7.5 kHz subcarriers. For 15 kHz subcarrier spacing, this gives a sample rate of $1/15$ kHz $= 66.7$ μs to obtain orthogonality. Each subcarrier can carry a maximum data rate of 15 ksym/s. For 20 MHz bandwidth, the raw data rate will be 18 Msym/s and if each symbol uses 64 QAM, the raw data rate becomes 108 Mb/s.

2.2.4 LTE OFDM Cyclic Prefix

One advantage of using OFDM in LTE systems is its resilience to multipath fading's and spread but it will be necessary to add more resilience to the system to avoid the so called inter-symbol interference (ISI). Inserting a guard period at the beginning of each symbol by copying a section from the end of the symbol to the beginning is known as CP which is a good method of adding resilience. In Figure 2.4 it is shown how the CP van be added to an OFDM symbol.

This helps the receiver to prepare the waveform for equalization and avoid ISI by times up to the length of CP. The length of CP should be chosen carefully. It should be long enough to act against the multipath reflections delay spread. In LTE the standard length of the cyclic prefix is 4.69 μs. CP length is varying for each LTE DL signal frequency. The Table 2.3 illustrates the normal CP allocations for each LTE channel bandwidth.

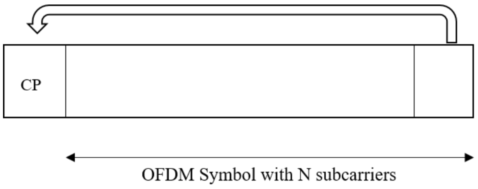

Figure 2.4: Concept of adding CP to each OFDM symbol

Table 2.3: Channel bandwidth and cyclic prefix normal lengths

Channel bandwidth	**CP normal length (Ts)**	
MHz	**OFDM symbol =0**	**OFDM symbol=1, 2..6**
5	40	36
10	80	72
15	120	108
20	160	144

2.3 Signal generation

MATLAB LTE System Toolbox provides functions and apps to design, simulate, analyze, and test the physical layer of LTE and LTE-Advanced wireless communication systems according to LTE standards. This system toolbox accelerates the simulation developments in the Physical layer (PHY).

2.3.1 Input arguments and output arguments

The input and output arguments are based on the LTE standardization. The test model configuration (TM), specified as a scalar structure. Using of `lteTestModel` function generates the various TM configuration structure. This configuration structure then can be modified as per requirements and used to generate the waveform. This input argument relates to different fields such as Channel Bandwidth (BW) which should be one of 1.4 MHz, 3 MHz, 5 MHz, 15 MHz or 20 MHz and test model number (TMN). Table 2.4 shows more details about the input arguments according to manual in Math works [3].

Table 2.4: Input argument for lteTestModel in matlab

Parameter field	**Values**	**Description**
TMN	’1.1’, ’1.2’, ’2’, ’3.1’, ’3.2’	Test model number
BW	’1.4’, ’3’, ’5’, ’10’, ’15’, ’20’	Frequency in MHz
Cyclicprefix	’normal’	Cyclic prefix length
Totsubframe	Nonnegative scalar integer	Subframes to generate

The generated waveform is a time-domain waveform (numeric matrix) of size T-by-P. Where P is the number of antennas and T is the number of time-domain complex samples. TM configuration, returned as a scalar structure. This argument contains information about the OFDM modulated waveform and TM configuration parameters. The TM contains TMN type, BW type, number of DL resource blocks (NDLRB) and CP.

2.3.2 LTE system toolbox to generate DL signal

To generate the DL baseband waveform using the LTE system Toolbox, the `lteTest ModelTool` function can be used. The `lteTestModelTool` starts a user interface for the generation of the test model waveforms. It is a T-by-P array, where T is the number of time-domain samples and P is the number of antennas and in this thesis the number of antennas is assumed to be only one antenna. The structure of the generated waveform is following the OFDM modulation characteristics.

Example of generating LTE DL signal

The generation of time domain signal with two dimensional array of resource elements, TM 3.2 and BW 15 MHz is shown in the following MATLAB code.

```
tmn = ’3.2’;
bw = ’15MHz’;
tmcfg = lteTestModel(tmn,bw);
[txWaveform,txGrid,tm] = lteTestModelTool(tmcfg);
```

where,

`tmn` is the test model number.
`bw` is the bandwidth.
`tmcfg` is the test model configuration.
`txWaveform` is the generated complex baseband signal.
`tm` is the test model

The output is a complex baseband signal and tm returns the specification of the generated signal such as NDLRB, Duplex mode, total number of subframes, etc.

2.4 Error Vector Magnitude

EVM is a measurement used to specify the behaviour of the digital transmitters and receivers. It is also called Receive Constellation Error (RCE). EVM measures how much the initial positions of the constellation points affected by the implementation imperfection resources (such as carrier leakage and phase noise). It is actually measures the difference in positions between the ideal locations and the received points locations.

In LTE systems, the EVM can be defined as the ideal received waveform and measured waveform for allocated resource blocks. The basic unit of EVM is measured for only active tones and over one subframe (1 ms) in the time domain and 12 subcarriers (180 kHz) in the frequency domain. This is why the EVM calculations is more important than the SNR which can be calculated only in the time domain and for all tones [17].

The formula below shows how to find EVM in general:

$$EVM(\%) = \sqrt{\frac{P_{\text{error}}}{P_{\text{reference}}}} \cdot 100(\%)$$

Where, P_{error} is the vector power and $P_{\text{reference}}$ is the power of the signal.

Chapter 3

Quantization

The process of converting the analog signal into digital signal consists of two steps, which is sampling and quantization. As we already have a sampled signal, there is no need for sampling. For the sampled signal, the next process we are doing is called the quantization which results in a binary vector. To get the reconstructed value of the input signal, inverse quantization is needed. This process is shown in the Figure 3.1 for the clear understanding of the system.

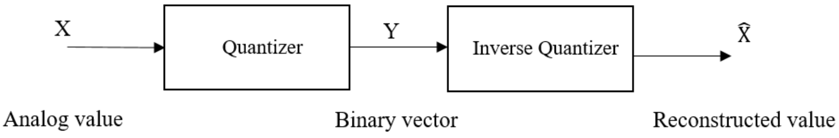

Figure 3.1: General quantization system

In this thesis, the quantization part is carried on and the process of mapping from an infinite set of values to a smaller set is called the quantization. It is an inherently nonlinear and irreversible processes because it is impossible to generate the exact input value from the only given output value. The quantization is a simple way to quantize the signal by choosing the digital amplitude value nearest to the original analog amplitude value. This gives the reconstructed signal and the difference between that and the original signal is called the quantization error. Here, the quantization is done for the sampled signal.

3.1 Analysing the characteristics of quantization

Quantization has characteristics like, it is an irreversible process, this becomes a source for information loss, this creates an impact on distortion of reconstructed signal. The input-output characteristics of quantizer is like the staircase function. When doing quantization, there are three main issues to be handled which helps in analysing the characteristics of quantization. The issues are how to (i) choose the number of quantization levels, (ii) find the values of quantization levels, (iii) map the original values of the signal to the quantization levels. This quantization levels play an important role when constructing a new quantized signal and it varies according to which type of quantization is used.

3.2 Types of quantization

There are two types of quantizers and they are uniform and non-uniform. In uniform quantizer, the quantization levels are equal whereas in the nonuniform its not equal.

3.2.1 Uniform quantization

In uniform quantization, all quantization regions (Δ) are equal size except the first and last regions if samples are not finite valued. Uniform quantizer has two types, Mid-rise and Mid-tread, their differentiation is based on how the quantization regions are divided around the value 0. The Mid-tread have a zero-valued reconstruction level (i.e. it has the zero output level) and the quantized signal with odd number of output steps. The Mid-rise have a zero-valued classification threshold (i.e. it does not have the zero output level) and the quantized signal with even number of output steps. The two quantizers are shown in the Figure 3.2 below.

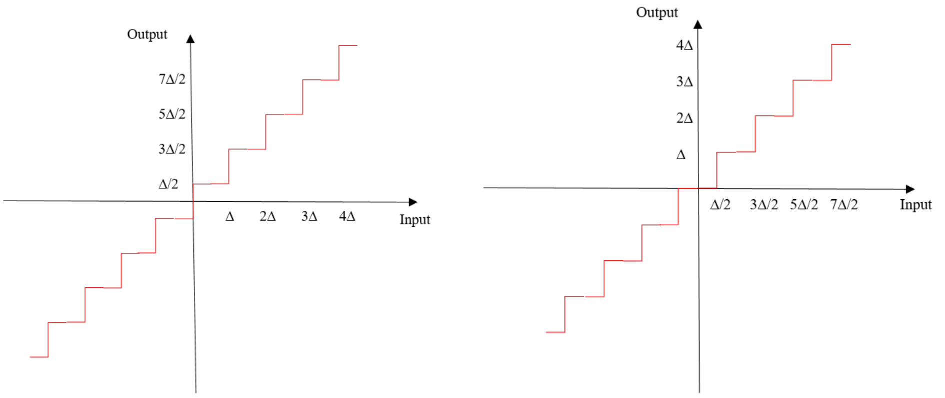

Figure 3.2: Mid-rise quantization (left) and Mid-tread quantization (right)

If the number of bits represented by R bits, then the mid-rise quantizer will have 2^R codes and the mid-tread quantizer will have $2^R - 1$ codes. The mid-tread quantizers generate better quantization with smaller number of codewords.

3.2.2 Nonuniform quantization

In nonuniform quantization, the quantization regions (Δ) are not equal size and it is required sometimes when the input signal distribution is nonuniform over the dynamic range. The quantization regions are arranged depends on the amplitude of the input signal. For the larger amplitude input signal the quantization is done by larger quantization steps and the smaller amplitudes are quantized by smaller quantization steps. To use nonuniform quantization effectively, the distribution of the quantizer input concentrates more in the lower amplitude range. To design a general approach for nonuniform quantization, compander design approach is used and it is shown in the Figure 3.3.

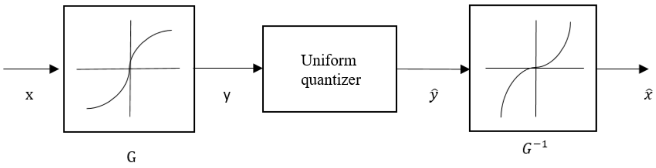

Figure 3.3: Design of nonuniform quantizer using the compander approach

In this approach, there are 2 stages, the first stage output $y = G(x)$ and the inverse transform of this called the second stage or last stage to produce output $x = G^{-1}(y)$. In the figure, the nonlinear transform G called the compressor and the inverse transform G^{-1} called the expander. The combination of both the compressor and expander called the compander approach.

3.3 Quantizer with Huffman coding

Huffman coding after quantizer is an another way to do bit reduction. It is a lossless data compression technique and is based on frequency of occurrence of a data item. The principle of huffman coding is to use the lower number of bits to encode the data which occurring more frequently.

The Huffman coding should be used after sampling and quantization because it requires a discrete set of values. Huffman coding is easy to implement and it will be optimal when the quantization regions (Δ) becomes very small. Here it is needed to use a prefilter with shaping followed by uniform quantizer. This is shown in the Figure 3.4.

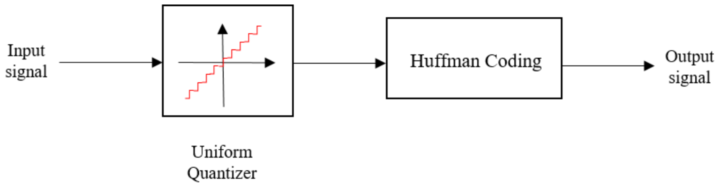

Figure 3.4: Uniform quantizer followed by huffman coding

3.4 Choosing a Quantization method

In this thesis, the uniform quantization method is preferred and it is one of the most commonly used scalar quantizer because of its simplicity. The input and output response of the uniform quantizer lies along a straight line with a unit slope. This makes the quantization easier and simpler. In uniform quantization, all intervals which are uniformly spaced along the x-axis are called the decision levels. Another level of the quantizer which is also uniformly spaced except the outer intervals called the reconstruction levels. All the intervals are of equal step size (Δ). The arithmetic average of the two decision levels of the corresponding interval along the x-axis called the inner reconstruction level. This mid-tread quantizer is chosen by the comparison between the mid-tread and mid-rise for example two bits and is shown in Figure 3.5 and 3.6 respectively.

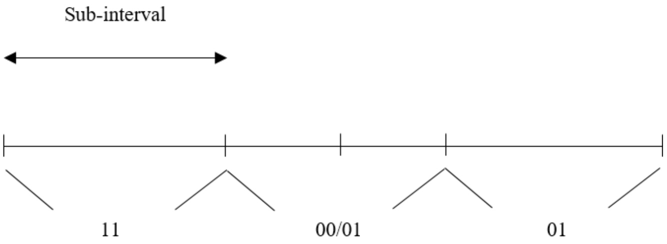

Figure 3.5: A 2-bit uniform mid-tread quantizer

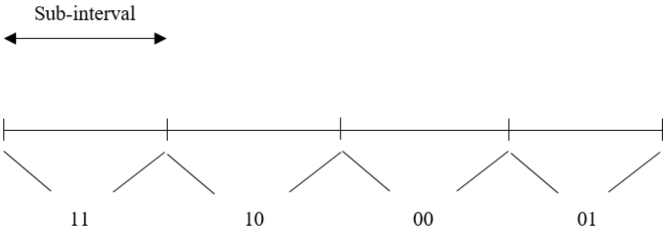

Figure 3.6: A 2-bit uniform mid-rise quantizer

Among the two types of uniform quantization, the Mid-tread quantizer is utilized for an odd number of reconstruction levels. This mid-tread quantizer is chosen by the comparison between the mid-tread and mid-rise for example two bits. In this comparison, the subinterval size of the mid-tread is larger than the mid-rise quantizer. The mid-tread quantizer gives better performance when representing signals with zero amplitude compared to the mid-rise. The quantization error of the mid-tread is larger than the mid-rise when the input signal is uniformly distributed. Due to these reasons, the mid-tread quantizer is preferred and used in this thesis.

3.5 Design of the uniform quantizer

As explained in Section 3.1, the first step in designing the quantizer is to find the quantization levels. According to the number of quantization levels, the desired resolution of the output can be determined. The number of quantization levels depends on the number of bits (R) used. The step size (Δ) value can be obtained from the formula below.

$$\Delta = \frac{max(input\ signal)}{\frac{L}{2} - 1}$$

where

$$L = 2^R$$

If the input signals have a wide dynamic range, then the input signals are bounded by a range value, which is between the $\left(-\frac{L}{2} - 1\right)$ to $\left(\frac{L}{2} - 1\right)$ and it will be in the multiples of the step size (Δ) value. i.e.,

$$\text{rangevalues} = \left(\left(-\frac{L}{2} - 1\right) \text{to} \left(\frac{L}{2} - 1\right)\right) \cdot \Delta$$

This is used to map the values. The values which are smaller than $\left(-\frac{L}{2} - 1\right) \cdot (\Delta)$ is mapped to $\left(-\frac{L}{2} - 1\right) \cdot (\Delta)$ and the values larger than $\left(\frac{L}{2} - 1\right) \cdot (\Delta)$ is mapped to $\left(\frac{L}{2} - 1\right) \cdot (\Delta)$. All the values inside this range are mapped according to this procedure to get the quantized values of input signal.

3.6 Quantization noise

The resulted signal after the quantization process has noise because it is mapping the same value for many input values (many to few mapping). So there is a need to separate noise from the output signal of the quantization process. By taking the difference between the original signal and the quantized signal, we can get the Quantization error, or Quantization noise. This quantization error is uniformly distributed for uniform quantization and the effect of the channel noise is reduced because of the system operates in an average signal power. This causes the performance limited especially by the quantization noise alone.

The quantization error is bipolar i.e. it can take both positive and negative values. In the uniform quantizers the quantization regions are constant which leads to a uniformly distributed quantization noise.

As an example, the quantization error for the first sample can be calculated as shown in the formula below.

$$Q_E(n) = y(n) - x(n)$$

where,

Q_E is the Quantization Error vector.
$y(n)$ is the Quantized sampled vector.
$x(n)$ is the Input's sampled vector.
n is the number of samples of the whole signal.

The LTE DL signal contains certain number of symbols and each symbol contains a certain number of subcarriers (samples) depends on the signal bandwidth.

The quantization error is calculated sample by sample for each symbol and the average for the whole LTE signal can be calculated as,

$$\mathbf{E}(Q_E(k)) = \mathbf{E}(y(k) - x(k))$$

where,
$k = 1, ..., N$.
N is the number of subcarriers.
$\mathbf{E}(Q_E(N))$ is the average Quantization Error vector.

The quantization noise level has been measured for different bandwidths but it is always in a constant level as shown in Figure 3.7. This is the reason behind the shaping of this noise. In the guard bands, there is no importance of the noise level so the shaping should be done by shaping the noise level in such a way the guard bands will contain as much noise as possible and the subcarrier band should have as less noise as possible. This can be defined as QNS which will be explained in the next chapter.

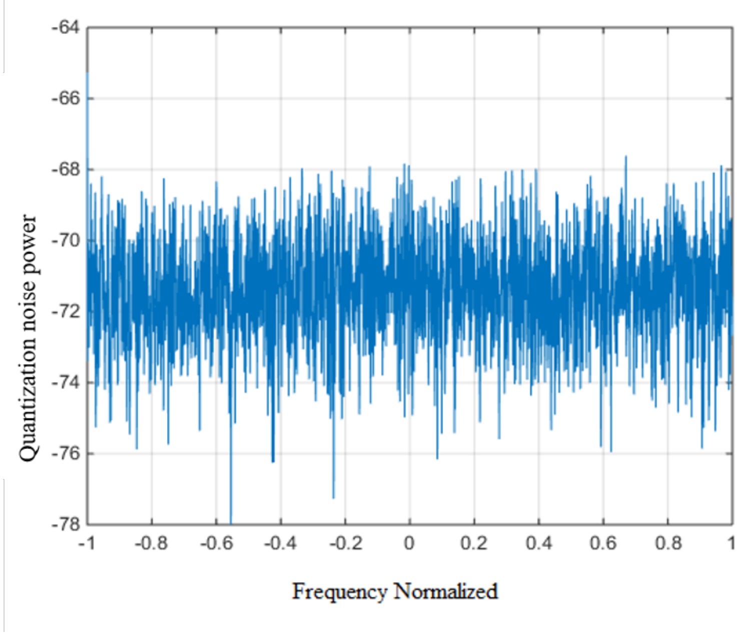

Figure 3.7: Power spectral density for quantization noise for 20 MHz

Chapter 4

Quantization Noise shaping and Oversampling

Noise shaping is a technique which is used to improve the compression done by the quantization. It mainly helps to increase the signal to noise ratio of the resulted signal. Quantization introduces distortion in the signal and the noise shaping changes the spectral characteristics of the error to improve the performances. These changes are achieved by using a noise shaping which is based on designing a feedback filtering system. The shaping which is needed to improve the compression, is done by lowering the noise in certain frequency bands and increasing the noise power at the remaining frequency bands. Noise shaping is used in some fields like digital image, audio and video processing. A way to improve the shaping is to oversample the signal before noise shaping. Oversampling is defined and discussed in this chapter.

4.1 Quantization Noise Shaping

QNS helps to increase the signal to noise ratio of the quantized signal. This helps to alters the spectral shape of the error caused by the quantization and to change the level of the noise power in frequency bands. Noise power is the power spectral density of the noise which can be calculated as follows:

$$N_p = \frac{|E|^2}{l}$$

where,
N_p is the noise power vector
$|E|$ is the absolute value of the error vector
l is the length of the error vector

Quantization error has lower noise power at higher frequency bands than lower frequency bands. This is because, it is not possible to lower the noise power in all frequencies with the help of noise shaping. In LTE DL system, it is required to decrease the noise level in the in-band and increasing the noise level in the guard bands.

To achieve the aim of this project, the design of the noise shaping system is used. The noise shaping method is used in many applications because it is a low cost system which gives high dynamic range and flexibility in converting the low band input signals. The filters in the noise shaping system helps to distributes the quantization noise to get the reduction of in-band noise level.

4.1.1 Overview of Noise shaping system

In this thesis, the concept of noise shaping system is used to shape the quantization noise. The noise shaping system is mainly aiming to get high transmission efficiency and the next thing is to shape the noise and moving most of the noise spectrum to higher frequencies, so the reduction of in band noise is achieved significantly. This concept is used to get higher signal to noise ratio (SNR) in a limited bandwidth.

The first and second order loop filter and 1 bit quantizer is used in the initial applications of this system. In this, if the high pass filter is used in forward path to the signal and also a high pass filter in feedback path to noise, then the noise shaping will be achieved. To apply the noise shaping in CRAN, a higher order loop filter and a multi bit quantizer is required. The noise shaping type used in this thesis is a single loop with 5 order filtering and 15 bit quantizer as shown in the Figure 4.1.

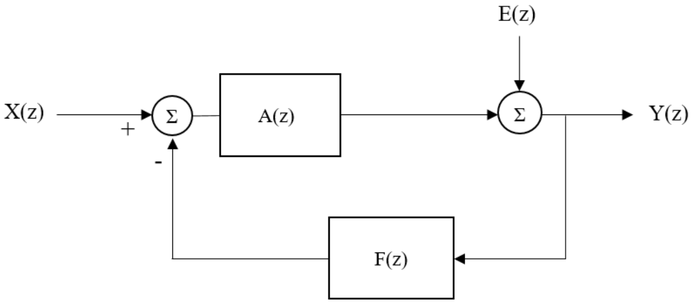

Figure 4.1: Single-loop Noise shaping system

The forward path filter $A(z)$ and the feedback path filter $F(z)$ are designed to shape the quantization noise $E(z)$. The signal is entering the system sample by sample. For each sample, it will be filtered by $A(z)$ and then quantized by the quantizer, which gives an error $E(z)$. The quantized sample with $E(z)$ will enter the $F(z)$ to be filtered and shaped and affecting the system again by minimizing the next sample. The derivation of the filtering shown in the figure can be given as formulas below:

$$Y(z) = (X(z) - Y(z)F(z)) \cdot A(z) + E(z)$$

$$Y(z) = X(z)A(z) - Y(z)A(z)F(z) + E(z)$$

$$Y(z) + Y(z)A(z)F(z) = X(z)A(z) + E(z)$$

$$Y(z) \cdot (1 + A(z)F(z)) = X(z)A(z) + E(z)$$

$$Y(z) = X(z) \cdot \frac{A(z)}{1 + A(z)F(z)} + E(z) \cdot \frac{1}{1 + A(z)F(z)}$$

Assuming that,

$$H_X(z) = \frac{A(z)}{1 + A(z)F(z)}$$

$$H_E(z) = \frac{1}{1 + A(z)F(z)}$$

so Y(z) becomes

$$Y(z) = H_X(z)X(z) + H_E(z)E(z)$$

$H_E(z)$ can be written as

$$H_E(z) = \frac{H_X(z)}{A(z)}$$

when $H_X(z) = 1$ then,

$$1 = \frac{A(z)}{1 + A(z)F(z)}$$

$$F(z) = 1 - \frac{1}{A(z)}$$

and $H_E(z)$ becomes

$$A(z) = \frac{1}{H_E(z)}$$

implementing $A(z)$ in $F(z)$ then,

$$F(z) = 1 - H_E(z)$$

4.1.2 Quantization noise shaping system

The QNS is done according to the concept of noise shaping which is described above in the Section 4.1.1. It is done commonly by putting the quantization error in the feedback loop. The feedback loop is a filter, so designing a proper filter for the error itself is considered as the biggest task in this thesis and this is explained briefly in Chapter 5. By this the error can be filtered as desired. Noise shaping always involve a certain amount of dither in the process so as to prevent the unwanted correlated errors. The adequate amount of dither usage is very important in the feedback loop. This helps to make the quantization error as a pure noise in which the process yields noise shaping properly.

The feedback filtering system is formed by using the concepts from noise shaping method. The forward path filter $A(z)$ and feedback filter $F(z)$ which are shown in the Figure 4.1 are implemented and designed to meet the aim of quantization noise shaping. The system which is followed in this thesis to shape the quantization noise is shown in the Figure 4.2.

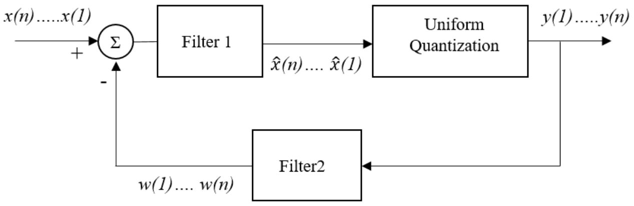

Figure 4.2: Quantization noise shaping. x denotes the input signal and y denotes the noise shaped output

4.2 Oversampling and noise shaping in LTE DL fronthaul

Oversampling can be defined as increasing the sampling rate for the discrete (digital) signal by repeating each sample a number of times. It is used in many applications such as Analog to Digital Converters (ADC) and Radar. In this thesis, oversampling helps to create a sharper cut off anti-aliasing in the guard bands. Oversampling with a low pass filter and down sampling could improve the shaping of the quantization noise. Usage of oversampling in a combination with the feedback noise shaping filtering system causes a better shape and could add further reduction in the number of bits for the quantizer.

4.2.1 Oversampling advantages and restrictions

Oversampling has many advantages, it increases the processing gain of the signal i.e. higher SNR and it can also handle with high bandwidths. In some applications like ADC, the high data rates caused by oversampling leads to many restrictions such as setup times, high power consumption's, marginal capturing (due to wrong setup time) and higher costs.

4.2.2 Oversampling to modify the shaping of the quantization noise

The quantization noise power density spectrum is constant over the entire bandwidth. In order to shape the quantization noise, the best way is to use a feedback noise shaping system for the oversampled signal. The basic concept of the shaping is to make the spectrum no longer frequency independent by increasing the noise in the guard bands and decreasing it in the carrier bands. To make a good reduction in the number of bits, a large oversampling ratio is required which is a high cost. However, this cost is much less if the oversampling is followed by the concept of quantization noise shaping. In this thesis the oversampling ratio is 2 which could not help in the bit reduction but helps to modify the shaping procedure. It is shown in Figure 4.3 that how the system looks like after adding the oversampling to the previous steps. The LPF used before down sampling is to reduce the higher levels of noise caused by the oversampling.

A comparison of the noise power levels between the signal before and after

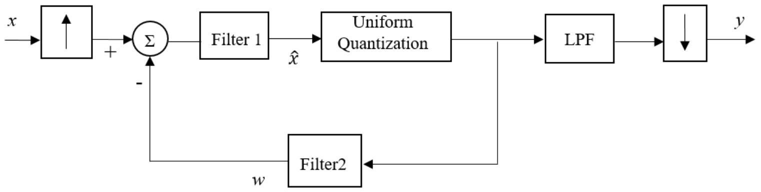

Figure 4.3: Oversampling and Quantization noise shaping where x denotes the input signal and y denotes the output

the down sampling is shown in the Figure 4.4 and 4.5. It is clearly shown the importance of LPF and the down sampling to decrease the noise caused by the oversampling process. Figure 4.4 shows the oversampling effects before downs ampling and Figure 4.5.

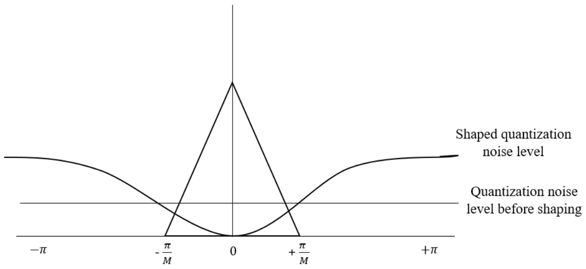

Figure 4.4: The power spectral density of the quantization noise levels and the signal before down sampling

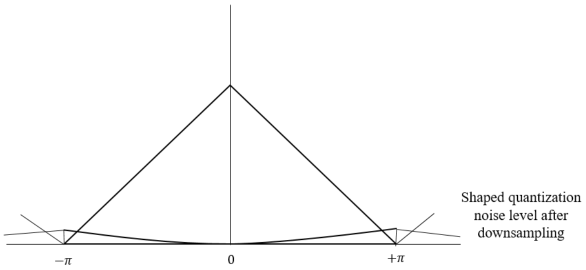

Figure 4.5: The power spectral density of the quantization noise and the signal after down sampling

The noise shaping strategy could be improved if the multi loop noise shaping system is used and if the oversampling ratio is increased. This way will give a better bit reductions result but increases the complexity of the system. A higher number of loops causes instability and oscillation. The multi loop system is shown in the Figure 4.6.

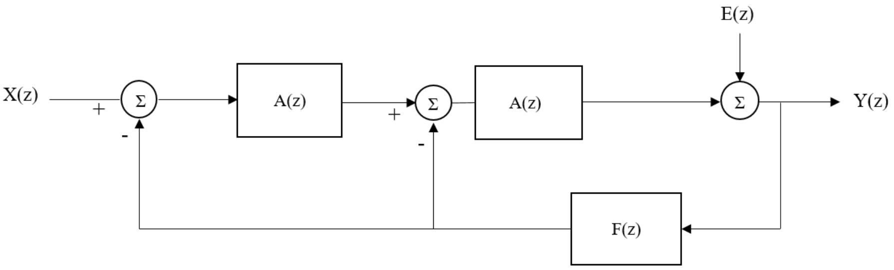

Figure 4.6: Multi loop Sigma-Delta modulator

Chapter 5

Implementation of the noise shaping system

In this chapter, the Implementation of the system will be explained. First step in building the system is to generate the LTE downlink signal to be quantized at the BBU. After quantizing the signal, filters are designed to shape the quantization noise and oversampling is implemented to improve the shaping of the quantization noise.

5.1 Testing of LTE generated signals

The signal Generation is done using the MATLAB LTE System Toolbox. In this thesis, the quantization and quantization noise shaping are tested for different bandwidths: 5 MHz, 10 MHz and 20 MHz and the modulation format used is 64QAM for all bandwidths [5]. The simulation is done also for different number of subframes.

`ltetestmodel` is a matlab function that allows to generate a complex baseband LTE DL signal. The following MATLAB code is showing how simply the MATLAB LTE System Toolbox provides the complex baseband signals to be tested. In this part of code, the number of subframes can be adjusted.

```
config = lteTestModel('2', '5MHz');
config.TotSubframes = 1;
[waveform, tmgrid, config] = lteTestModelTool(config);
```

This results in a complex baseband signal which includes CP. To prepare the signal for the next step, the CP should be removed. This gives a small compression for the signal before quantization. The MATLAB is used to design a uniform mid-tread quantizer and built step by step by us.

5.2 Design of the filters for quantization noise shaping

As a next step, the quantization noise should be shaped as described in Chapter 4 and for this, the filters are needed. The design of the QNS system is based on the noise shaping method. The designed noise shaping system used in this thesis is a single loop feedback noise shaping system with a 7 bits quantizer. The type of both feed forward and feed backward filters used in the system are fifth-order Chebyshev type-II filters are used. The properties of the Chebyshev type-II filter in terms of ripples are, it has no ripples in pass band and equiripple in stop band. This causes smoothness in the pass band, which leads to a greater advantage for choosing the Chebyshev type-II filter. The filter is designed based on the criteria described in Subsection 5.2.1.

5.2.1 Implementation of Chebyshev type-II filter

Chebyshev type-II filter is implemented using `cheby2` matlab function which returns filter coefficients in the vectors a and b. The filters can filter the input data using a rational transfer function which is defined by the numerator and denominator coefficients a and b respectively. This process is done in the Z-transform domain and called a rational transfer function. The following formula shows the rational transfer function:

$$Y(z) = \frac{b(1) + b(2)z^{-1} + \cdots + b(n_b + 1)z^{-n_b}}{1 + a(2)z^{-1} + \cdots + a(n_a + 1)z^{-n_a}} X(z)$$

Where,
n_a is the feedback filter order.
n_b is the feedforward filter order.

The rational transfer function can also be expressed as the following equation:

$$\begin{aligned} a(1)y(1) = b(1)x(n) + b(2)x(n-1) + \cdots + b(n_b + 1)x(n - n_b) \\ -a(2)y(n-1) - \cdots - a(n_a + 1)y(n - n_a) \end{aligned}$$

First Filter

The first filter is designed according to $A(z)$, which is explained in Chapter 4 in Section 4.1.1, where $H_E(z)$ relates to the filter coeffficients a and b which is,

$$H_E(z) = \frac{b}{a}$$

Now $A(z)$ becomes

$$A(z) = \frac{a}{b}$$

this means that the first filter $A(z)$ is the inverse filter of $H_E(z)$ and this can be represented in the Figure 5.1.

The equations below show the mathematical representation of the first filter.

$$\begin{aligned} y(m) &= (a(1)x(m) + z_1(m-1))/b(1) \\ z_1(m) &= a(2)x(m) + z_2(m-1) - b(2)y(m) \\ z_{n-2}(m) &= a(n-1)x(m) + z_{n-1}(m-1) - b(n-1)y(m) \\ z_{n-1}(m) &= a(n)x(m) - b(n)y(m) \end{aligned}$$

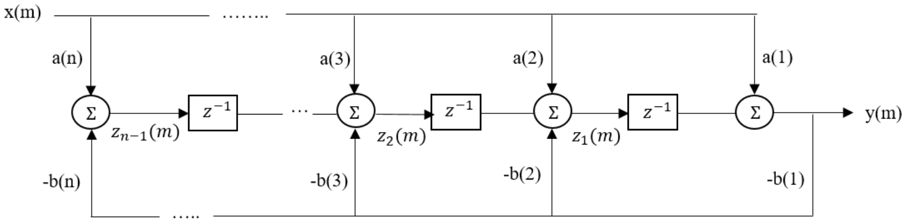

Figure 5.1: Design of first filter

Second Filter

The second filter is designed according to $F(z)$, which is explained in Chapter 4 in Section 4.1.1, where $H_E(z)$ relates to the filter coefficients a and b which is,

$$H_E(z) = \frac{b}{a}$$

Now $F(z)$ becomes

$$F(z) = \frac{a-b}{a}$$

This is shows how the filter coefficients looks like for the second filter $F(z)$ which can be represented in the Figure 5.2:

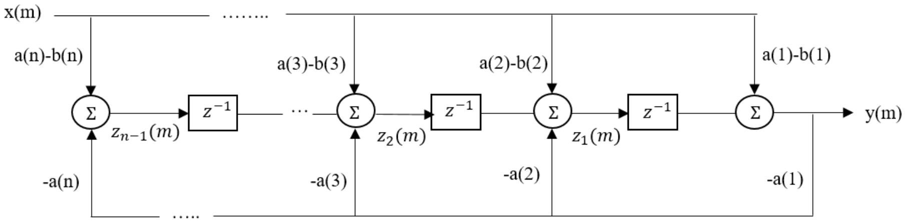

Figure 5.2: Design of second filter

The equations below show the mathematical representation of the second filter.

$$\begin{aligned} y(m) &= (a(1)-b(1))x(m) + z_1(m-1) \\ z_1(m) &= (a(2)-b(2))x(m) + z_2(m-1) - a(2)y(m) \\ z_{n-2}(m) &= (a(n-1)-b(n-1))x(m) + z_{n-1}(m-1) - a(n-1)y(m) \\ z_{n-1}(m) &= (a(n)-b(n))x(m) - a(n)y(m) \end{aligned}$$

This criteria of designing has been followed in the simulation. The filters had been built step by step to be used in the QNS system by implementing the derived formulas to filter the signal.

5.2.2 Implementation of Quantization noise shaping

As described in the Chapter 4 in Section 4.1.2, the quantization noise shaping is done sample by sample of the signal $x[n]$, where $x[n]$, $n = 1, \dots, N$ and N is the number of samples. For sample k, when $x[k]$ enters the system, it will be minimized from the output value of filter 2 $w[k-1]$. The result will be filtered by filter 1 and then quantized to give the output $y[k]$. The states of filters will be changed after filtering each sample. For the first sample, the output $y[1]$ is the quantized value of the output of filter 1 since the initial states of the filters are zero. This is shown in Figure 5.3.

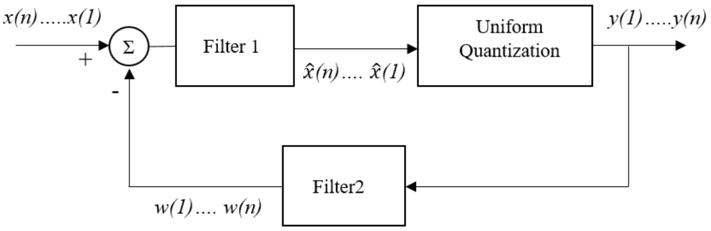

Figure 5.3: Quantization noise shaping

This process required an update in states of filters for each sample of the whole signal to give the final quantized and noise shaped output $y[n]$.

5.3 Implementation of oversampling

Oversampling is the process of increasing the sampling rate than the normal rate by a certain factor. After oversampling, the signal power and total quantization noise power will not change. Therefore, the SQNR is not changed. However, the quantization noise will spread over a larger frequency range.

The oversampling is implemented using the `resample` and `down sample` MATLAB functions to oversample the signal by a factor of 2. This leads to optimization in the shape occurred from QNS. The implementation of oversampling is done by oversampling the sampled signal by a factor of 2 and insert it to the QNS system. After the process of quantization and QNS is finished for the over sampled signal, the output must be filtered by a low pass filter to get rid of the high noise edges occurred by oversampling. The down sampling should be done by the same factor to get the original sampling rate. The Figure 5.4 illustrates how the system is been implemented and the low pass filter used is Chebychev type 2.

5.4 EVM in thesis work

As mentioned in Chapter 2, EVM is a standard way to evaluate the results and it can be checked in the DL transmission according to 3GPP standardization. In

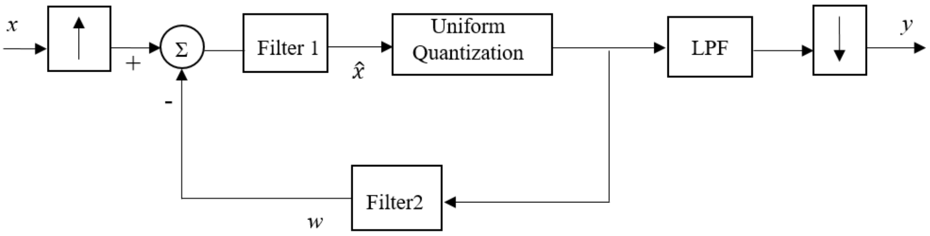

Figure 5.4: Oversampling and Quantization noise shaping

this thesis work, the EVM is founded for the quantization noise and the removed CP signal. EVM equation below describes how EVM is used in the thesis work.

$$EVM(\%) = \sqrt{\frac{P_{\text{error}}}{P_{\text{reference}}}} \cdot 100(\%)$$

Where, P_{error} is the quantization error and $P_{\text{reference}}$ is the power of removed CP signal.

In case of quantization and depending on the modulation form, the EVM percentage should not exceed a certain number in that position of LTE system as shown in Table 5.1 below, according to 3GPP standardization (TS36.104) in DL communication.

Table 5.1: EVM transmit specification

Modulation format	**Required EVM %**
QPSK	17.5 %
16 QAM	12.5 %
64 QAM	8 %

Another way to check the performance of the work is to check the Signal to Quantization Noise Ratio (SQNR) which shows the signal strength. SQNR is the ratio to the power of the signal and power of quantization noise in dB. It can be calculated by the following formula.

$$SQNR_{dB} = 10 \cdot \log_{10}\left(\frac{P_{\text{signal}}}{P_{\text{error}}}\right)$$

Where, P_{signal} is the power of the removed CP signal and P_{error} is the quantization noise power.

The importance of using EVM is that EVM can be calculated for both time and frequency domain but SQNR is calculated in time domain only. The EVM is evaluating the signal in active tones only and not for the whole signal. This gives more accurate evaluation of the signal compared to SQNR.

Chapter 6

Simulation and results

This chapter gives the final results of implementing the quantization, QNS and oversampling according to Chapter 5. The implementation is done to test different LTE DL signals generated by MATLAB LTE system toolbox and this process is carried out for frequency bands: 5, 10 and 20 MHz. The modulation format used for all frequencies is 64-QAM. All results described in this chapter are following the 3GPP standardization.

6.1 Simulation results for 10 MHz LTE DL signals

In this Section, the simulation is done for a 10 MHz LTE DL signal generated by MATLAB LTE System Toolbox. The simulation results are carried on three main steps (i.e. quantization, QNS and oversampling) and to compare the effect of each step on the signal, the EVM and SQNR values are calculated and discussed. Observe that the EVM and SQNR values are done for different number of subframes.

6.1.1 Quantization for 10 MHz

The generated sampled signal is to be quantized, so the first step is to remove the CP from the signal and then the signal will be ready for quantization process. The `quantiz` MATLAB function is used to quantize the signal. The absolute maximum and minimum values of the signal and the number of levels is needed to find quantization region (Δ). For 10 MHz, the quantization is successful until it reaches 6 bits i.e. no more reduction in number of bits. The EVM and SQNR for 1, 50, 105 subframes is shown in Table 6.1.

Table 6.1: EVM, SQNR values for 10 MHz after quantization (6 bits)

Subframes	EVM %	SQNR (dB)
1	6.4048	23.8699
50	6.3299	23.9720
105	6.3298	23.9722

6.1.2 Quantization Noise Shaping for 10MHz

The quantization noise shaping process is discussed in Chapter 5. As a result the quantization noise is shaped, according to our expectation. The EVM, SQNR values are improved and the reduction of one bit is occurred compared to quantization as shown in Table 6.2 below.

Table 6.2: EVM, SQNR values for 10 MHz after QNS (6 and 5 bits)

No. of bits	Subframes	EVM %	SQNR (dB)
6 bits	1	3.7668	28.4805
	50	3.7574	28.5023
	105	3.7588	28.4990
5 bits	1	6.8394	23.2997
	50	6.7799	23.3755
	105	6.7807	23.3745

6.1.3 Oversampling for 10MHz

To improve the results of shaping the quantization noise, the oversampling process is used. The `resample` and `down sample` MATLAB functions are used for oversampling the signal by a factor of 2. This leads to optimization in the shape occurred from QNS. The reduction in number of bits is not improved compared to QNS (i.e. 5 bits). This means, the maximum reduction are only one bit which is shown in the Table 6.3 below.

Table 6.3: EVM, SQNR values for 10MHz oversampling (6 and 5 bits)

No. of bits	Subframes	EVM %	SQNR (dB)
6 bits	1	4.2394	27.4533
	50	4.1927	27.5502
	105	4.1935	27.5484
5 bits	1	7.1847	22.8719
	50	7.1588	22.9031
	105	7.1602	22.9016

6.1.4 Comparison of QNS and Oversampling effects on 10 MHz LTE DL quantization noise spectrum

According to the EVM values of both QNS and Oversampling, it has been shown that the EVM values are a bit higher but we still have the same number of bit reduction. The Oversampling helps to shape the noise in the whole band.

In this Section, for 10 MHz, the graphs are plotted with reference to the power spectral density(PSD)in dB and normalized frequency of the quantization noise, QNS before oversampling and QNS after oversampling. The effect of QNS before oversampling is compared to the quantization noise and the effect of QNS after oversampling is compared to the quantization noise for different subframes is shown in the figures.

For 1 subframe, the comparison is shown in the Figure 6.1:

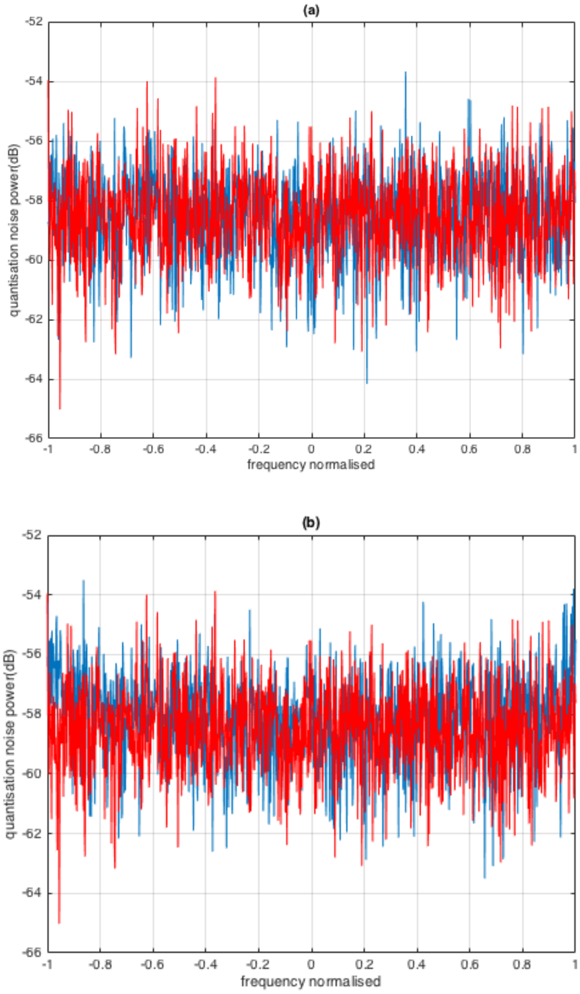

Figure 6.1: Comparison of (a) QNS before oversampling, (b)QNS after oversampling for 10 MHz LTE DL signal (1 subframe)

In the Figure 6.2, the signal generated contains 50 subframes. Logically, the result should be smoother and EVM value is better than 1 subframe as shown before.

For 50 subframe, the comparison is shown in the Figure 6.2:

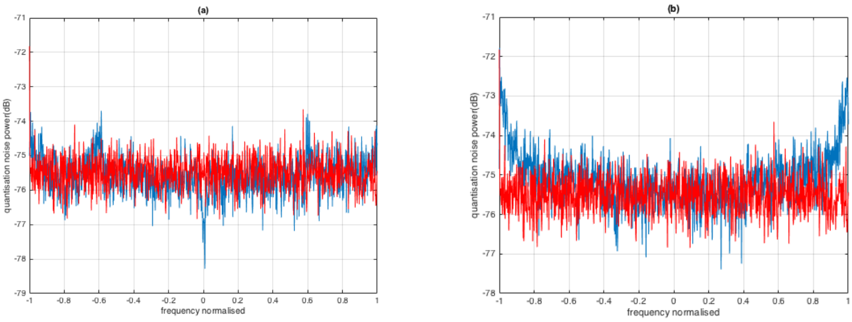

Figure 6.2: Comparison of (a) QNS before oversampling, (b)QNS after oversampling for 10 MHz LTE DL signal (50 subframe)

For further improvements, higher number of subframes is used, so that the shape becomes smoother and better for oversampling. It is clearly shown in the Figure 6.3 in case of using 105 subframes.

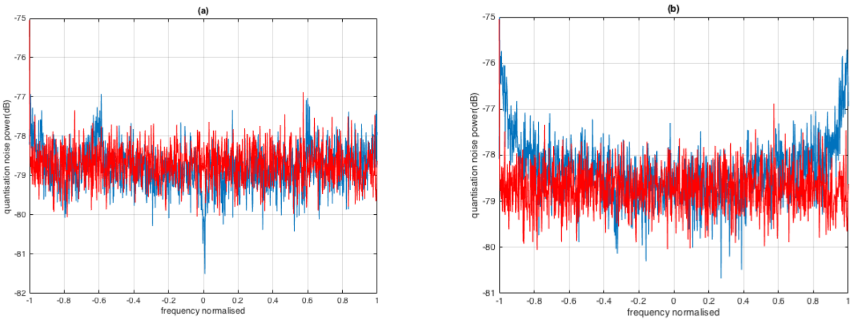

Figure 6.3: Comparison of (a) QNS before oversampling, (b)QNS after oversampling for 10 MHz LTE DL signal (105 subframe)

6.2 Simulation results for 5 MHz and 20 MHz LTE DL signal

In this section, the simulation is repeated for 5 and 20 MHz bandwidths for different number of subframes. The signals have been generated by MATLAB LTE system toolbox.

6.2.1 Quantization results

For both 5 MHz and 20 MHz, the quantization is successful until it reaches 7 bits i.e no more reduction in number of bits can be achieved. The EVM and SQNR for 1, 50, 105 subframes for both bandwidths is shown in Table 6.4.

Table 6.4: EVM, SQNR values for 5 MHz and 20 MHz after quantization (7 bits)

Bandwidth	Subframes	EVM %	SQNR (dB)
5 MHz	1	4.9031	26.1906
	50	6.2745	24.0485
	105	6.2669	24.0589
20 MHz	1	4.3260	27.2783
	50	4.3097	27.3110
	105	4.3096	27.3112

6.2.2 Quantization Noise Shaping Results

As a result the quantization noise is shaped according to our expectations. The EVM, SQNR values are improved and the reduction of one bit is occurred compared to quantization for both bandwidths as shown in Table 6.5.

Table 6.5: EVM, SQNR values for 5 MHz and 20 MHz after QNS

Bandwidth	No. of bits	Subframes	EVM %	SQNR (dB)
5 MHz	7 bits	1	2.8820	30.8060
		50	3.6262	28.8109
		105	3.6260	28.8115
	6 bits	1	5.3013	25.5123
		50	6.7212	23.4511
		105	6.7145	23.4593
20 MHz	7 bits	1	2.8623	30.8656
		50	2.3460	30.9184
		105	2.8467	30.9132
	6 bits	1	4.8447	26.2946
		50	4.8195	26.3400
		105	4.8205	26.3381

6.2.3 Oversampling results

For both 5 MHz and 20 MHz, the reduction in number of bits is not improved compared to QNS (i.e. 6 bits). This mean, the maximum reduction is only one bit. This is shown in the Table 6.6.

Table 6.6: EVM, SQNR values for 5 MHz and 20 MHz after oversampling

Bandwidth	No. of bits	Subframes	EVM %	SQNR (dB)
5 MHz	7 bits	1	4.2912	27.3485
		50	4.2459	27.4406
		105	4.2431	27.4463
	6 bits	1	6.1296	24.2513
		50	7.0761	23.0041
		105	7.0696	23.0121
20 MHz	7 bits	1	3.3576	29.4994
		50	3.3861	29.4059
		105	3.3869	29.4039
	6 bits	1	5.1302	25.7973
		50	5.1347	25.7896
		105	5.1352	25.7889

6.2.4 Comparison of QNS and Oversampling effects on 5 MHz and 20 MHz LTE DL quantization noise spectrum

As shown Section 6.1.4, the oversampling helps to improve the shapes of the guard bands more compare the in band. In this section the effects of oversampling will be introduced for 5 and 20 MHz.

For 5 MHz

For 1 subframe, the comparison is shown in the Figure 6.4:

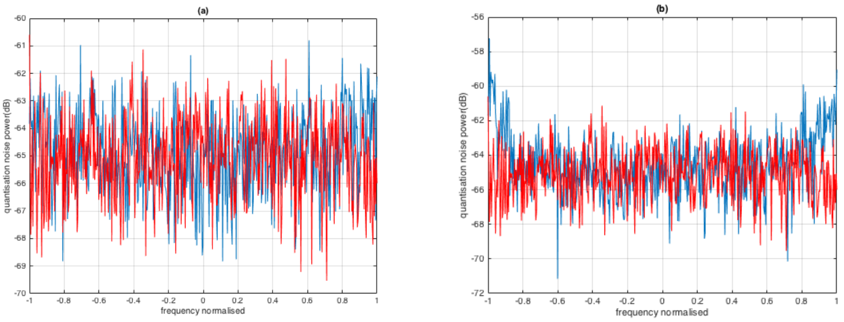

Figure 6.4: Comparison of (a) QNS before oversampling, (b)QNS after oversampling for 5 MHz LTE DL signal (1 subframe)

For 50 subframes, the comparison is shown in the Figure 6.5:

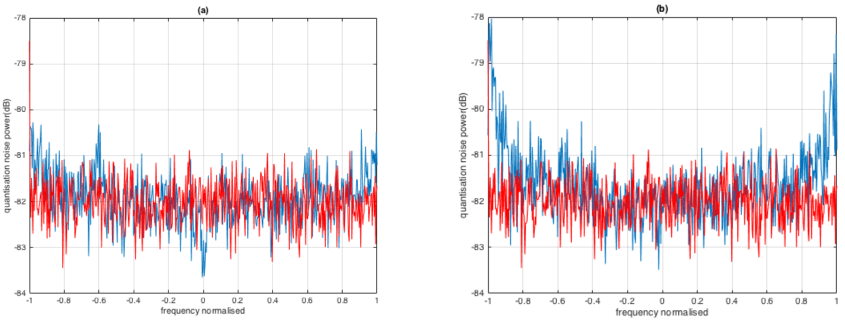

Figure 6.5: Comparison of (a) QNS before oversampling, (b)QNS after oversampling for 5 MHz LTE DL signal (50 subframe)

For 5 MHz, it is clear from the Figures 6.4 and 6.5, that oversampling improves the shape of quantization noise shaping compared to the quantization noise. For further improvements, higher number of subframes is used, so that the shape becomes smoother and better for oversampling. It is clearly shown in the Figure 6.6 in case of using 105 subframes.

For 105 subframes, the comparison is shown in the Figure 6.6:

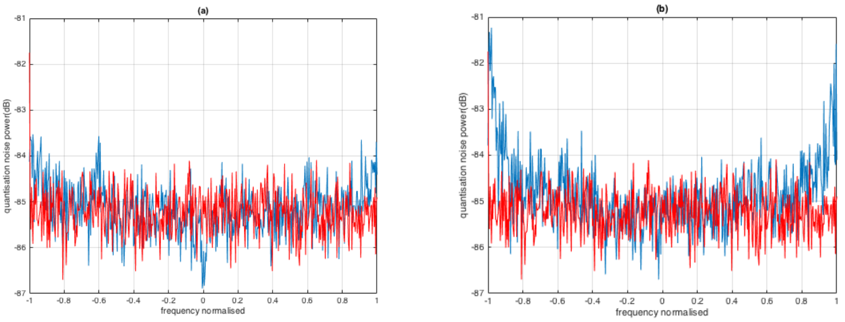

Figure 6.6: Comparison of (a) QNS before oversampling, (b)QNS after oversampling for 5 MHz LTE DL signal (105 subframe)

For 20 MHz

Now the comparison is done for 20 MHz frequency bandwidths for 1,50 and 105 subframes.

For 1 subframe, the comparison is shown in the Figure 6.7:

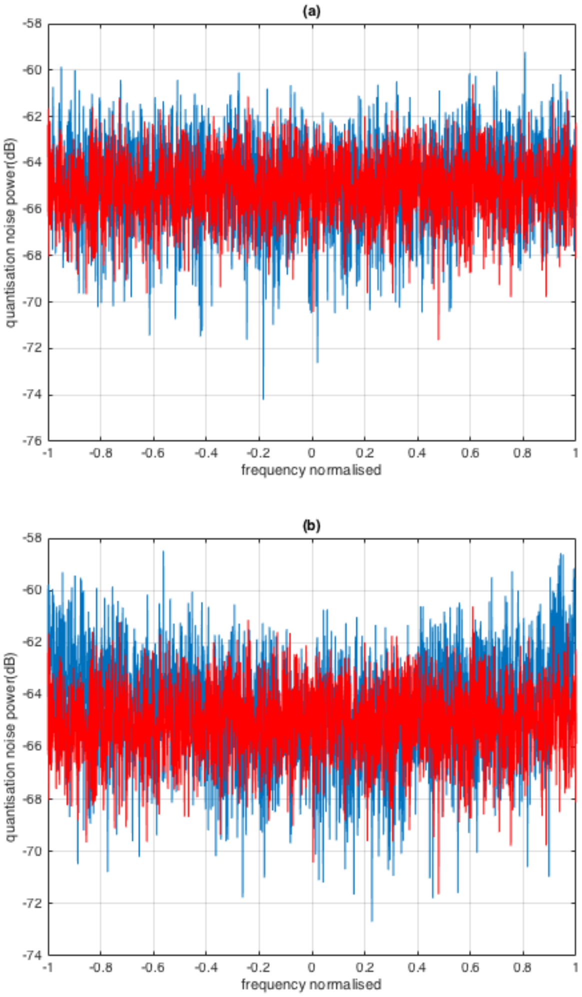

Figure 6.7: Comparison of (a) QNS before oversampling, (b)QNS after oversampling for 20 MHz LTE DL signal (1 subframe)

For 20 MHz, the Figure 6.8 shown for the 50 subframes and it is clear from the Figures 6.7 and 6.8, that oversampling improves the shape of quantization noise shaping compare to the quantization noise.

For 50 subframes,

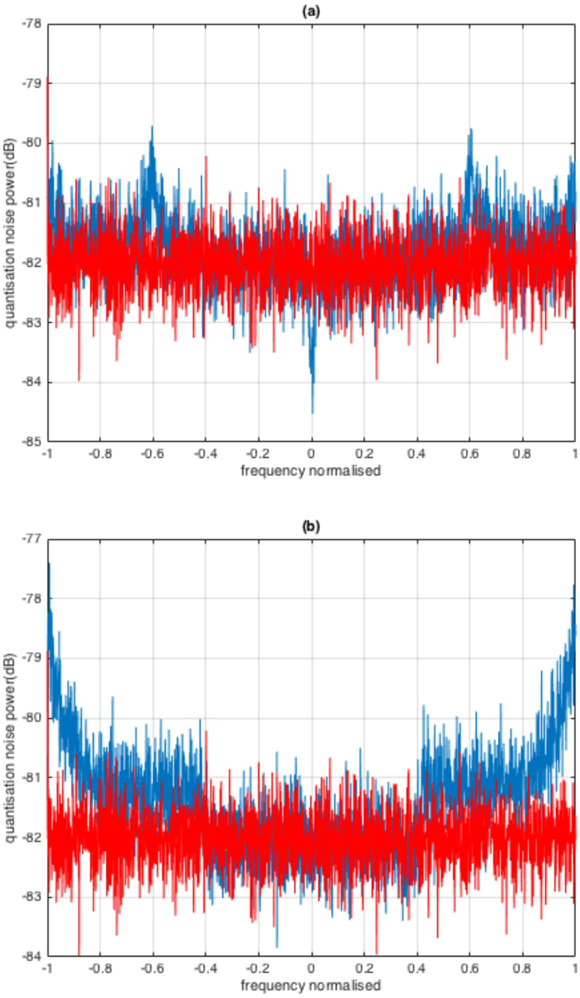

Figure 6.8: Comparison of (a) QNS before oversampling, (b)QNS after oversampling for 20 MHz LTE DL signal (50 subframe)

For further improvements, higher number of subframes is used, so that the shape becomes smoother and better for oversampling. It is clearly shown in the Figure 6.9 in case of using 105 subframes.

For 105 subframes,

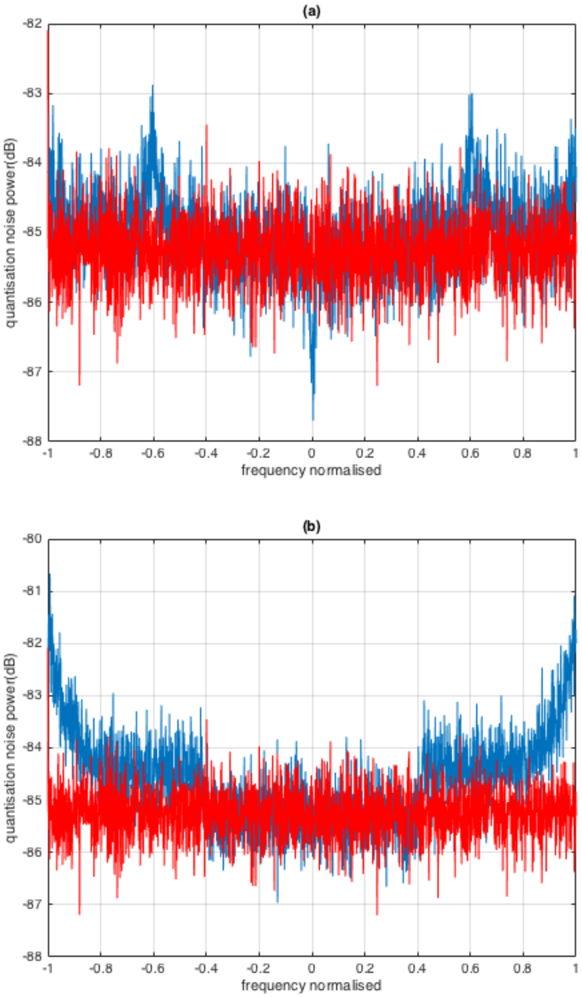

Figure 6.9: Comparison of (a) QNS before oversampling, (b)QNS after oversampling for 20 MHz LTE DL signal (105 subframe)

6.3 EVM Values for random number of subframes

In this section, we are going to observe how EVM values will be varying, in case of higher subframes. How EVM will vary? Is it decreasing constantly? As a trial for 10 MHz, we tried for many subframes in between 1 and 50, the EVM values are calculated and plotted, as shown in the Figure 6.10. It is observed that the variation differs up and down according to the number of subframes and EVM variation is lower for high number of subframes. This result leads to a constant EVM for very high number of subframes.

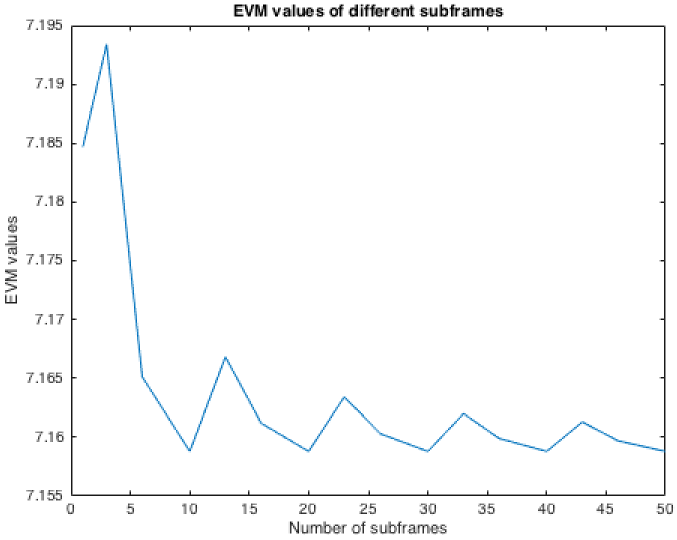

Figure 6.10: Comparison of EVM values for different subframes on 10 MHz LTE DL signal

The EVM values are shown in Table 6.7 for different number subframes for 10 MHz LTE DL signal after oversampling and for 5 bits. All the EVM values are almost closer to each value but still they have a minuet variation. This makes the up and down variation but this will get to reduce when the subframes are increased further.

Table 6.7: EVM values for 10 MHz for many number of subframes

Number of subframes	EVM value
1	7.1847
3	7.1934
6	7,1651
10	7.1588
13	7.1668
16	7.1612
20	7.1588
23	7.1634
26	7.1603
30	7.1588
33	7.1620
36	7.1599
40	7.1588
43	7.1613
46	7.1597
50	7.1588

6.4 Discussion of the results

Compression algorithm has been tested using MATLAB LTE System Toolbox. The metrics used to evaluate the compression results are the EVM and SQNR which is shown in Sections 6.1 and 6.2. These results show a good compression in the data traffic in the LTE DL signal to meet the aim of this thesis work.

As explained in Section 6.3, the results of compression are better if the number of subframes is increased. For 105 subframes, a brief discussion of the final results will be explained in this section. For 10 MHz, the maximum bit reduction achieved is 5 bits and the following Table 6.8 shows the EVM and SQNR values for 10 MHz before and after noise shaping.

Table 6.8: EVM, SQNR values for 10 MHz for 6 bits quantizer

Process	EVM %	SQNR (dB)
Quantization	6.3298	23.9722
QNS with Oversampling	4.1935	27.5484

The noise shaping for oversampled 10 MHz DL signals improves the EVM percentage by 2.14% and the SQNR is improved by 3.5 dB. Furthermore, one more bit reduction can be achieved i.e. 5-bit compression. This gives a high EVM value (7.1602%) and lower SQNR (22.9016) which is still under 3GPP standardization but requires an optimization in other parts of the LTE system such as RRU and

UE.

For 5 and 20 MHz, the maximum bit reduction achieved is 6 bits and a comparison between EVM and SQNR values before and after noise shaping is shown in the Table 6.9 and 6.10 respectively.

Table 6.9: EVM, SQNR values for 5 MHz for 7 bits quantizer

Process	EVM %	SQNR (dB)
Quantization	6.2669	24.0588
QNS with Oversampling	4.2431	27.4463

From the Table 6.9, an improvement of 2% is achieved and SQNR has improved by 2.6 dB. For one more bit compression i.e. 6 bit, which gives an EVM value of 7.0696% and SQNR value of 23.0121 which could be accepted by 3GPP standardization but still has some limitations as explained before for 10 MHz.

Table 6.10: EVM, SQNR values for 20 MHz for 7 bits quantizer

Process	EVM %	SQNR (dB)
Quantization	4.3096	27.3112
QNS with Oversampling	3.3869	29.4039

For all bandwidths included in this thesis, the quantization noise shaping has improved the results of EVM and SQNR. The compression algorithm introduced to this thesis work can solve the problem of the upcoming huge data traffic. It gives a minimum EVM value and a good compression rate. For current quantization at the BBU, the compression is 30 for the complex baseband signal but here, the compression rate achieved helps the BBU to reduce the data word length before entering the CPRI links. The compression based on the noise shaping is a low-cost efficient and a simple algorithm. This result may be improved and easily applied on LTE and LTE-A.

Chapter 7

Conclusion

In this thesis work, the aim of the project was to compress the data rates at the BBU. This is important because of the upcoming challenges which will happen to the next generation. The 5_th generation requires multiple antenna systems and much higher data rate. It can be achieved by using a combination of uniform quantizer and QNS system. This will provide a reduction in the number of bits for the quantizer at the BBU and still meet the requirements of the EVM value at the RRU.

To test the performance of the QNS system, signals with different bandwidths and different number of subframes are applied. First, the quantizer has to be set at the minimum number of bits and an acceptable EVM value. QNS system is applied to get more reduction in the number of bits compared to using quantizer only. The usage of oversampling improves the noise shaping.

As mentioned in Chapter 6, we discussed the results of applying the system for 5, 10 and 20 MHz LTE bandwidths and for different number of subframes. This has been done for two cases which are QNS with oversampling and QNS without oversampling. The bit reduction caused by the both cases is same but the shape due to oversampling is improved. The best results achieved were for the 10 MHz although that the quantization step (which is fixed for all bandwidths and depending on the range of 5 MHz signal) is not fitting the range of 10 MHz compared to 5 and 20 MHz. The Table 7.1 shows the EVM values for 5, 10 and 20 MHz, after oversampling and for 105 subframes (6 bits).

Table 7.1: EVM for 5, 10 and 20 MHz for 6 bits quantizer and 105 subframes

Band width	**EVM** (%)
5 MHz	7.0696
10 MHz	4.1935
20 MHz	5.1352

The EVM values for 10 MHz are noticed to be better than those for 5 and 20 MHz in all cases so the EVM values for 10 MHz is calculated for random number of subframes to check its variations along the subframes. It is founded that the EVM values are changing to get narrower and we expect that for a very high number of subframes the EVM will not be affected.

This thesis work is a low cost way to achieve the aim of compression. Applying of QNS system may give a good compression rate for the current LTE systems with acceptable values of EVM at the BBU. The bit reduction of the LTE DL complex baseband signal is achieving 14 bits instead of 30 bits and could achieve 10 bits for 10 MHz.

Future work

This compression method will be widely used in the LTE fronthaul. For more number of antennas and for the uplink applications a furthermore bit reductions is required. This implies an improvement in QNS method and to find solutions for restrictions faced in this thesis. As a future work, this thesis work could be carried on by using nonuniform quantizer and multi loop QNS system. We are expecting that this may give a more bit reduction.

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