NTNU – Trondheim
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QoS performance of LTE networks with network coding

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Master of Telematics - Communication Networks and Networked Services (2
Submission date: June 2015
Supervisor: Yuming Jiang, ITEM

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Problem description:

This master's thesis is a partial fulfillment of the international master degree program in the department of Telematics, IME faculty, 30.0 credits.

3GPP Long Term Evolution (LTE) is a promising technique and standard for fourth generation (4G) mobile wireless communication systems. The increasing attention on LTE, together with the shift in mobile consumer habits from voice calls only to IP-based data services such as web browsing, video streaming, video conferencing and social networking, are causing a growing interest in the quality of service (QoS) performance of LTE networks. As the demand for massive multimedia delivery over fourth generation wireless cellular standards such the LTE increases, the need for reliable and flexible transmission paradigm over the LTE networks becomes equally important. Network coding is one of the promising solution for reliable multimedia delivery over wireless networks.

The overall objective of this project is to study the architectural background and to evaluate the QoS performance of LTE networks. The evaluation method to be used is simulation. Specifically, the ns3 network simulator will be adopted. In addition, another objective of the project is to investigate the effect of network coding on the QoS performance of LTE networks. For this purpose, it is intended to use current functionality of KODO library to enable transport of coded packets across the LTE network.

Responsible professor: Yuming Jiang, ITEM, NTNU
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Abstract

Nowadays the widespread use of variety of smart phones and tablets with wide range of multimedia application support is driving more data service users towards full mobility causing a rapid increase in demand for mobile data rates. These new devices and multimedia applications require high data rates and reduced latency to provide better Quality of Service (QoS). To address these requirements the 3rd Generation Partnership Project (3GPP) introduces Long-Term Evolution (LTE) with a capability to move towards Fourth Generation (4G) wireless systems. It is designed to be a high data rate and low latency system that aiming to support different types of services. As the demand for massive multimedia delivery over LTE network increases, a novel transmission techniques such as Network Coding (NC) are needed.

In this thesis work we present the QoS performance analysis of downlink LTE using an open source simulation libraries, Network Simulator-3 (ns-3) and Kodo. The main performance parameters considered are the throughput, packet delay, spectral efficiency, capacity and coverage. Factors affecting the overall performance such as the fading, shadowing, buildings, User Equipment (UE) speed, UE-Evolved Node B (eNB) distance and traffic load are considered. The scenario used for the analysis includes multiple UEs and eNBs for different system antennas and system bandwidths. Moreover, Random Linear Network Coding (RLNC) coding scheme is implemented in LTE networks for a simple scenario composed of a single UE, eNB and remote host to assess the usefulness of NC.

The results obtained shows the impact of different factors on the system QoS performance. The throughput, delay, spectral efficiency, capacity and coverage performances are evaluated and discussed for different system bandwidth and different system antennas with varying transmission power. In addition, network coding has been shown to improve the throughput at a cost of higher packet delay. Moreover, alternatives ways of improving the throughput and different variants of NC are discussed. Since the results are based on both theory and experiments, the analysis and discussions made could be considered as a start point in dimensioning an LTE commercial networks. Suggestions for future work and a draft of a conference paper are also given.

Acknowledgment

This work would not be possible without help from the following people, and I wish to express my sincere thanks to Katina Kralevska for her continuous supervision, advice and knowledge she has shared with me through out the work. I am also grateful to Yuming Jiang, professor. I am thankful and indebted to him for sharing expertise,sincere and valuable guidance and encouragement extended to me. I would like to thank Steinwurf-Kodo and ns-3 community for sharing their experience and knowledge. I also thank my parents and friends for the unceasing encouragement, support and attention.

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List of Acronyms

2G Second Generation.

3G Third Generation.

3GPP 3rd Generation Partnership Project.

4G Fourth Generation.

ACK Acknowledgment.

AL-RLNC Application Layer-RLNC.

AMC Adaptive Modulation and Coding.

API Application Program Interface.

ARP Allocation and Retention Priority.

ARQ Automatic Repeat Request.

COST European Union Forum for cooperative scientific research.

CP Cyclic Prefix.

CQI Channel and Quality Indication.

CRC Cyclic Redundancy Check.

DFT Discrete Fourier Transformer.

EARFCN E-UTRAN Absolute Radio Frequency Channel Number.

eNB Evolved Node B.

EPC Evolved Packet Core.

EPS Evolved Packet System.

E-UTRAN Evolved Universal Terrestrial Radio Access Network.

FDD Frequency Division Duplexing.

FEC Forward Error Correcting.

GBR Guaranteed Bit Rate.

GERAN GSM EDGE Radio Access Network.

GPRS General Packet Radio Service.

GSM Global System for Mobile Communications.

GTP GPRS Tunneling Protocol.

HARQ Hybrid Automatic Repeat Request.

HSS Home Subscriber Server.

IP Internet Protocol.

ITU International Telecommunication Union.

LTE Long-Term Evolution.

MAC Media Access Control.

MIMO Multiple Input- Multiple Output.

MME Mobility Management Entry.

NACK Negative-Acknowledgment.

NAS Non Access Stratum.

NC Network Coding.

NGBR Non-Guaranteed Bit Rate.

ns-3 Network Simulator-3.

OFDM Orthogonal Frequency Division Multiplexing.

OFDMA Orthogonal Frequency Division Multiple Access.

PAPR Peak to Average Power Ratio.

PCRF Policy Control and Charging Function.

PDCP Packet Data Convergence Protocol.

PDN Packet Data Network.

PDN-GW Packet Data Network Gateway.

PDU Packet Data Unit.

PHY Physical Layer.

PSS Priority Set Scheduler.

QAM Quadrature Amplitude Modulation.

QCI QoS Class Identifier.

QoS Quality of Service.

QPSK Quadrature Phase Shift Keying.

RAN Radio Access Network.

RB Resource Block.

RLC Radio Link Control.

RLNC Random Linear Network Coding.

RRC Radio Resource Control.

RTT Round Transmit Time.

SAE System Architecture Evolution.

SC-OFDM Single Carrier-OFDM.

SGSN Serving GPRS Support Node.

SGW Serving Gateway.

SINR Signal to Interference and Noise Ratio.

SISO Single-Input Single-Output system.

TB Transport Block.

TCP Transmission Control Protocol.

TDD Time division Duplexing.

TDMA Time Division Multiple Access.

TEID Tunnel End ID.

TFT Traffic Flow Templates.

TTI Transmission Time Interval.

UDP User Datagram Protocol.

UE User Equipment.

UML Unified Modeling Language.

UMTS Universal Mobile Telecommunications System.

WCDMA Wideband Code Division Multiple Access.

WiMAX Worldwide Interoperability for Microwave Access.

Chapter 1
Introduction

1.1 Motivation

The rapid increase of mobile data usage and emergence of new applications such as online gaming, mobile TV and streaming contents have motivated the 3GPP to work on the LTE on the way towards 4G mobile networks. The need to ensure the continuity of competitiveness of the Third Generation (3G) system in the future, the user demand for higher data rates and quality of service and continued demand for cost reduction are also some of the motivations.

According to global mobile data traffic forecast by Cisco in [Cis], there will be 5.2 billion global mobile users, 11.5 billion mobile-ready devices and connection. An average mobile connection speed demand will increase by 2.4 fold and the global mobile Internet Protocol (IP) traffic will reach an annual run rate of 292 exabytes by 2019. In addition, IP video will represent 79 percent of all traffic by 2018, up from 66 percent in 2013. Second Generation (2G)/Global System for Mobile Communications (GSM) and 3G/Universal Mobile Telecommunications System (UMTS) has been the key mobile communication technologies, chosen by more than 2 billion people around the world. In order to adapt to new services, increasing demand for user bandwidth, quality of service and requirements for network convergence, major evolutions are introduced in 3G network standard by the 3GPP.

The new 4G/LTE technology supports scalable carrier bandwidths, from 1.4 MHz to 20 MHz and supports both Frequency Division Duplexing (FDD) and Time division Duplexing (TDD). It has the ability to manage fast-moving mobiles and supports multicast and broadcast streams. Moreover, it is based on Orthogonal Frequency Division Multiplexing (OFDM) in combination with higher order modulation, large bandwidths and spatial multiplexing in the downlink to provide downlink peak rate of 300 Mbps, uplink peak rate of 75 Mbps and QoS provisions permitting a transfer latency of less than 5 ms in the Radio Access Network (RAN). In addition, to the new access network solution, 3GPP also specifies the IP-based network architecture called the Evolved Packet Core (EPC). It is designed to replace the General Packet Radio Service (GPRS) core network, supports seamless handovers for both voice and data to cell towers with older network technology such as GSM and UMTS. This simple all IP architecture results in lower operating costs and compatibility with the previous generation networks [3gp].

Moreover, a reliable data transmission is an important factor to achieve the 3GPP specified data rates and QoS. LTE uses state of the art Hybrid Automatic Repeat Request (HARQ) to ensure data is sent reliably between network nodes. In recent years, a promising reliable data transmission paradigm called NC has been deployed in many wireless networks. It has been shown that it can effectively improve the efficiency and capacity of wireless networks by exploiting the broadcast nature of the wireless medium.

Based on these motivations, we believe that combining the features of LTE and NC will enable network operators to deploy a mobile communication networks with high capacity, spectral efficiency and data rate, low network latency with simple QoS provisioning techniques. In general, these networks can meet the growing demand of IP traffic with low deployment cost and complexity for both users and network operators. Thus, analysing the performance of LTE networks with different reliable transmission schemes is an important step towards this goal.

Furthermore, this work is inspired by the rapid increment in the use of smart phones, tablets and social medias and the fact that in order to deliver service to these devices and applications a very dependable and efficient mobile wireless network, such as the LTE, is a requirement. For this purpose we conduct a theoretical and experimental analysis to determine the performance of LTE networks with different performance metrics such as throughput, delay, spectral efficiency, capacity and coverage. We have evaluated the QoS performance of a defined LTE network and discuss the practical meaning of the experimentally obtained results. The analysis and discussions can be used to foresee the performance of an LTE/4G network before deploying them.

Moreover, to the author's knowledge there are no many practical works related to deployment of NC in LTE networks. Thus, in this thesis work we have implemented a NC scheme to enhance the performance of LTE networks. As a result, we have prepared a conference paper draft and is attached to this work so as to provide a running start in integration of NC into LTE networks.

1.2 Objective and Methodology

A thorough performance evaluation of mobile communication networks is of a significant importance for network operators. Therefore, we prepare a comprehensive study and evaluation of the different performance metrics of LTE networks. Thus, the overall objective of this thesis work is to study the architectural background of LTE networks thoroughly, conduct an experiment on LTE networks and analyse the QoS performance of the RAN. Another objective is to introduce the NC concept, conduct an experiment on a defined topology and analyse the obtained results.

To achieve these objectives, a methodology based on open source simulation libraries, namely ns-3 and Kodo is used. These libraries enables us to build an experimental LTE network topology, simulate it and collect the required results for analysis and evaluation of QoS performance in LTE network. QoS performance in terms of system throughput, delay, spectral efficiency, coverage and capacity is evaluated and discussed. While most of the simulation parameters are fixed throughout the simulation, some of the performance evaluation require a change of some of the parameters.

1.3 Thesis' Structure

The thesis consists of eight chapters and three appendixes. Chapter 1 gives a brief introduction to the thesis' motivation, objective and methodology. Chapter 2 includes the theoretical background on protocol stack, reference points and interfaces of LTE and EPC. It also includes the QoS provisioning and enforcement techniques and the main features of 4G networks in comparison with 2G and 3G. Chapter 3 describes the concept of NC and its integration into LTE networks including two different alternatives for doing this. Chapter 4 describes the simulation libraries used in this work, ns-3 and Kodo. Chapter 5 gives details about the experimental network topology, simulation parameters and analysis of the results obtained regarding QoS performance in the access network. Chapter 6 includes the integration of NC into LTE networks, the topology and architecture of the network used and the results are analysed and discussed. Chapter 7 discusses the results obtained in chapter 5 and chapter 6. Connclusion and future works are presented in chapter 8. In addition, there are three appendixes: a table that shows the standard LTE bands, a Unified Modeling Language (UML) class diagram that shows the structure of ns-3 modules and a conference paper draft tittled *Performance of LTE networks with RLNC.*

Chapter 2

Theoretical Background

This chapter explains the theoretical background of the thesis work. It discusses the migration of 2G and 3G networks towards the 4G network, the main entities in LTE and EPC, protocols, interfaces and reference points. In addition, the main features of LTE and QoS provisioning and enforcement and others concepts that are used in later part of the thesis are explained.

2.1 Architectural Study

2.1.1 Migration towards Evolved Packet System

The Evolved Packet System (EPS), also referred as System Architecture Evolution (SAE), is purely IP based core network architecture of 3GPP's LTE wireless communication standard. In EPS deployment radio access technologies such as LTE, are a primary consideration because they directly affects the mobile operators' most valued asset: spectrum. Other equally important aspect to consider is the multimedia core network that will play a central role in simplifying the migration from 2G/3G to 4G [lon][3gp].

In all the wireless network technologies prior to 4G, the circuit switching and packet switching dual-domain concept is kept on the core and access networks. This means that circuits are established between calling and called parties throughout the telecommunication networks. In GSM, all services are transported over circuit-switches, in UMTS and GPRS data is transported in packets without the establishment of dedicated circuits.

The first step towards an IP based packet switched solution was taken with the evolution of GSM to GPRS, using the same air interface and access method, Time Division Multiple Access (TDMA). A further step was taken in UMTS where a new access technology Wideband Code Division Multiple Access (WCDMA) was developed but allocating the IP address to UE for data services is still dependent on the circuit switched core for paging as the previous access technologies. The LTE 4G adopt the "Always-ON" concept whereby an IP address is allocated to UE when the power to the terminal is ON, the address is kept so that the service-provision side can provide IP services same as in fixed communications network.

When designing the evolution of the 3G system, the 3GPP community decided to use IP as the key protocol to transport all services. The new EPS architecture, shown in Figure 2.1, comprises of the EPC as the core network of the LTE wireless communication standard with support for mobility between multiple heterogeneous access networks including Evolved Universal Terrestrial Radio Access Network (E-UTRAN), 3GPP legacy systems such as GSM EDGE Radio Access Network (GERAN), but also non-3GPP systems such as Worldwide Interoperability for Microwave Access (WiMAX). In addition EPS would not have a circuit-switched domain and the EPC should be the evolution of the packet-switched architecture used in GPRS/UMTS. This decision had consequences not only on the architecture itself but also on the way that the services were provided. Traditional use of circuits to carry voice and short messages needed to be replaced by IP-based solutions in the long term [3gp].

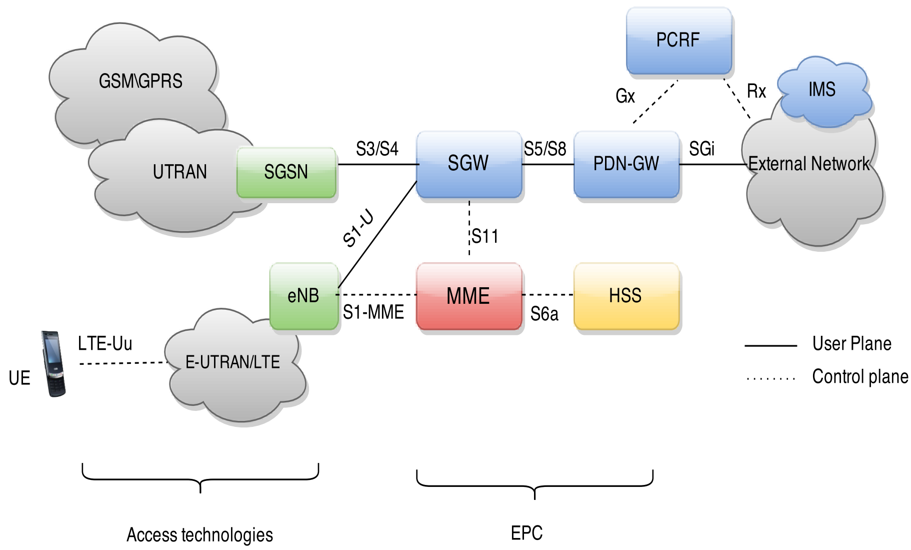

Figure 2.1: Evolved Packet System Architecture: Entities and Interfaces

Functional description of EPS components

The following list describes the main functional components of EPS network [lteg] [EUb].

– User Equipment (UE)

  UE is the mobile equipment which has similar Internet architecture to the one used by UMTS and GSM. It could also be any device used by end user.

– eNB

  eNB is a base station that controls the UEs in one or more cells and handles the radio communication between the UE and the EPC.The UE communicates with just one eNB and one cell at a time. The eNB supports two main functions: sends and receives radio transmissions to all the UEs using LTE air interface, controls the low-level operation of all its mobiles by sending them signalling messages such as the handover commands.

– Home Subscriber Server (HSS)

  HSS is a database that contains user-related and subscriber-related information. It also provides support functions in mobility management, call and session setup, user authentication and access authorization.

– Serving Gateway (SGW)

  SGW is the point of interconnect between the radio-side and the EPC. This gateway serves the UE by routing the incoming and outgoing IP packets and is the anchor point for the intra-LTE mobility and between LTE and other 3GPP accesses.

– Packet Data Network Gateway (PDN-GW)

  The PDN-GW is the point of interconnect between the EPC and the external IP packet data network, it is logically connected to SGW. The PDN-GW performs the routing packets to and from the external network and IP address/prefix allocation.

– Policy Control and Charging Function (PCRF)

  PCRF is an entity that interfaces with the main packet gateway and takes control of charging policy and enforcement decisions.

– Mobility Management Entry (MME)

  MME handles the signalling related to mobility and security for E-UTRAN access.

Figure 2.2 shows the functional split between the LTE/E-UTRAN and the EPC components of EPS architecture.

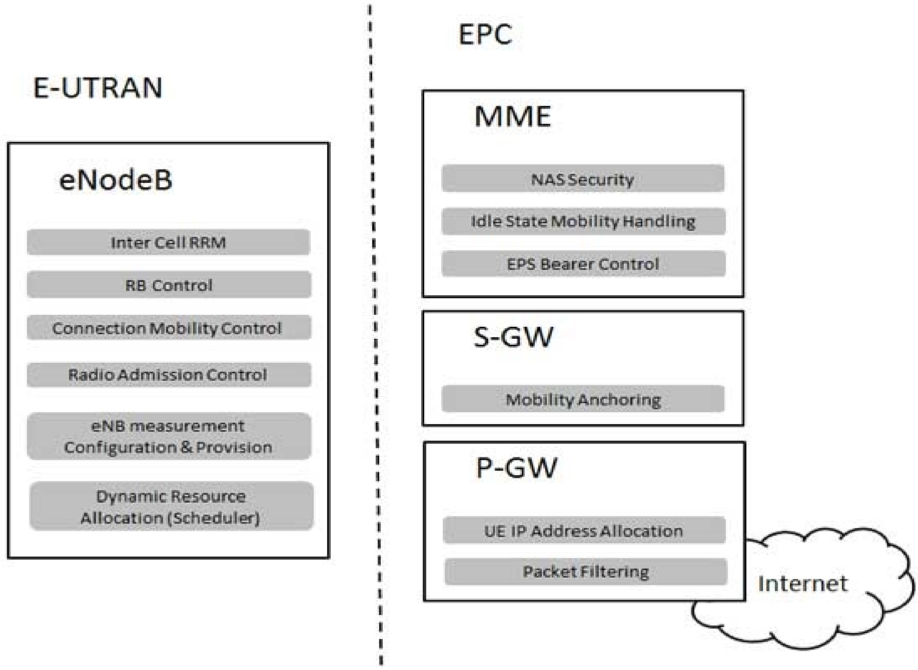

Figure 2.2: Functional Split of EPS Entities [lteg]

2.1.2 LTE Reference Points

The required LTE features are decomposed into functional entities without specific implementation assumptions about physical network entities. Figure 2.1 shows the logical representation of the network architecture, i.e. the reference model. It identifies the functional entities in the architecture and the reference points between the functional entities over which interoperability is achieved [Bar][EUb][ltec].

Table 2.1 show the most important reference points in LTE, these are defined as a conceptual link that connects two groups of functions that reside in different functional entities of the E-UTRAN and EPC.

Table 2.1: LTE Reference Points

Reference Point	End Point	Description
LTE-Uu	eNB and UE	An interface for the control and user planes between a UE and an eNB.
X2	eNB	An interface for the control and user plane between two eNBs. It is used during handover between two eNBs for self organizing networks
S1-U	E-UTRAN and SGW	An interface for the user plane between E-UTRAN(eNB) and an SGW. It provides a GPRS Tunneling Protocol (GTP) tunnel per bearer for UE.
S1-MME	E-UTRAN and MME	An interface for the control plane between the eNB and the MME
S3/S4	Serving GPRS Support Node (SGSN), MME and SGW	An interface to provide user and bearer information exchange for inter-3GPP access networks. It also provides control and mobility support between GPRS core and the SGW function.
S5	SGW and PDN-GW	An interface defined between an SGW and a PDN-GW. It is used for SGW relocation due to UE mobility and if SGW needs to connect to a non-collocated PDN-GW for the required Packet Data Network (PDN) connectivity
S6a	MME and HSS	An interface to enable the transfer of subscription and authentication data.
S11	MME and SGW	An interface for the control plane between an MME and an SGW. It provides a GTP tunnel per user.
SGi	and PDN	An interface to connect to a PDN-external public or private data network or an intra-operator packets data network.
Gx	PDN-GW and PCRF	An interface for the control plane between a PCRF and a PDN-GW to transfer policy control and charging rules for QoS policy and charging control.
Rx	PCRF and PDN	An interface that provides the transport of application level session information between PCRF and external network.

2.1.3 LTE Protocol Stacks

The LTE protocol stack is separated between the user plane and the control plane according to the final purpose service. The separation of the control and user plane is an important feature of the LTE network architecture, it enables operators to implement QoS control on data traffic [AY11] [lteg].

User Plane

An IP packet for a UE is encapsulated in an EPC-specific protocol and tunneled between the PDN-GW and the eNB for transmission to the UE. A 3GPP-specific tunneling protocol called the GTP is used over the EPC interfaces, S1 and S5/S8. Figure 2.3 shows the user data plane protocol stack.

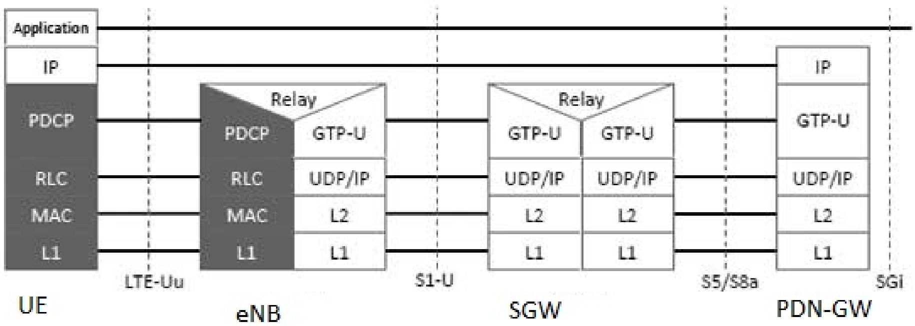

Figure 2.3: The E-UTRAN user plane protocol stack [lteg]

- Packet Data Convergence Protocol (PDCP) : The PDCP protocol performs header compression, ciphering and integrity protection and packet retransmission during handover.
- Radio Link Control (RLC) : The RLC protocol performs segmentation/concatenation of PDCP Packet Data Unit (PDU)s during construction of the RLC PDU for transmission and reassembly of the RLC PDUs to reconstruct the PDCP PDU during reception.
- Media Access Control (MAC) : The MAC protocol supports multiplexing and de-multiplexing between logical channels and transport channels. The MAC protocol supports QoS by scheduling and prioritizing data from logical channels.
- GTP-U: This is used to forward user IP packets over the EPC interfaces, S1 and S5/S8.

Control Plane

The control plane protocol function is to control the radio access bearers and the connection between the UE and the network. The protocol stack for the control plane between UE and MME is shown in Figure 2.4. The PDCP, RLC and MAC layer performs same function as the user plane with the exception that there is no header compression function for the control plane.

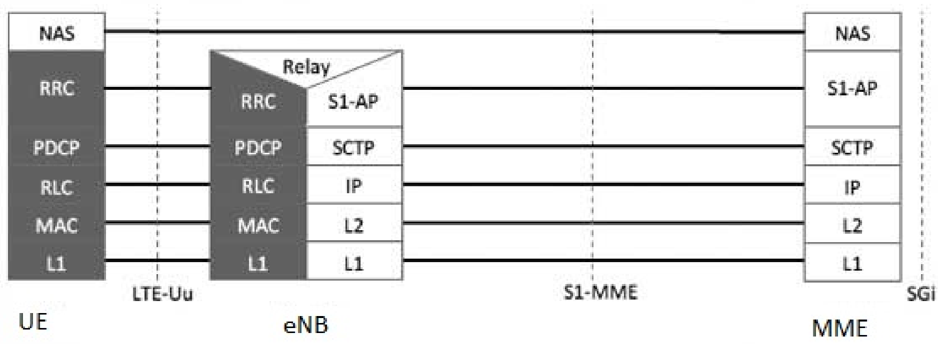

Figure 2.4: The E-UTRAN control plane protocol stack [lteg]

- Non Access Stratum (NAS): NAS protocol performs mobility management and bearer management functions.
- Radio Resource Control (RRC): RRC protocol supports the transfer of the NAS signaling. It is responsible for establishing the radio bearers and configuring all the lower layers using RRC signaling between the eNB and the UE.
- S1-AP: This protocol supports functions such as S1 interface management, NAS signalling transport and UE context management. It delivers the initial UE context to the eNB to setup radio bearers and manages modification or release of the UE context thereafter.

2.2 LTE Access Network Key Technologies

The E-UTRAN commonly know as the LTE is the access part of the EPS. The main requirements for the new access network are high spectral efficiency, high peak data rates, short round trip times and flexibility in frequency and bandwidth [3gp][BAEG][ltef].

The LTE is simply a network of base stations called eNBs. LTE introduces a flat all IP architecture that reduces the time it takes to access the radio and core network resources, typical initial data packet connection 50ms and round trip latency of 12-15ms. The reduced number of network elements in the flat architecture Figure 2.5 leads to less number of connection states as compared to previous generation access networks. IP connections established between the UE and the eNB remains constant, unless the UE is switched off, this eliminates the need to re-establish connections each time a user makes a service request.

Moreover, there are no centralized intelligent controllers and intelligence is distributed among the eNBs in which they are interconnected via the X2 interface and S1 interface to the core network. One of the advantage of the distribution approached is on the MAC protocol layer which is represented only in the UE and the eNB. This implementation of the MAC layer enables a more reliable and QoS aware scheduling techniques to achieve fast communication and decisions between UE and eNB on utilization of radio resources without an additional requirement for MAC sub-layer or controller. Thus, the flat all IP architecture helps to speed up the connection setup, easing the handover process and reducing the overall system latency.

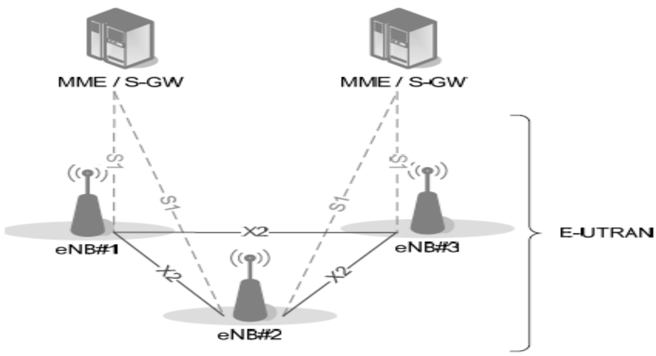

Figure 2.5: LTE Access Network Architecture [BAEG]

In the next sections we explain the main features that are included in 3GPP's 4G networks to attain the desired overall system improvements with LTE networks.

2.2.1 Orthogonal Frequency Division Multiplexing

As one of the key elements in LTE design, a multicarrier approach OFDM was chosen by 3GPP as the signal bearer for multiple access. OFDM is a form of transmission that uses a large number of close spaced subcarriers of 180KHz each that are modulated with low rate data. The subcarrier signals are orthogonal to each other so that the normally expected mutual interference can be avoided. OFDM is robust to multipath fading and interference, supports both FDD and TDD. It allows digital signal processing schemes, compatible with WiMAX and suitable for carrying high data rates. In view of these advantages, the use of OFDM and associated access technologies are natural choices for the new LTE cellular standard [ofd][3gp].

Figure 2.6 shows OFDM system structure. OFDM modulates a block of data symbols simultaneously over one OFDM symbol, where one OFDM symbol is the time used to transmit all of subcarriers. At the transmitter side the baseband modulator modulates the input block data using different modulation formats such as Quadrature

Phase Shift Keying (QPSK) or 16-64 Quadrature Amplitude Modulation (QAM). These modulated symbols are then mapped to subcarriers in case of Orthogonal Frequency Division Multiple Access (OFDMA). While in Single Carrier-OFDM (SC-OFDM) an N-point Discrete Fourier Transformer (DFT) transforms these symbols in to frequency domain before mapping. An inverse DFT is used to transform the modulated subcarriers in frequency domain in to time domain samples. A cyclic prefix copies a portion of the samples at the end of the time domain samples block to the beginning. The block of samples are then serialized in the time domain and converted to analog signals, finally the RF section modulates the I-Q samples[1] to final transmission radio frequency. An exact inverse operation is performed at the receiver side [sin][kok].

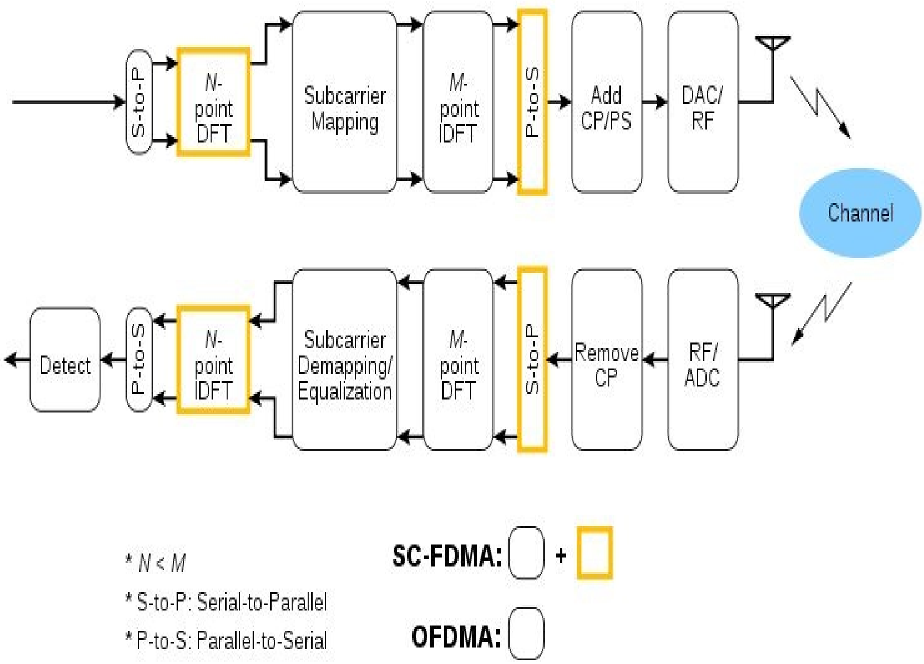

Figure 2.6: OFDM transmitter and receiver structure [sin]

Subcarrier Mapping

DFT output of the data symbols is mapped to a subset of subcarriers, a process called subcarrier mapping. The subcarrier mapping assigns DFT output complex values as the amplitudes of some of the selected subcarriers.

[1] I-Q samples are samples that are converted from a polar coordinate system to a Cartesian (X,Y) coordinate system.

There are two main types of subcarrier mappings: Figure 2.7

1. *Localized mapping*: the DFT outputs are mapped to a subset of consecutive sub-carriers thereby confining them to only a fraction of the system bandwidth.

2. *Distributed mapping*: the DFT outputs of the input data are assigned to sub-carriers over the entire bandwidth non-continuously, resulting in zero amplitude for the remaining subcarriers.

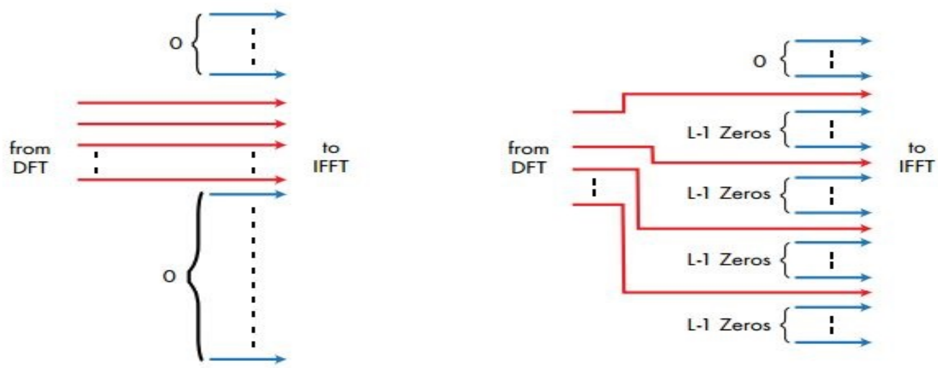

Figure 2.7: Localized vs Distributed Mapping [kok]

The actual implementation of the technology is different between the downlink and the uplink as a result of the different requirements between the two directions and the equipment at either end. To enable possible deployment around the world, supporting as many regulatory requirements as possible, LTE is developed for a number of frequency bands currently ranging from 700 MHz up to 2.7GHz. The available bandwidths are also flexible starting with 1.4 MHz up to 20 MHz.

Downlink OFDM: OFDMA

The OFDMA signal used in LTE comprises a maximum of 2048 different subcarriers having a spacing of 15 kHz. Although it is mandatory for the UE to have capability to be able to receive all 2048 subcarriers, not all need to be transmitted by the eNB which only needs to be able to support the transmission of 72 subcarriers. In this way all mobiles will be able to talk to any base station. The subcarriers are split in to units called Resource Blocks (RBs), it is the smallest unit of resources that can be allocated to a user. This enables the system to be able to compartmentalise the data across standard number of subcarriers. One RB comprises of 12 subcarriers regardless of the overall LTE bandwidth [Net11] [lteb].

Table 2.2 shows how many subcarriers and RB there are in each bandwidth for uplink and downlink.

Table 2.2: LTE standard bandwidth and corresponding RBs

Bandwidth	Resource Blocks	Sub-carriers (downlink)	Sub-carriers (Uplink)
1.4 MHz	6	73	72
3 MHz	15	181	180
5 MHz	25	301	300
10 MHz	50	601	600
15 MHz	75	901	900
20 MHz	100	1201	1200

In OFDMA the RB is 180kHz wide in frequency and 0.5ms long in time domain. In frequency, the standard number of subcarriers used per RB are 12 for most channels. The minimum unit of the time domain is a Symbol, which amounts to 66.7 us. Regardless of bandwidth, the symbol length does not changes. The largest unit in time domain is a frame, each of which is 10 ms in length. Each of the frame consists of 10 sub frames, each of which is 1 ms in length. Each of sub frame consists of 2 slots, each of which is 0.5 ms in length. Each of slots consists of 7 symbols, each of which is 66.7 us. Consequently, the allocated resource blocks determines the modulation technique to be used and the data transmission rate. Which resource blocks and how many the user gets at a given point in time depend on advanced scheduling mechanisms in the frequency and time dimensions.

A generic LTE radio frame format is shown in Figure 2.8, it has a time duration of 10 ms, consisting of 20 slots of each 0.5 ms. Two adjacent slots form a subframe of 1 ms duration, which is also one Transmission Time Interval (TTI). Each slot consists of seven OFDM symbols with short/normal Cyclic Prefix (CP) or six OFDM symbols with long/extended CP. CP is the process of extending each symbol to avoid inter-symbol-interference by duplicating a portion of the signal at the symbol ends, which is removed at the receiver [lted] [kok].

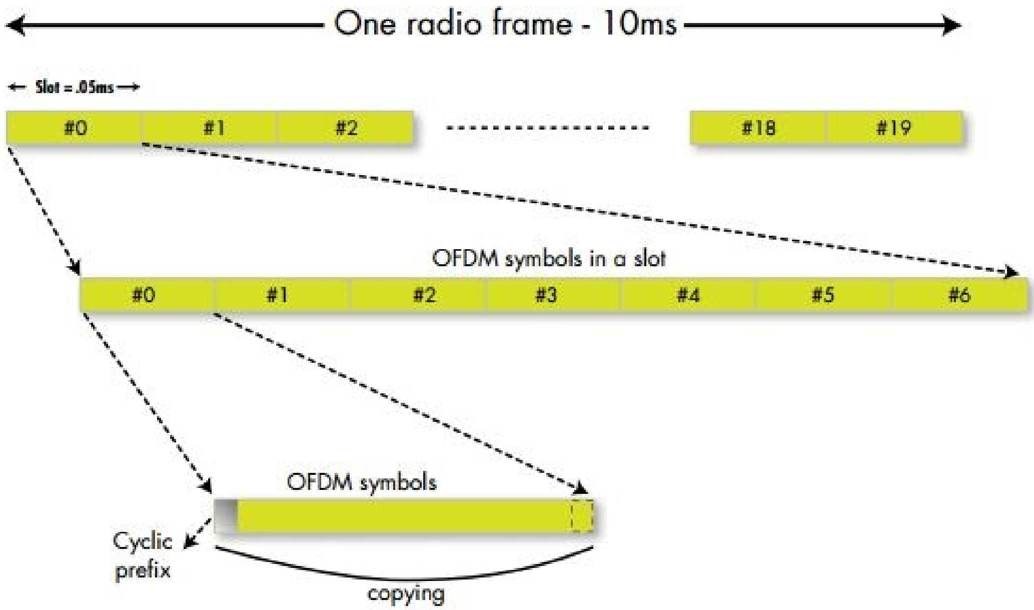

Figure 2.8: LTE Radio Frame Format[kok]

Uplink OFDM: SC-FDMA

In the uplink LTE uses a modified form of OFDMA called SC-OFDM. It has similar throughput performance and complexity as OFDMA. OFDMA has a very high Peak to Average Power Ratio (PAPR). High PAPR requires expensive and inefficient power amplifiers with high requirements on linearity, which increases the cost of the terminal and drains the battery faster. SC-OFDM brings the benefit of low peak-to-average power ratio compared to OFDMA making it suitable for uplink transmission by user terminals.

In OFDMA each subcarrier only carries information related to one specific symbol while in SC-OFDM contains information of all transmitted symbols. For a single sample in time the signal being transmitted is composed of the summation of all symbols, due to mapping of the symbols' frequency domain samples to subcarriers. Thus, SC-OFDM offers spreading gain in a frequency selective channels [kok].

The main difference between OFDMA and SC-OFDM transmitter is the DFT mapper. After mapping data bits into modulation symbols, the transmitter groups the modulation symbols into a block of N symbols. An N-point DFT transforms these symbols in time domain into frequency domain. The frequency domain samples are then mapped to a subset of M subcarriers where M is typically greater than N.

2.2.2 Multiple-Input Multiple-Output Systems

Multiple Input- Multiple Output (MIMO) systems are one of the major enabling technologies for LTE. They allows higher data rate transmission through the use of multiple antennas at the transmitter and receiver to provide simultaneous transmission of multiple parallel data streams over a single radio link [Sha10][DFJ+08].

The idea behind MIMO is that the signals on the transmitting antennas at one end and the receiving antennas at the other end are combined in such a way that the quality of the communication for each MIMO users will be improved. The key MIMO feature is its ability to turn multipath propagation into its benefit. MIMO takes the advantage of random multipath fading. In the presence of random fading, the probability of losing the signal decrease with the number of decorrelated antenna elements being used [GSD+03].

MIMO Capacity

The Single-Input Single-Output system (SISO) system is the most commonly used one due to high power consumption of MIMO systems. In SISO the maximum channel capacity is given by the Shannon-Hartley relationship:

$$C = B \times log_2(1 + SINR_{\mathrm{avg}})$$

where C is the channel capacity in bits per second, B is the channel bandwidth in Hz and $\mathrm{SINR}_{\mathrm{avg}}$ is the average Signal to Interference and Noise Ratio (SINR) at the receiver.

For MIMO system the capacity is given by :

$$C = B \times log_2(1 + M_{\mathrm{T}} \times N_{\mathrm{R}} \times SINR_{\mathrm{avg}})$$

where M_{T} is the number of transmitting antennas and N_{R} is the number of receiveing antennas. Thus, obtaining an $\mathrm{M}_{\mathrm{T}}\mathrm{N}_{\mathrm{R}}$ fold increase in the $\mathrm{SINR}_{\mathrm{avg}}$ and increasing channel capacity.

In addition to high power consumption, MIMO system also comes with implementation complexity of identifying correlation matrices between the transmit/receive antennas, as well as the channel propagation conditions.

LTE supports 2x2 and 4x4 MIMO systems, in this thesis work we have used a SISO and 2x2 MIMO, 2 transmitting and 2 receiving antenna system shown in Figure 2.9.

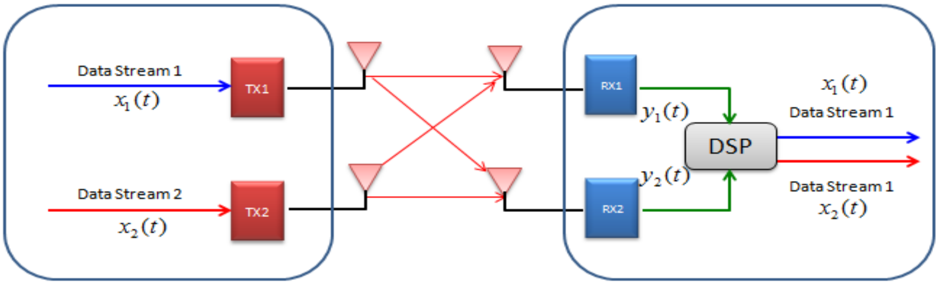

Figure 2.9: Simplified MIMO structure [mim]

2.2.3 Adaptive Modulation and Coding

In LTE, Adaptive Modulation and Coding (AMC) is implemented on the uplink and downlink, where the modulation scheme as well as the coding scheme are changed automatically for best transmission performance according to channel conditions [EUa][Sha10].

In bad channel conditions with low SINR level, a low constellation modulation scheme QPSK is used. In QPSK, two bits are encoded into a single word for transmission. The signal constellation of a QPSK modulation consists of a square grid. The modulated signals contain a level based on the number of bits used. The 16-QAM and 64-QAM modulation schemes are used in better channel conditions, and the data are mapped into both phase and amplitude changes on the carrier frequency. For 16-QAM, every 4 bits are given a signal value from the 16-level constellation. Figure 2.10 shows the difference between the QPSK and the 16-QAM signal constellations. A 64-QAM modulation scheme follows that of the 16-QAM but instead encodes 6-bits into one signal level/phase compared to 4-bits in 16-QAM.

2.2.4 Hybrid Automatic Repeat Request

LTE networks deploys two standard retransmission schemes HARQ and Automatic Repeat Request (ARQ). HARQ is implemented to correct the error packets in the Physical Layer (PHY), while ARQ is implemented in RLC layer to take care of residual errors. In standard ARQ, redundant bits are added to data to be transmitted using an error-detecting code such as a Cyclic Redundancy Check (CRC). Receivers detecting a corrupted message will request a new message from the sender. In HARQ, the original data is encoded with a Forward Error Correcting (FEC) code, and the parity bits are either immediately sent along with the message or only transmitted upon request when a receiver detects an erroneous message [DFJ+08][hyb].

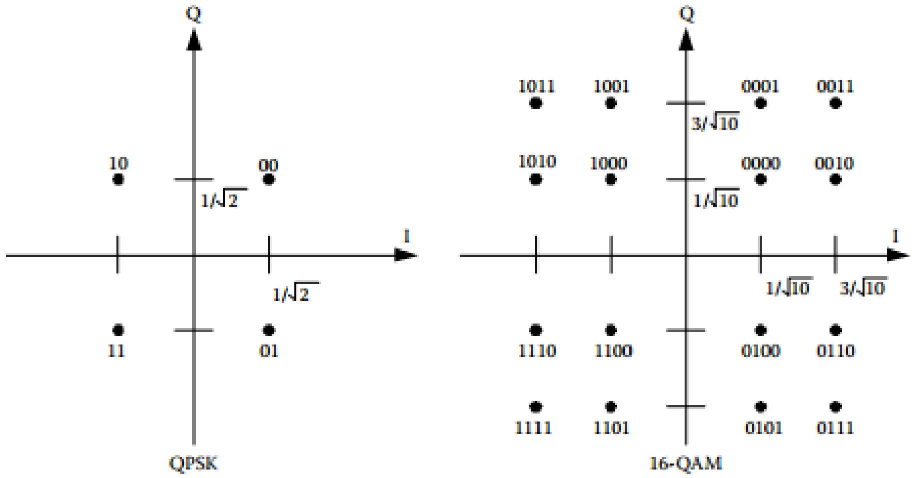

Figure 2.10: QPSK and 16-QAM signal constellations, gray coded [EUa]

According to 3GPP specification TS 36.321 [rGPP12] there is one HARQ entity at the UE with 8 stop-and-wait processes containing HARQ buffer. A number of parallel HARQ processes are used in the UE to support the HARQ entity. This allows transmissions to take place continuously while waiting for the feedback on the successful or unsuccessful reception of previous transmissions. One MAC scheduler process sends in single TTI of 1ms, receiver takes 3ms for processing, 1ms for Acknowledgment (ACK)/Negative-Acknowledgment (NACK) and 3ms for processing back in transmitter, total 8ms. This 8ms is called Round Transmit Time (RTT), during this time the MAC scheduler would not know whether to transmit a new or retransmit an old data. Therefore to use these 7ms between RTTs, the LTE HARQ entity uses one of these 8 processes at a given ms and the MAC scheduler picks up which process to be used. HARQ works at the PHY layer but is controlled by the MAC layer, the scheduler at the MAC layer is in charge of controlling the 8 HARQ processes.

Figure 2.11 shows the HARQ process behaviour of each one of the 8 processes in the HARQ entity. If a received data has an error, the receiver sends NACK and a retransmission process is initiated by the PHY. The receiver does not discard the erroneous data but rather stores it in a buffer. On the other hand, upon the retransmission request the transmitter will send same data again but with different set of coded bits. The receiver then combines the previously stored erroneous data with the new data, this helps the retransmission performance. This will repeat as long as the receiver is not able to recover the complete data. If the receiver receives the data correctly, it will send an ACK indication, this would delete the stored data and frees the buffer for next transmission. According to the standard [rGPP12], the uplink retransmissions are synchronous and therefore are allocated 7 ms after the original transmission. On the other hand, for the downlink, they are asynchronous and therefore can be allocated in a more flexible way starting from 7 ms.

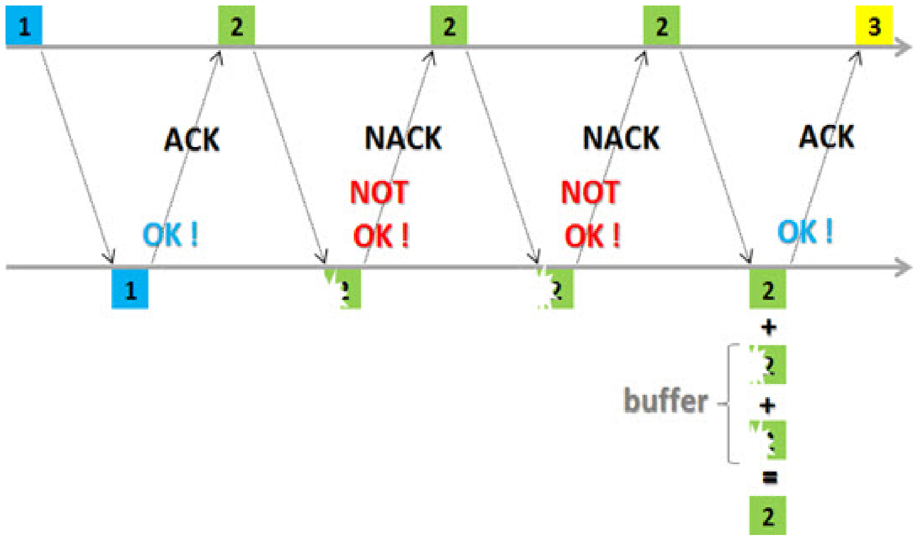

Figure 2.11: HARQ processes behaviour in LTE [tel]

2.3 LTE Quality of Service

QoS in networks is the ability of the network to enforce different priorities for different application types, subscribers, or data sessions, while guaranteeing a certain level of performance to a data session.

In 2G/3G networks QoS classification was not available, but with LTE as an all IP 4G network, it defines QoS to not only guarantee the quality of a service but also support different level services for other latency or bit-rate sensitive applications. LTE has adopted a class-based QoS model called QoS Class Identifier (QCI), shown in Table 2.3, to ensure bearer traffic is allocated with appropriate QoS [BAEG][Pcca].

In LTE network QoS is implemented between UE and PDN-GW and is applied to a set of bearers. A bearer is a virtual concept and is a set of network configuration to provide special treatment to set of traffic, e.g. VoIP packets are prioritized by network compared to web browser traffic. QoS is applied on Radio bearer, S1 bearer and S5/S8 bearer, collectively called as EPS bearer, each being associated with a QoS as shown in Figure 2.12 [EUb][Pcca].

There are two categories of bearers in LTE, default and dedicated. When ever a UE is attached to LTE network at least one default EPS bearer is created with a Non-Guaranteed Bit Rate (NGBR) QoS, i.e. support of best effort delivery and remains activated until the UE detaches from the network. On the other hand, dedicated bearer is always established when there is a need to provide QoS to specific service such as VoIP, Video etc. The dedicated EPS bearers can be either Guaranteed Bit Rate (GBR) or NGBR depending on the service and QCI value. Multiple bearers can be established for a user in order to provide different QoS streams or connectivity to different PDNs. For example, a user might be engaged in a voice call while at the same time performing web browsing. A VoIP bearer would provide the necessary QoS for the voice call, while a best-effort bearer would be suitable for the web browsing [Bas][qos15].

According to [Pcca] each EPS bearer established has an associated QCI and an Allocation and Retention Priority (ARP). QCI is needed to classify the different types of bearers into different classes with each class having appropriate QoS parameters for the traffic type. Each QCI is characterized by key parameters such as GBR or NGBR, priority, packet delay budget and acceptable packet loss rate. These key parameters from the QCI index determine how the scheduler in the MAC handles packets sent over the bearer in terms of scheduling policy, queue management policy and rate-shaping policy. For example, a packet with higher priority can be expected to be scheduled before a packet with lower priority. For bearers with a low acceptable loss rate, an acknowledged mode can be used within the RLC protocol layer to ensure that packets are delivered successfully across the radio interface. Thus, QCI provides operators with an effective and simple way to differentiate between services. This is an important feature and one which has a major impact on the subscriber experience and on service delivery.

The ARP of a bearer is used to decide whether or not a requested new bearer should be established in case of radio congestion. It is also used for prioritization of the bearer for preemption with respect to a new bearer establishment request. ARP does not affect the priority of the delivered packet once a bearer is created, and thus the network nodes forwards the packets regardless of their ARP values[2]. ARP is most used in an emergency VoIP call where an existing EPS bearer can be removed if a new one is required for a emergency VoIP call.

2.3.1 QoS Provisioning and Enforcement

As shown in Figure 2.12 an EPS bearer has to cross multiple interfaces from UE/PDN-GW to PDN-GW/UE. Across each interface, the EPS bearer is mapped onto a lower layer bearer, each with its own bearer identity. Each node must keep track of the

[2]an integer ranging from 1 to 15, with 1 being the highest level of priority

Table 2.3: Standardized QCI characteristics [Pcca]

QCI	Resource Type	Priority	Packet delay budget (ms)	Packet error loss rate	Example services
1	GBR	2	100	10^{-2}	Conversational Voice
2	GBR	4	150	10^{-3}	Conversational Video (Live Streaming)
3	GBR	3	50	10^{-3}	Real Time Gaming
4	GBR	5	300	10^{-6}	Non-Conversational Video (Buffered Streaming)
5	Non-GBR	1	100	10^{-6}	IMS Signalling
6	Non-GBR	6	300	10^{-6}	Video (Buffered Streaming) TCP-based (e.g., www, e-mail, chat, ftp, p2p file sharing, progressive video, etc.)
7	Non-GBR	7	100	10^{-3}	Voice, Video (Live Streaming) Interactive Gaming
8	Non-GBR	8	300	10^{-6}	Video (Buffered Streaming) TCP-based (e.g., www, e-mail, chat, ftp, p2p file sharing, progressive video, etc.)
9	Non-GBR	9	300	10^{-6}	Video (Buffered Streaming) TCP-based (e.g., www, e-mail, chat, ftp, p2p file sharing, progressive video, etc.)

binding between the bearer IDs, also called Tunnel End ID (TEID), across its different interfaces.

The QoS parameters, shown in Table 2.3, applied to a default bearer are provisioned to an HSS as subscription information by a network operator. And then, when the default bearer is activated, an MME downloads the QoS profile for the bearer from the HSS and sends it to EPS entities appropriately. QoS parameters for the default bearer provided by the HSS can be modified by the MME upon creation of a new EPS session with different QCI index. QoS parameters applied to a dedicated bearer are provisioned by PCRF. The PCRF determines QoS parameters for the bearer based on the subscription information it received when the bearer is activated. QoS parameters for EPS bearers are enforced in UE, eNB, SGW and PDN-GW, that deliver user traffic between UE and PDN-GW [ltee][Bas].

IP packets mapped to the same EPS bearer receive the same bearer-level packet forwarding treatment. In order to provide different bearer-level QoS, a separate EPS bearer must therefore be established for each QoS flow. User IP packets must then be filtered into the appropriate EPS bearers based on Traffic Flow Templates (TFT) on UE and PDN-GW. TFTs use the five-tuple structure[3] to filter packets such as VoIP from web-browsing traffic, so that each packet can be sent to respective bearers [ltee].

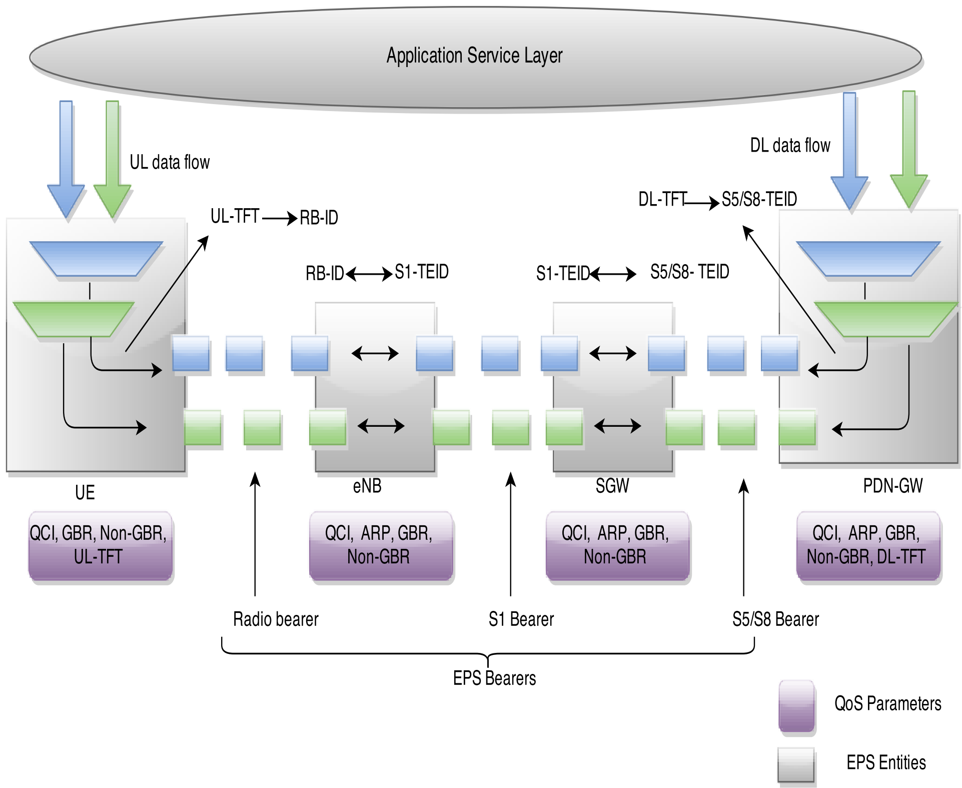

Figure 2.12: LTE QoS Parameters, Provisioning and Enforcement

[3]five-tuple structure - contains the Source and destination IP Address, source and destination port number and transmission protocol type

Chapter 3

Network Coding and Related Works

In this chapter the concept, implementation and variants of NC is explained. The deployment of NC in LTE networks including the alternative ways of deployment is explained, related works are also discussed.

3.1 Network Coding

NC is a networking technique in which transmitted data is encoded and decoded to increase network throughput, reduce delays and make the network more robust. It allows the system to mix and re-encode different data packets at intermediate nodes rather than store and forward them. Under this premise, it is no longer required for the system to keep track of which packets have been received: receivers need only aim at accumulating enough coded packets in order to recover the information. Thus, algebraic algorithms are applied to the data to accumulate the various transmissions. The received transmissions are decoded at their destinations. This means that fewer transmissions are required to transmit all the data, but this requires more processing at intermediary and terminal nodes [KRH+08].

Wireless networks exhibit significant data redundancy i.e., there is a large overlap in the information available to nodes and wireless broadcast increases this redundancy. NC could benefit in improving network utilization by leveraging the redundant information in nodes [FKM+07]. Thus, NC is perceived to be useful in wireless mesh networks, messaging networks, storage networks, multicast streaming networks and other networks where the same data needs to be transmitted to a number of destination nodes. Large networks can increase their efficiency through the use of NC, but high overhead costs may make them less amenable for small networks.

3.1.1 Random Linear Network Coding

Nowadays the most prominent type of NC is RLNC.The simplicity of its coding principle has been shown to allow the code to be transported or stored along with data, which enables a range of NC applications for networks, storage and mobility applications [HPF11], [GQLM09], [ltea].

In RLNC, coded packets are created from a block of data that are split into a number of symbols of a specified size and combined with random coefficients. The network nodes receive a series of encoded packets that are passed to the decoder which will be able to reconstruct the original symbols after receiving sufficient linearly independent packets. In order to decode the coded packets, the coefficient used to encode must be sent along with the packets. The architecture of a coded packet is shown in Figure 3.1.

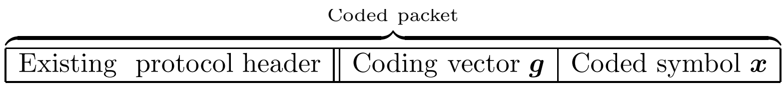

Figure 3.1: Coded Packet Structure [HPF11]

Encoding

The process of creating the coded packets at the source node is called encoding. Figure 3.2 shows the encoding process in RLNC. To transmit a block of data, the data block M is divided into blocks of certain sub blocks of definite size, also called generation designated as g, $\boldsymbol{M} = [\mathrm{m}_1, \mathrm{m}_2, \mathrm{m}_3 ... \mathrm{m}_\mathrm{g}]$. Each of the sub-blocks are then multiplied with a randomly generated coefficients of same size as the sub blocks in the Galois field[1]. Thus the multiplication result is also in the same field. The resulting sub blocks are bitwise XORed to form the coded packets $\boldsymbol{X}$ to be sent over the network. Any number of encoded packets can be generated for a single g size generation of coded packet, each symbol of the resulting coded packet is a linear combination of the corresponding symbols in the native packets.

The coding coefficients for encoding the coded packets are chosen at random from a Galois field. Choosing the coefficients randomly allows the sender and receiver to generate coded packets with little overhead. Since both encoder and decoder must use the same coefficients, it is sufficient for the encoder to transmit only a seed together with the coded packet, and both encoder and decoder can use the seed to generate the same pseudo random coefficients.

Decoding

For the decoding process the receiver must receive sufficient linearly independent symbols and coding vectors from the generation to decode the data successfully. All

[1] Galois field is a library for finite field arithmetic.

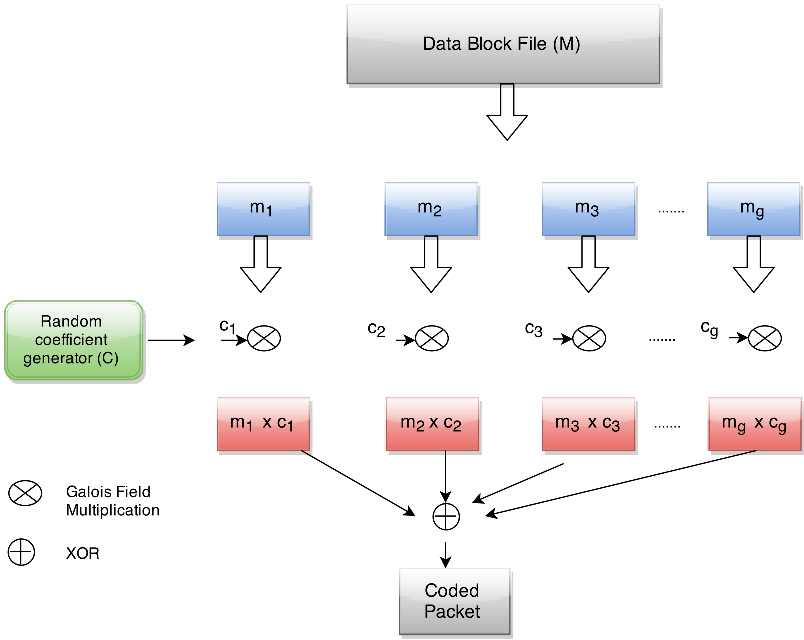

Figure 3.2: RLNC encoding process

received symbols are placed in the matrix $\boldsymbol{X} = [x_1, x_2, x_3...x_g]$ and all coding vectors are placed in the matrix $\boldsymbol{G} = [g_1, g_2, g_3...g_g]$ we denote G as the decoding matrix. The original data M can then be decoded as $\boldsymbol{M} = \boldsymbol{X}$ x $\boldsymbol{G}^{-1}$ by the decoder.

Since the coefficients are chosen randomly, finding the inverse of the coefficient matrix G can be a problem and therefore more coded packets might be needed. An efficient and common way of finding the inverse of G or whether it exist or not is the use of Gauss-Jordan elimination. Figure 3.3 shows a modified version of the Gauss-Jordan elimination method, on each run the algorithm attempts to get the decoding matrix in to reduced echelon form.

First the received vector and symbol $\boldsymbol{g}^\frown$ and $\boldsymbol{x}^\frown$ is forward substituted into the previous received vectors and symbols $\boldsymbol{G}^\frown$ and $\boldsymbol{X}^\frown$ respectively, and subsequently backward substitution is performed. If the packet was a linear combination of previous received packets it is reduced to the zero vector 0_g and discarded, otherwise the packet is novel and is kept. This reduces the computational cost when a linear dependent packet is received. In real world scenarios the probability of receiving a linearly dependent packet can be high, in such cases this approach would be beneficial.

Input: $\hat{\boldsymbol{x}}$,$\hat{\boldsymbol{g}}$

pivotPosition $\leftarrow 0$ ▷ 0 Indicates that no pivot was found

recipe $\leftarrow \mathbf{0}_g$

for $i \leftarrow 1 : g$ **do**

  if $\hat{\boldsymbol{g}}[i] = 1$ **then**

    if $\hat{\boldsymbol{G}}[i, i] = 1$ **then**

      $\hat{\boldsymbol{g}} \leftarrow \hat{\boldsymbol{g}} \oplus \hat{\boldsymbol{G}}[i]$ ▷ substitute into new vector

      recipe$[i] \leftarrow 1$

    else

      pivotPosition $\leftarrow i$ ▷ pivot element found

      break

if pivotPosition > 0 **then**

  ExecuteRecipe($\hat{\boldsymbol{x}}$,recipe)

return pivotPosition

Figure 3.3: Forward Substitute suppress null algorithm: modified Gauss-Jordan elimination used in decoding coded packets [HPF11]

Systematic Vs Non-systematic

In RLNC data can be sent in a systematic or non-systematic form. In systematic RLNC the sender can send some or all of the original symbols within a given block uncoded. Coded packets can then be generated later to repair any packet losses. Non-systematic RLNC on the other hand sends coded packets at the start of the transmission and for any packet losses additional combination of packets will be sent.

[SJ09], [HPFL09] explains that if systematic RLNC is used at certain nodes within a network, the throughput will not be reduced relative to non-systematic. Furthermore, if packets can traverse the entire network in their systematic (uncoded) form, per-packet delay can be reduced, decoding complexity can be reduced, and the potential to recover packets from incomplete coded blocks will be improved.

In comparison to non-systematic, systematic may increase the incidence of out-of-order packet delivery and may require additional coordination to determine the coding strategy used at each node. These costs of systematic network coding will depend on the size and topology of the network. For instance for smaller topologies and less lossy links systematic can be more advantageous while for large networks with broadcasting nature the user of non-systematic can perform better. This two schemes of RLNC are compared in chapter 6 of this thesis work.

3.2 Network Coding in LTE Network

Despite the fact that the LTE physical layer represents the state-of-the-art in communications technology, novel solutions for LTE adaptation to mutimedia services are needed. Recently RLNC is becoming a promising approach for a low complexity, adaptive and content aware packet level error resilience solution. Given the unique flexibility of RLNC to efficiently bridge the upper layer media packetization and the lower layer wireless packet transmission, RLNC is considered as a powerful cross layer solution for reliable mutimedia delivery over the LTE [VKST12].

The HARQ retransmission schemes, discussed in section 2.2.4, has some shortcomings such as high signalling overhead and delays. These shortcomings result in the risk of significant performance degradation, in terms of energy efficiency and data rates, under delay constraint applications [vRia]. An alternative to HARQ is RLNC, if supported in future networks could enable to overcome the shortcomings of HARQ. Some of the key aspects of the RLNC over HARQ in LTE are, HARQ incurs signalling overheads with its retransmissions including ACK and NACK messages while in RLNC only one ACK message is sent once the subscriber decodes all the received packets. RLNC dynamically adopts the transmission rate to the channel condition to achieve optimum throughput while in HARQ case retransmissions are sent at the same rate as the first packet. For erroneous packets there is an 8ms delay between retransmission attempts, multiple packet errors can cause timeouts and potential loss of overall service quality, RLNC the feedback delay can be reduced to as low as 1-2ms and timeouts are not necessary. Generally, RLNC is more suitable for multihop relaying techniques and multimedia streaming applications.

As [HL12] explains the implementation of the RLNC in the LTE protocol stack poses several challenges beyond designing the efficient encoding and decoding algorithms. First, one has to decide at which layer of the protocol stack to implement the RLNC. Second, the RLNC implementation should incur as little changes to the existing functionality of the protocol stack as possible. Third, the RLNC implementation should exploit the existing services provided by the protocol stack such as the AMC and HARQ as much as possible. Last but not least, one has to also carefully judge the achievable performance improvement due to RLNC such as the energy savings and the reduced transmission delay against the added implementation complexity.

3.2.1 Application Layer Network Coding

Application Layer-RLNC (AL-RLNC) cab be useful in providing a packet-level protection at the application layer to complement the bit-level FEC at the physical layer. RLNC can be implemented on both the eNBs and on each individual UEs. The basic encoding process follows the same process described in section 3.1.1 and Figure 3.2 for RLNC.

In AL-RLNC solution, the standardized LTE protocol layer stack shown in Figure 3.4 remains the same. AL-RLNC is deployed on top of the existing MAC layer HARQ based packet transmission process. As [KVT12], [VKST14] explains it, in the AL-RLNC/MAC-HARQ solution , as IP packets enters the eNBs PDCP layer the PDCP layer performs header compression and ciphering. Then the PDCP encapsulated IP packets are delivered to the RLC layer. The RLC layer performs segmentation/ concatenation of IP packets into RLC packets to fit the MAC frame size requirements. Each MAC frame is allocated a single PHY Transport Block (TB) for transmission over the eNB/UE interface, every TTI. The PHY layer carries all the information from the MAC transport channels over the air interface. In addition takes care of the link adaptation for realizing AMC in the MAC layer to provide a matching of the modulation and coding techniques to the radio link interface condition.

Thus, the AL-RLNC solution enhances the transmission of packets over the LTE networks by applying RLNC coding technique on the existing MAC-HARQ solution. The encoded packets are treated same as if they were normal uncoded IP packets. In downlink the eNB keeps sending a coded packets until the UE sends a single ACK to notify the reception of sufficient linearly independent packets to fully decode the packets.

In this thesis we chose to implement AL-RLNC solution for two reasons. First, the Kodo library is applicable only to the upper layers and is not designed to use the encoding-decoding functionality of the library in the lower layers. Second, the encoding packets at the application layer has been shown to improve the performance of wifi X-topology in the author's previous work. Another reason is from implementation perspective, deploying the RLNC in the application layer allows to simply integrate the Kodo library coding scheme on top of the current stack and with out affecting the functionality of LTE protocol stack and/or compatibility of UEs and eNBs. The performance of the LTE network with AL-RLNC solution is experimented and analysed in chapter 6.

3.2.2 MAC Layer Network Coding

The MAC-RLNC solution in Figure 3.5 is proposed as an alternative for the MAC-HARQ protocol [VKST14][VKST12]. In contrast, the proposed MAC-RLNC scheme

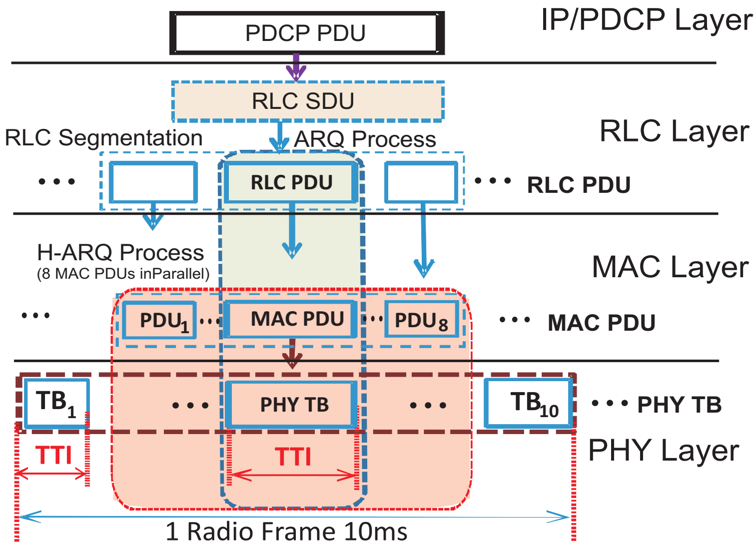

Figure 3.4: eNB RAN protocol: downlink AL-RLNC/MAC-HARQ solution [KVT12]

does not segment the PDCP packet. Instead, the RLC layer encapsulates the PDCP packet directly into the RLC packet. In case the larger RLC packets are desirable by the MAC layer, the RLC layer concatenates multiple PDCP packets into a single RLC packet. At the MAC layer, RLC message is processed by the MAC-RLNC sublayer: it is divided into K equal length source symbols from which a stream of encoded symbols is produced.

An appropriate number of equal-length encoded symbols are grouped into a MAC frame to fit the upcoming PHY TBs reported by the MAC scheduler. The MAC frame is encapsulated into the PHY TB and transmitted without HARQ retransmissions. From each correctly received PHY TB at the UE, the set of encoded packets is extracted and delivered to the decoder at the MAC-RLNC decoding sublayer. As soon as K linearly independent encoded packets are received from the stream of MAC frames, the UE MAC entity feeds back a single ACK message finalizing the RLC packet delivery.

Besides modifying the LTE protocol stack by addition of a new MAC-RLNC sublayer, this solution faces one key design issue of selecting an appropriate encoding symbol size to fit the varying PHY TBs as a result of the link adaptation functionality of the LTE protocol. The protocol architecture, implementation and benefits of this solution are further discussed in the work listed in [VKST14] and [KVT12].

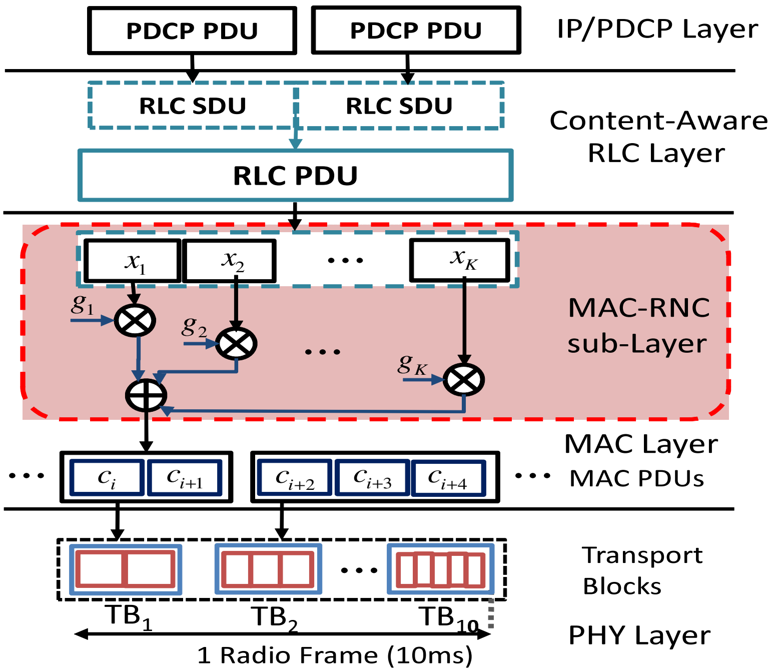

Figure 3.5: eNB RAN protocol: downlink MAC-RLNC solution [VKST14]

3.3 Related Works

The deployment of LTE wireless mobile system imposes a high demand on network performance. Different operators and organizations are lining up behind LTE to bring their own set of services and devices. Therefore, assessment and analysis of QoS performance of the LTE networks beforehand is fundamental step to take.

A lot of theoretical works has been done in the area of performance analysis and validation of the main features of the LTE networks. Channel and QoS aware scheduling performance was studied in [Tsa13] and [BB14]. They both discuss the optimization of scheduling and resource allocation for separate streams and show that a significant gain can be obtained in terms of spectral efficiency, QoS requirements and system capacity with schedulers.

Huang *et al.* [HQG$^+$13] investigates the effect of network protocol and application behaviour on performance of LTE networks and developed a novel and light weight passive bandwidth estimation technique for LTE networks. The end-to-end performance management across the LTE networks is discussed in [LS11]. It investigates how to deliver QoS traffic from different applications and presents a high level performance management system architecture for monitoring and harmonization of QoS based service differentiation. Furthermore, Oana *et al.* [IB13] analyzes performance of QoS networks using system level simulator, which the first part of this thesis is highly inspired by.

NC was introduced by [ACLY00] in 2000, the aim of this work was to show how bandwidth can be saved with NC and to find out if NC has a significant impact on the design of switching systems in a point-to-point communication. Following the breakout of the concept NC, it was further studied and implemented in wireless mesh networks by [KRH+08] where it proposes the mixing of different sources to increase the information content of transmissions, called the intersession NC.

Even though there has been many theoretical and practical works related to integration of NC into wireless mesh networks, this is not the case in LTE networks. The works done in [KHT+11][VKST14] [VKST12][HL12] and [HLOH11] explains the two most popular approaches for implementing NC at the application and MAC layer of LTE networks. Similar RLNC based schemes have been considered for integration in WiMAX technology in [JLK08] and [TFM+12]. Moreover, [FGP14] and [TKV+14], explains how NC in LTE networks can be used to improve the spectrum efficiency by facilitating resource reuse by multiple transmissions, how video streams are delivered as multimedia and multicast service flows that utilize the RLNC principle. Chadi *et al.* in [KVT12] proposes and investigates a detail cooperative access network wide MAC layer protocol based on MAC-RLNC, it also analyzes the energy efficiency of RLNC in LTE networks.

Chapter 4

Network Simulator-3 and Kodo

In this chapter we discuss the simulation environment, the simulation tools, ns-3 and Kodo, that are used for performance analysis of LTE networks and implementing NC in LTE network.

4.1 Network Simulator-3

ns-3 is a free discrete-event network simulator software, licensed under the GNU GPLv2 license, and is publicly available for research, development and use with the goal of developing a preferred, open simulation environment for networking research. It is a C++ library which provides a set of network simulation models implemented as C++ objects and wrapped through python for GNU/Linux, FreeBSD and MAC OS X operating systems. The ns-3 simulation core supports research on both IP and non-IP based networks with majority focus on wireless/IP simulations which involve models for wifi, WiMAX, LTE and a variety of static or dynamic routing protocols for IP-based applications [wha][pNsnRn][snRn] [doc].

One of the reasons we chose to use ns-3[1] as a simulation tool for the thesis is that the author conducted a wifi experiment on previous work and believes that this library represents the LTE network architecture to a good degree of detail and has a high level design features in comparison with other discrete event network simulators, and it is briefly summarized in the rest of the section.

[nsk] explains the main features of ns-3. The use of C++ and Phyton allows users to take full advantage of support of each language. ns-3 features 'helper' layer Application Program Interfaces (APIs) alongside a full APIs to provide easier-to-use functions and function callbacks used to reduce compile-time-dependencies between simulation objects. In addition, The simulation design is oriented towards use cases that allow the simulator to interact with the real world. For example ns-3 packet

[1] Version ns-3.22 at the time of writing this work

objects are stored internally as packet byte buffers ready to be serialized and sent on a real network interface. Network nodes, sockets and netdevices are patterned after the Linux networking architecture. This better facilitates code reuse and improves realism of the models. However, ns-3 does not maintain an integrated development environment to configure, debug, execute, and visualize simulations in a single application window. Instead, users normally interact with this library by writing a C++ or a python application which instantiates a set of simulation models to set up the simulation scenario of interest and integrate configuration and visualization tools as needed.

The ns-3 library includes various modules consists of different models for different simulation purposes, below we mentioned the main modules used in this thesis work.

4.1.1 LTE Module

The LTE module consists of LTE model and EPC model for LTE-EPC simulation purpose, shown in Figure 4.1. The LTE model includes the LTE radio protocol stack which resides entirely within the UE and eNB nodes. It allows the simulation of LTE networks. While the EPC model includes the core network interfaces and protocols which reside in SGW, PDN-GW and MME and partially within the eNB nodes.

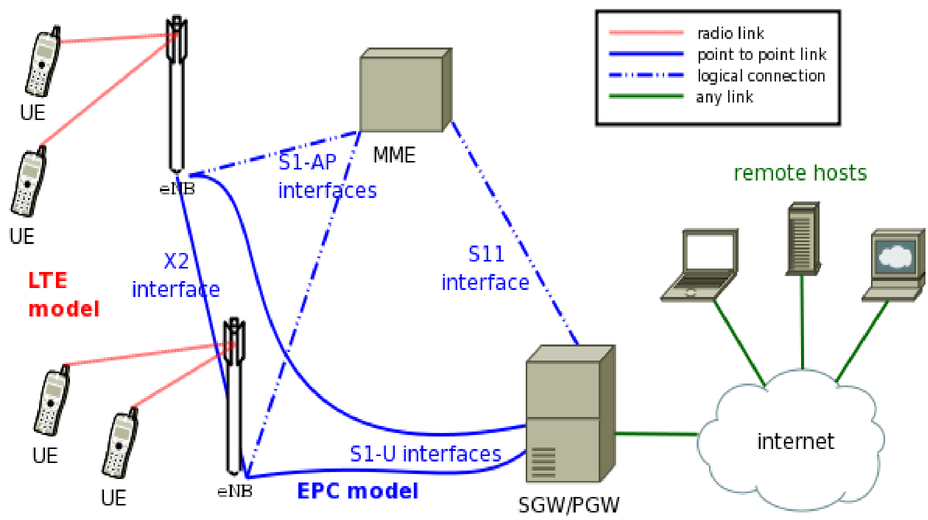

Figure 4.1: Overview of the LTE-EPC simulation model [ns3]

LTE Model

The ns-3 LTE model has been designed to support the evaluation of radio resource management, QoS-aware packet scheduling, inter-cell interference coordination and dynamic spectrum analysis aspects of the LTE systems. In order to evaluate these aspects to certain level of detail the ns-3 project considers the following requirements [ns3].

- At the radio level, the granularity of the model should be at least that of the RB, for packet scheduling and inter-cell-interference modeling.
- The model simulator must support the MAC Scheduler API published by the FemtoForum [v1.] for the implementation of scheduling and radio resource management algorithms.
- The LTE simulation model contains its own implementation of APIs with compatibility layer to allow the simulator to be independent from vendor-specific implementation of interface specifications based on FemtoForum [v1.].
- In real LTE network scheduling and radio resource management do not work with IP packets, but rather with RLC PDUs so the model should accurately model RLC functionality.
- The simulator scales up to tens of eNBs and hundreds of UEs and is possible to configure different cells with different carrier frequency and system bandwidth.

The LTE model architecture defined in ns-3 project is designed to be more representative of LTE architecture and standards discussed in chapter 2. The model includes Channel and Propagation model, Building model and Fading model provided by other modules to support channel propagation, building and shattering effects and fading effect respectively. In addition, it also includes the PHY layer models such as PHY Data and Control Channel Error Models, MIMO model. MAC layer models and functionalities including Resource Allocation model, QoS aware schedulers and AMC are also included. Moreover, it implements HARQ scheme, PDCP layer, RLC layer and RRC layer functionality and protocols, various types of handover and frequency reuse algorithms. The first part of the thesis work makes full use of this model.

EPC Model

The EPC model provides a simulation capability for end-to-end IP connectivity over the LTE model. It supports the interconnection of multiple UEs to the Internet, via a RAN of multiple eNBs connected to a single SGW/PDN-GW node. The only PDN type supported is IPv4 and the SGW and PDN-GW entities are implemented within a single node. Any regular ns-3 application working on top of Transmission Control Protocol (TCP) or User Datagram Protocol (UDP) can be used with EPC to model realistic applications. In addition, the model also allow UEs to use different applications with different QoS profiles and to perform handovers between eNBs. However, it lacks the accurate modelling of EPC control plane and hence uses a simplified way by leveraging on direct interaction among the different simulation objects via the provided helper objects. The second part of this work includes this model in addition to the LTE model.

4.1.2 Propagation Module

This module defines two generic interfaces, namely Propagation Loss Model and Propagation Delay Model, for the modeling of propagation loss and propagation delay respectively. The Propagation loss models calculate the received signal power considering the transmitted signal power and the mutual receiver and transmitter antennas positions. The two most common models used in ns-3 and this thesis work are briefly explained below.

Okumura Hata Model

This model is one of the most common propagation loss models for urban areas. It is designed for a frequency range of 150-1500MHz, distance of 1-100 Km and assumed eNB height of 30-100m. The pathloss expression of the standard Okumura Hata model in urban area is given as follows [plo]:

$$L(dB) = 69.55 + 26.16logf - 13.82logh_{\mathrm{b}} + (44.9 - 6.55logh_{\mathrm{b}})logd - C_{\mathrm{H}}$$

where for small or medium sized city

$$C_{\mathrm{H}} = 0.8 + (1.1logf - 0.7)h_{\mathrm{m}} - 1.56logf$$

and for large cities

$$C_{\mathrm{H}} = \begin{cases} 8.29(log(1.54h_{\mathrm{m}}))^2 - 1.1 & if 150 \leq f \leq 200 \\ 3.2(log(11.75h_{\mathrm{m}}))^2 - 4.97 & if 200 \leq f \leq 1500 \end{cases}$$

where f: frequency [MHz], h_{b}: eNB height above the ground [m], h_{m}: UE height above the ground [m], d distance [km] and $\mathrm{C_H}$ is the gain factor generated by the environment in which the system is operating.

COST231 Model

The European Union Forum for cooperative scientific research (COST) developed COST231 model according to various experiments and researches. This model extends the Okumura Hata Model for frequency range from 1500 MHz to 2000 MHz, h_{b} from 30m to 200m, h_{m} from 1m to 10m and link distance up to 20km [plo].

The pathloss expression of the COST231 is given as:

$$L(dB) = 46.3 + 33.9logf - 13.82logh_{\mathrm{b}} + (44.9 - 6.55logh_{\mathrm{b}})logd - F(h_{\mathrm{M}}) + C$$

where

$$F(h_{\mathrm{M}}) = \begin{cases} (1.1log(f)) - 0.7xh_{\mathrm{M}} - (1.56xlog(f) - 0.8 & \text{medium and small size cities} \\ 3.2(log(11.75h_{\mathrm{m}}))^2 & \text{for large cities} \end{cases}$$

$$C = \begin{cases} 0dB & \text{medium and small size cities} \\ 3dB & \text{for large citiess} \end{cases}$$

In addition to the above models ns-3 also includes International Telecommunication Union (ITU) propagation models such as ITU-R P.1411 and ITU-R P.1238. These models are desinged for line-of-sight and non-line-of-sight short range outdoor communication in the frequency range 300MHz to 100GHz.

4.1.3 Building Module

This module provides classes that models the presence of buildings in a simulation scenario and is designed with LTE in mind. It allows to specify the location, size and characteristics of buildings present in the simulated area, allows placement of UEs in side or outside of those buildings. In addition, it includes a path loss model called *Hybrid Buildings Propagation Loss Model* that allows to model the phenomenon of indoor/outdoor propagation in the presence of buildings. It is a combination of the above discussed path loss models in order to mimic different environmental scenarios such as urban, suburban and open areas and building types such as residential, office or commercial. Furthermore, it provides an additional set of building-dependent path loss model elements that are used to implement different path loss logics such as external wall loss, internal wall loss and shadowing effect.

4.1.4 Application Module

This module enables to implement realistic applications working on top of TCP or UDP such as UDP-client-server applications, packet sink application and On-Off applications on network nodes. We have implemented a packet sink application for the EPC-LTE simulation functionality on the second part of the work.

In addition to the above listed modules, other ns-3 modules such as network module, flow monitor module and mobility module are also used in the simulations to conduct experiment on LTE networks. Appendix B shows the UML class diagram of ns-3 LTE module.

4.2 Kodo Network Coding Library

Kodo is a C++ library for implementing erasure correcting codes, in particular RLNC. The library is intended to be used for reliable communication protocols and systems and for research on implementation of network codes. It supports various NC schemes such as Systematic RLNC, Non-Systematic RLNC and Sparse RLNC for various network scenarios. It is portable to wide range of platforms and operating systems including Windows 7, Windows 8.1, Mac OS X and Linux on PC and mobile devices, also different C++ compilers such as GCC 4.8, 4.9 and Clang3.5 [ste]. It is a header-only library and relies on a number of external libraries such as Cpuid, Boost, Fifi, Platform and Sak. These external libraries must be available and compile as static libraries in order to run an application with Kodo. To build Kodo and the dependencies a stand-alone build system called waf is used. waf is a build automation tool written in Python and designed to assist in the automatic compilation and installation of software.

4.2.1 Kodo Architecture

Kodo is based on a special C++ design technique known as mixin-layers that allows to create a component based library with high level of code reuse [kod]. Current Kodo[2] functionalities are independent of the lower layers and is implemented in the transport layer and the NC schemes behave in the same way if we change any of the lower layers, given that we are coding IP packets. This is one of the reason we chose to use Kodo to create coded packets in LTE networks.

Kodo exposes a simple API which allows a programmer to take advantage of NC in different applications. For researchers, Kodo's layered structure, shown in Figure 4.2, greatly simplifies the implementation of new and experimental RLNC variants, since typically only a new layer needs to be added. The functions of the most important API layers are summarized in the rest of the section.

- User API: This could be any function with any specific functionality such as debugging and can be implemented anywhere in the stack depending on the desired functionality.

[2] Kodo Version 19.0.0 at the time of writing this thesis

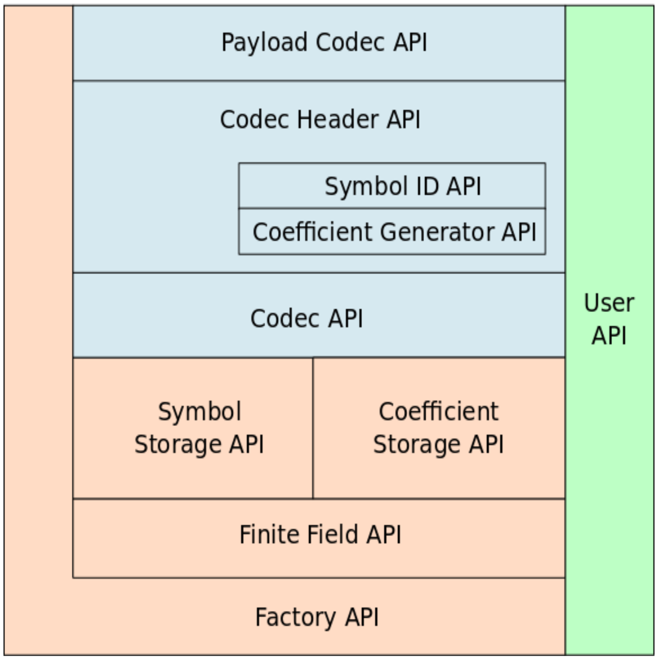

Figure 4.2: Overview of typical Kodo codec stack [kod]

- Payload Codec API: This provides a simple encoding and decoding APIs which either produces an encoded symbol or decode the symbol stored in a payload buffer.
- Coefficient Generator API: This API generates the coding coefficients used to produce and encoded symbol, it is the most important aspect of NC.
- Codec API: This specifies the functions needed when implementing a new encoder or decoder algorithm. This is where encoded symbols are produced through linear combinations of the original source symbols and where the operations implemented by the encoder is reversed producing the original source symbols at the decoder.
- Finite Field API: The finite field API layer provides the support of finite field mathematics. A fast implementation of finite fields is a prerequisite for fast RLNC.
- Factory API: The factory API defines the functions responsible for constructing and initializing a codec stack. All layers have access to the factory API, which can be used to preallocate memory for the different codecs and even share resources between codecs built using the same factory.

The second part of this thesis work uses the RLNC capability of Kodo library to create coded packets to send over the LTE networks.

Chapter 5
LTE Network Experiment and Analysis

In this chapter we will discuss the main factors affecting the LTE networks performance, a theoretical performance analysis of the LTE networks will be explained. Explanation of LTE experiment conducted, the topology and the simulation setup is briefly discussed. In addition, based on the obtained results, performance analysis and evaluation of the experimental LTE network setup is presented.

5.1 Factors Affecting LTE Network Performance

There are many factors that affect wireless networks performance that are dependent on various areas. For instance the technology of the devices used, the environment the signals will travel through, the fundamental concept behind wireless transmission and more.

Typically in a wireless communication system signals propagate via multiple paths between the transmitter and receiver with different attenuation and delay. The reason for this is the different distances between transmitter and receiver along different paths. As the signals propagate covering a wider areas, the signal spreads more and becomes much weaker. For certain transmitted signals the entire transmitted signal may experience fading due to destructive multipath cancellation which causes a severe drop in SINR. Propagation path loss is another factor that affects a transmitted signal, a received power level can greatly influenced by the distance between transmitter and receiver, the environment, the propagation medium and height and location of antennas. Another prominent effect is shadowing. Shadowing occurs due to building and other obstacles obstructing the line of sight path between the transmitter and receiver. This causes the received signal level to vary as the receiver moves around buildings.

In addition to the above well known factors, network and system architecture are also important factors which can affect the performance. For example the transmission power level and antenna configuration determines the coverage area. The capacity of a given network is affected by the number of simultaneous active users in a cell.

In the following sections we conducted performance analysis on how the above mentioned factors affect the overall performance of LTE networks both theoretically and experimentally.

5.2 Theoretical Throughput Performance Analysis

In order to fulfil the LTE goals of providing a higher data rates and better quality of service, it is very important to mitigate the above listed factors so that the overall network performance and user experience will be improved. For this purpose, LTE is based on new features such as OFDM in combination with higher order modulation techniques, large and scalable bandwidths, AMC and spatial multiplexing as discussed in section 2.2.

As shown in Table 2.2 LTE can be deployed with different bandwidths ranging from 1.4MHz to 20MHz, besides it is able to operate in different frequency bands shown in Appendix A [rtr]. It can operate in both paired and unpaired spectrum with OFDM that supports FDD and TDD operation. In ns-3 LTE only FDD is implemented and all the obtained results in this work are based on FDD operation mode.

To achieve the required datarates LTE provides Channel and Quality Indication (CQI) report scheme where the UEs can send information about how good or bad the communication channel quality is. The CQI values range from 1-15 as shown in Table 5.1 is depending on the SINR measurements at the UE. 15 being best channel quality and 0 the poorest channel quality [lp]. Each reported value of a CQI will initiate AMC scheme , where the modulation technique and coding scheme is changed to suit the channel condition as discussed in section 2.2.3. This in turn triggers a QoS aware scheduling and the network transmits data with different TB size and different code rate to keep the redundancy of the data sent at a reasonable level. For example if network gets high CQI value from UE, it transmit the data with larger TB size and high code rate and vice versa.

Moreover, in LTE the QoS aware scheduler is a key component for the achievement of a fast adjusted and efficiently utilized radio resource. In this experiment we have used a QoS aware scheduler called Priority Set Scheduler (PSS) from the ns-3 module. During TTI duration, i.e. 1 ms, the UEs report their perceived radio quality, as an input to the eNB scheduler to decide which AMC to use. Then the scheduler prioritize the QoS requirements amongst the UEs. After that it informs the UEs of allocated radio resources both on the downlink and on the uplink.

The theoretical achievable data rate in LTE is 300Mbps in the downlink and 75Mbps

Table 5.1: LTE standard CQI Table [lp]

CQI index	Modulation	Code rate x 1024
0	out of range	out of range
1	QPSK	78
2	QPSK	120
3	QPSK	193
4	QPSK	308
5	QPSK	449
6	QPSK	602
7	16QAM	378
8	16QAM	490
9	16QAM	616
10	64QAM	466
11	64QAM	567
12	64QAM	666
13	64QAM	772
14	64QAM	873
15	64QAM	948

in the uplink. Below we show steps how this can be achieved with 20MHz bandwidth, similar steps works for the other bandwidths also, approximate data rate values are summarized in Table 5.2.

- Step 1: for 20 MHz, there are 100 RBs and each RB have 12 subcarriers x 7 symbol x2 slots = 168 Symbols per ms in case of normal CP.
- Step 2: so there are 16800 Symbols per ms or 16800000 Symbols per second or16.8 Mbps for all 100 RBs.
- Step 3: assuming modulation used is 64 QAM with 6 bits code rate, throughput will be 16.8x6=100.8Mbps for SISO.
- Step 4: for 4x4 MIMO, the throughput will be 4x 100.8Mbps = 403.2Mbps.
- Step 5: deduct 25% for controlling and signalling overhead gives effective throughput of 300Mbps in the downlink, most UEs uses SISO so an effective 75Mbps in the uplink is expected.

Table 5.2: LTE standard bandwidth and corresponding approximate data rates for 2x2 MIMO system

Bandwidth (MHz)	1.4	3	5	10	15	20
Uplink Data rate(Mbps)	3	7.5	12.6	25.5	37.9	75
Downlink Data rate(Mbps)	8.8	22.4	36.7	73.7	110.1	300

Due to overheads from encryption, packet translation and partial utilisation of channel bandwidth for user data, theoretically advertised throughput can then be further reduced. Basically, the maximum advertised speeds are in general much higher than users will probably experience. Nowadays some manufactures are listing real world throughput on their products to give customers a better idea at what they can expect to achieve.

5.3 Experiment

In order to analyse and evaluate the QoS performance of LTE networks an experiment has been done on a simple topology which further will be discussed in the following sections. After retrieving experimental results it is then possible to analyse the possible performance of the LTE networks in real world scenario.

5.3.1 Topology and Simulation Setup

In order to include the real world performance factors of the wireless mobile communications the topology shown in Figure 5.1 considers the main factors such as the building, fading models and propagation loss. The topology of this simulation program is inspired from [GR] which specifies some of the parameters that should be used in order to simulate a topology in a typical urban area LTE network topology. The ns-3 LTE module provides the necessary tools called *helpers* and classes to setup the topology and carry out the experiment. Before we analyse and evaluate the results let see some of the helpers below.

LteHexGridEnbTopologyHelper

This helper class allows to easily create a topology with eNBs layed out on an hexagonal grid. This enables us to specify the number and the height of eNB and distance between the eNBs. In addition, we can use this helper to adjust the grid width, the number of eNBs along the x and y axis. In this setup we have put 2 eNBs in the X axis and 1 eNB in the Y axis with a distance of 500 meters between them.

UniformDiscPositionAllocator

In order to represent the real world placement of UEs around eNBS we have used this helper to enable the UEs to position themselves around the vicinity of the eNBs within a defined uniform disc radius.

GridBuildingAllocator

This class allows to create a set of buildings positioned on a rectangular 2D grid. With this class we have inserted a rectangular 300 x 200 x 10 building with 3 number

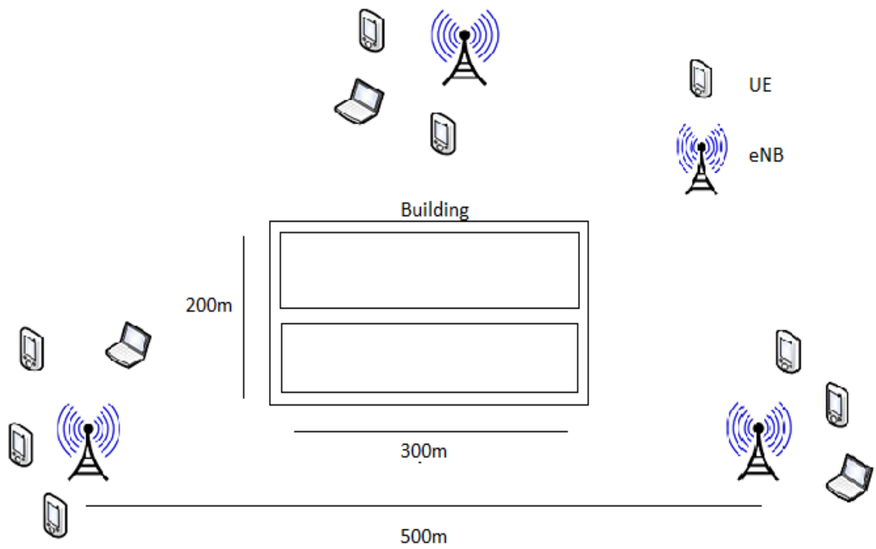

Figure 5.1: Simulation topology for LTE network performance analysis: Three eNBs and multiple UEs.

of floors and is placed between the eNBs in order to introduce the effect of building losses.

HybridBuildingsPropagationLossModel

This enables us to use a propagation model that includes combination of Okumura Hata model, COST231, ITU-R P.1411 and ITU-R P.1238 propagation models in order to be able to evaluate the path loss from 200 to 2600 MHz frequency ranges, in different environments with buildings and shadowing effect.

EpsBearer

This class helps to specify the traffic type and the corresponding bearer type. In this experiment video packets are used with a NGBR bearer type with QCI index 9, as shown Table 2.3.

The simulation setup parameters are summarized in Table 5.3. Some of the parameters are fixed throughout the simulation duration while some are changed for performance analysis purpose. The E-UTRAN Absolute Radio Frequency Channel Number (EARFCN) defined reflects the center frequency of an LTE standard bandwidths and does not take into account channel bandwidth. In our simulation 100 EARFCN corresponds to 2GHz carrier frequency, [rtr]. Another constant value is the shadowing effect standard deviation, which is used in the normal distribution used for calculation of the shadowing for UEs.

Table 5.3: LTE Simulation Setup Parameters

Parameters	Value
Simulation time (s)	0.25
eNBs Distance(m)	500
UE-eNB radius(km)	0 - 0.8
Distance dependent pathloss model	Hybrid buildings propagation loss model
Downlink Bandwidth (MHz)	1.4, 3, 5, 10, 15, 20
Resource Blocks	6, 15, 25, 50, 75, 100
eNB TX power (dBm)	30 - 60
Antenna type	2x2 MIMO, SISO
Fading model	Fast fading
Shadowing deviation	7
UE speed of interest (Kmph)	0, 3, 30, 60
Operating frequency band	2GHz
Downlink EARFCN	100
Traffic model	video
Video packet generation interval (ms)	100
Radio bearer type	NGBR
MAC Scheduler Type	PSS

5.4 Results and Analysis

The experiment is executed in a way that can help to analyse the performance of LTE network by varying different parameters such the distance between UEs and eNBs, UEs speed with corresponding fading models, number of UEs, system bandwidth and system antenna. In order to do that, each of the execution makes use of the topology in Figure 5.1 with the defined simulation parameters. Packets are sent from the eNBs to UEs in the simulation duration with the given interval. Thus, all the results obtained in this work are only for downlink. The reason for this is that with the growing interest of LTE network users analysis of the performance of downlink communication is more important than uplink and the traffic is usually higher in the downlink. In addition all results and analysis considers fading and shadowing effect so as to represent the real systems as close as possible in the simulation.

In this work the QoS performance of LTE networks is measured in the most important user service performance metrics , system throughput[1] and total packet delay. The system throughput is the sum of the data rates that are delivered to all UEs in the given network topology. The total packet delay is the sum of all delays for all received packets of the downlink flow. In addition, the spectral efficiency, coverage and serving capacity of the LTE network are analysed.

5.4.1 Fading and Shadowing Effect on Performance

Fading and shadowing effect on system throughput

The LTE module includes a trace-based fast fading model and a Matlab script for generating traces. The ns-3 Matlab script includes the typical taps configurations accroding to 3GPP specification TS36104 [Mata]. The fading evaluation during simulation run time is based on pre-calculated traces. This is done to limit computational complexity of the simulator. We have generated four fading trace profiles using the rayleighchan Matlab function [Matb]. The excerpt of fading for 3kmph urban scenario is shown in Figure 5.2.

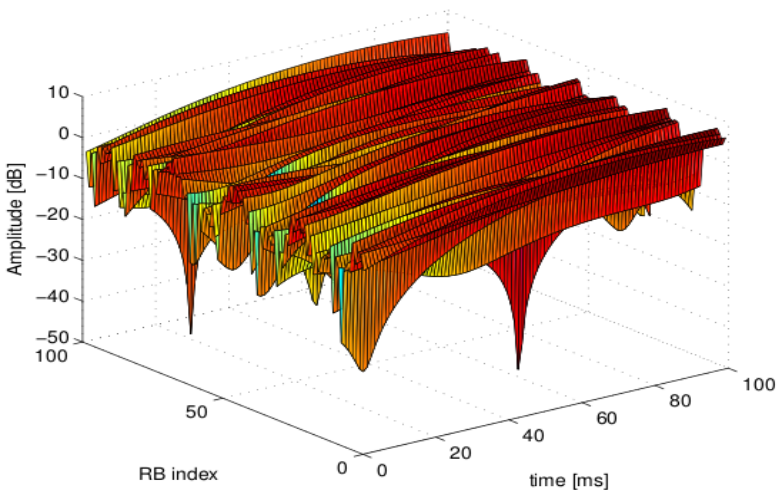

Figure 5.2: Fading excerpt of 3kmph

Figure 5.3 shows how the system throughput performance degrades due to fading and buildings shadowing for 5MHz system bandwidth, UE speed of 3Kmph and 40 number of UEs around a disc of radius 80m. Since the carrier frequency is 2GHz and is independent of the system bandwidth we have used 5MHz system bandwidth. So

[1] All throughput values are presented in logarithmic scale

the result obtained is valid for the other bandwidths also. The hybrid propagation loss model is set to evaluate the pathloss at a propagation frequency of 2.10GHz. As depicted in the figure the fading and shadowing cause poor performance as compared to the ideal transmission, i.e. with out considering fading and shadowing. The fading and shadowing results in a loss of signal power without reducing the power of the noise, as a result the system throughput is reduced.

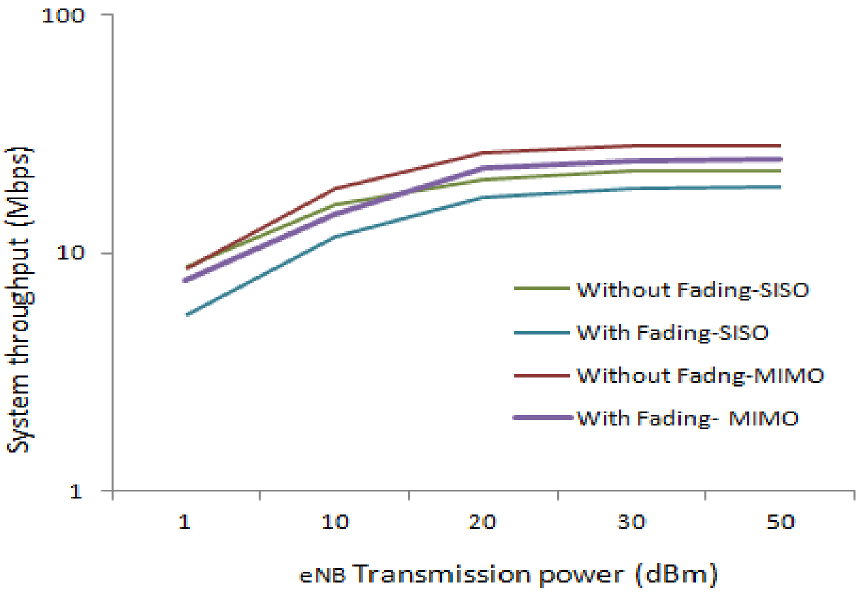

Figure 5.3: Effect of fading, buildings shadowing on system throughput: UE speed 3Kmph, 40 UEs, UE-eNB distance 80m and 5MHz bandwidth

One method of boosting the received signal power is increasing the eNBs transmission power. Figure 5.3 also shows how the system throughput is improved for high transmission power but still the effect of the fading is apparent. The effects of fading can also be combated by using diversity to transmit the signal over multiple channels that experience independent fading. Thus, LTE employs the MIMO system[2] where the subcarrier channels send data simultaneously in parallel with each other using multiple antennas, besides the OFDM capability of reducing the fading effect by using orthogonal subcarrier channel. From the figure we also see that with low eNB power transmission fading and shadowing reduced the MIMO system throughput resulting in less performance than SISO system. While for high eNB transmission

[2]MIMO system refers to the antennas used in each of the network devices, i.e. in UE, eNB and remote hosts

power the MIMO system outperforms SISO. Thus, to overcome this effects eNBs with MIMO system antenna and high transmission power is a requirement.

Fading and shadowing effect on total packet delay

Figure 5.4 shows the variation on the total packet delay[3] of the received packets for SISO and MIMO systems under the fading and shadowing effect. As we can see, the total packet delay is less with MIMO systems. This indicates MIMO improves not only the system throughput but the network latency also. However, the MIMO shows improvement both in throughput and delay in case where the transmission power is higher and when the network can take advantage of multiple transmissions. Thus, there is a trade off between the eNBs transmission power and the MIMO systems deployment. Therefore, For low power transmission in a short distance or networks with less load using SISO systems might be a better choice than MIMO, since MIMO systems have their own limitation as discussed in section 2.2.2. On other hand, for large networks with high traffic load and high propagation loss and fading effects MIMO could be a better option.

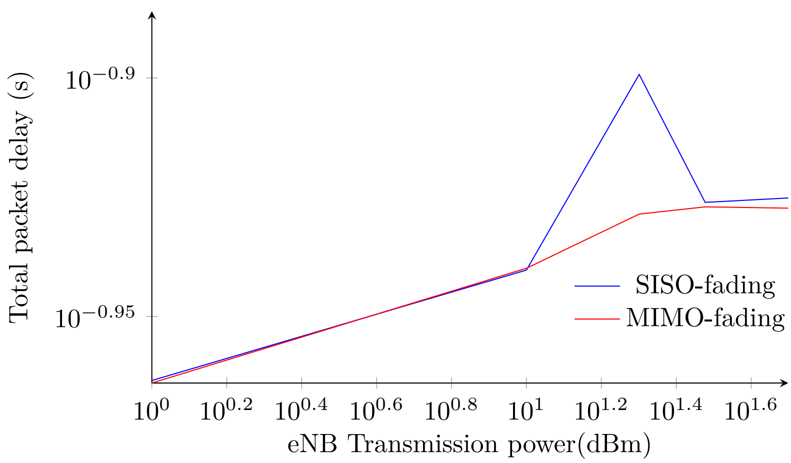

Figure 5.4: Effect of fading, buildings shadowing on total packet delay: UE speed 3Kmph, 40 UEs, UE-eNB distance 80m and 5MHz bandwidth

5.4.2 UE Speed Effect on Performance

System throughput vs UE speed

One of the deciding factors for how much fading can affect the strength of a radio signal is the speed of the transmitting or the receiving end. In this subsection we

[3] Delay value is presented in logarithmic scale

carried out an experiment to show how the speed of UEs can affect the system throughput. In order to do this, as mentioned above we have generated four different fading trace profiles at 0, 3, 30 and 60 kmph speed for UEs moving at speed of 0, 3, 30 and 60 kmph respectively. The ns-3 LTE module supports an urban scenario simulation with speed of 0, 3, 30 and 60 kmph so we chose these range of speed value.

When a UE is moving, the UE's velocity causes a shift in the frequency of the signals transmitted along each signal path. As a result the magnitude of the change in amplitude will vary, a phenomenon known as the Doppler shift. Thus, signals traveling along different paths with different speed can have different Doppler shifts, corresponding to different rates of change in phase. The difference in Doppler shifts between different signal components contributes to a signal fading at the receiving end.

Figure 5.5 shows how UEs moving with different speed affects the system throughput, for all the standard system bandwidths. The result is obtained for SISO system antenna, 40 number of UEs around a disc of radius 80m, 30dBm eNB transmission power. As the UEs travels with high speed the vulnerability to multipath fading due to Doppler shift increases resulting in reduced system throughput. The speed affects the system throughput regardless of the system bandwidth used.

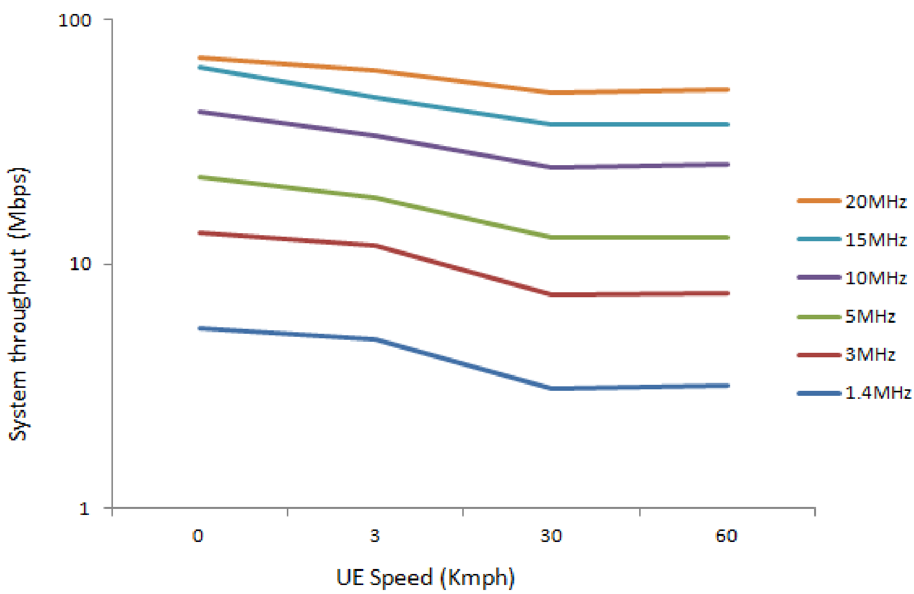

Figure 5.5: Effect of UE speed on system throughput: SISO system, 40 UEs, 80m UE-eNB distance, 30dBm eNB Transmission power

Total packet delay vs UE speed

Figure 5.6 shows UEs speed effect on the total packet delay[4] for different system bandwidths. The simulation parameters used are the same as above. As shown in the figure, the total delay increases as the UEs travel with increasing speed due to the degradation in the received SINR level, for the same reason discussed above. Thus eNBs has to send more data using the HARQ scheme to compensate the loss due to fading at high speed and as a result the delivery of packets is delayed which increments the total delay. From the graph we can also observe that the total packet delay becomes less with higher bandwidths. This is due to the fact that with the given number of UEs the the wider bandwidths can serve the active UEs quickly and with less congestion than the narrow bandwidths. This is further discussed in section 5.4.4.

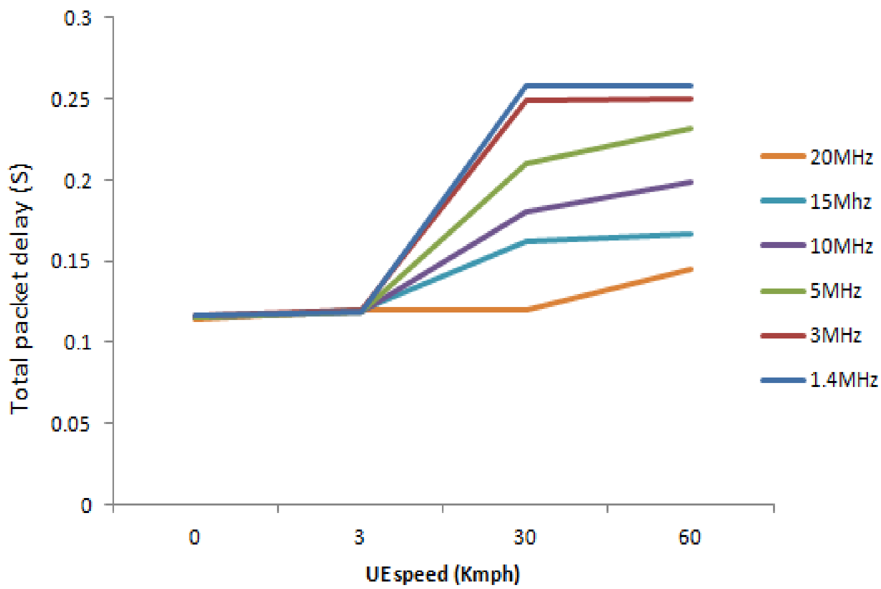

Figure 5.6: Effect of UE speed on the total packet delay: SISO system, 40 UEs, 80m UE-eNB distance, 30dBm eNB Transmission power

5.4.3 UE-eNB Distance Effect on Performance

Throughput vs UE-eNB Distance

Wireless mobile communication is very much distance dependent, as discussed in 5.1, to show this a simulation is conducted with 30dBm eNB transmission power,

[4] Delay values are presented in linear scale in the rest of this chapter

SISO system antenna and 40 UEs around the three eNBs. In order to observe the effect of distance between eNBs and UEs we have assumed speed of UEs as 0kmph with the corresponding fading trace of 0kmph. The distance is defined as the radius of the uniform disc around the eNBs, since the UEs are randomly placed by the *UniformDiscPositionAllocator* in the disc, thus there is no specific position for a specific UE.

Figure 5.7 shows how the system throughput is affected by the eNBs-UEs distance for different system bandwidths. As shown in the figure the system throughput is highly affected as the distance increases. The strength of radio signals degrades as the distance between the transmitter and receiver increases due to the propagation losses they face from buildings and terrain along the way. As shown in the figure as UEs appear further from the eNBs the system throughput is reduced regardless of the system bandwidth used. For instance at 2km distance the narrow system bandwidth such as 1.4, 3 and 5 MHz are severely affected, while 10, 15 and 20 MHz comparatively provides a reasonable throughput compared with the other technologies. For distance greater than 2km the wider system bandwidths can still provide a system throughput which is better than other generation technologies, but with the goal of the LTE this might not be a desirable system throughput.

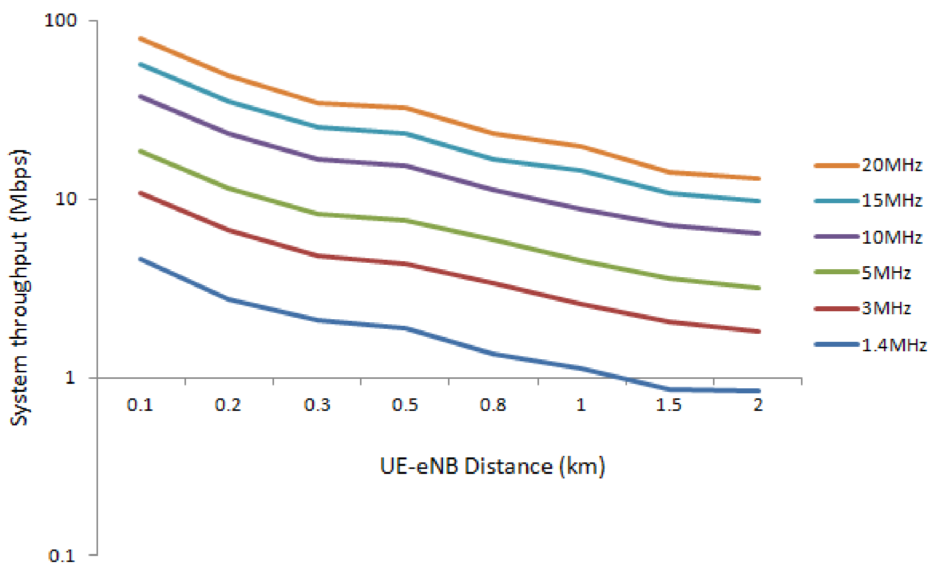

Figure 5.7: Effect of UE-eNB distance on system throughput: SISO, 40 UEs, 30dBm eNB Transmission power and 0kmph UE speed.

Spectral Efficiency vs UE-eNB Distance

Figure 5.8 shows the spectral efficiency of different system bandwidths. The spectral efficiency shows us how efficiently a specific bandwidth is utilized by the network. For all the system bandwidths, the spectral efficiency is reduced with increasing distance. We can also see that the wider bandwidths has better spectral efficiency than the narrow bandwidths regardless of the distance.

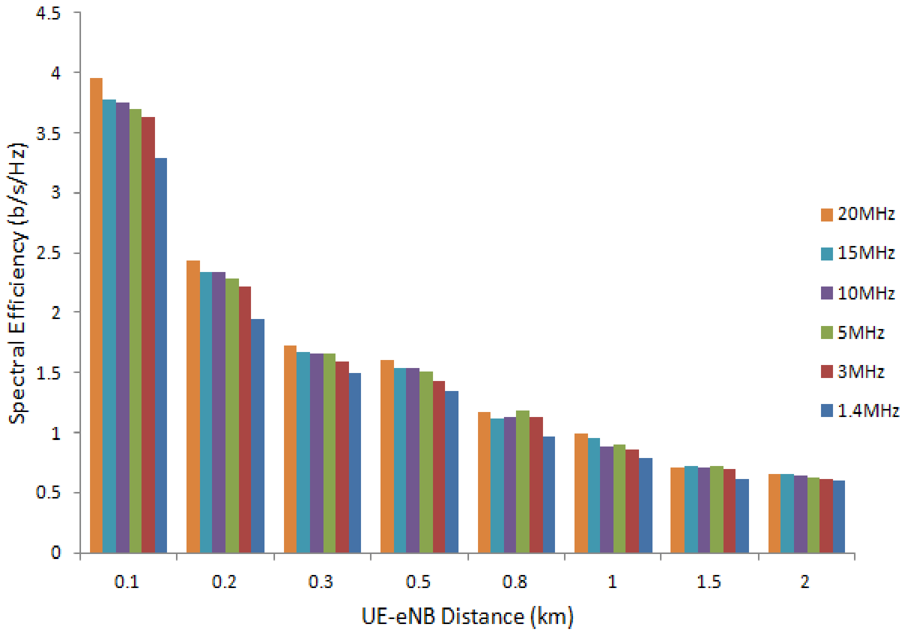

Figure 5.8: Spectral efficiency of LTE system bandwidths: SISO, 40 UEs, 30dBm eNB Transmission power and 0kmph UE speed.

5.4.4 Number of UEs Effect on Performance

System throughput vs Number of UEs

One of the key parameters that affect the performance of a mobile communication networks is the number of simultaneous active users in the network, i.e. the traffic load. The traffic load usually determines the network operators bandwidth, the transmission power and the QoS and more. In this section we simulate the same topology as above with SISO system, 30dBm eNB transmission power, 3kmph UE speed at a distance of disc radius 80m to observe and analyse the system throughput, total packet delay and the serving capacity for different number of users/UEs. The number of UEs is the total number of UEs trying to connect to one of the three eNBs (offered traffic), so anyone of the eNBs can serve any number of UEs. As the simulation runs UEs are distributed randomly among the three eNBs.

Figure 5.9 shows the variation of the system throughput for different number of UEs. The system throughput is reduces with the increased number of UEs in the LTE network. The reason for this is the congestion in the network, since the system bandwidth has to be shared by the UEs located in the area. We have observed that as the number of UEs increase, the network only serves a limited number of UEs (carried traffic) by dropping some of the UEs. For 5, 10, 15 and 20MHz, the maximum carried traffic is approximately 300 UEs with reduced system throughput. While for 1.4 and 3MHz the maximum carried traffic is approximately 125 and 170 respectively but with a much reduced system throughput. However, the indicated system throughput are achieved at a cost of high total packet delay.

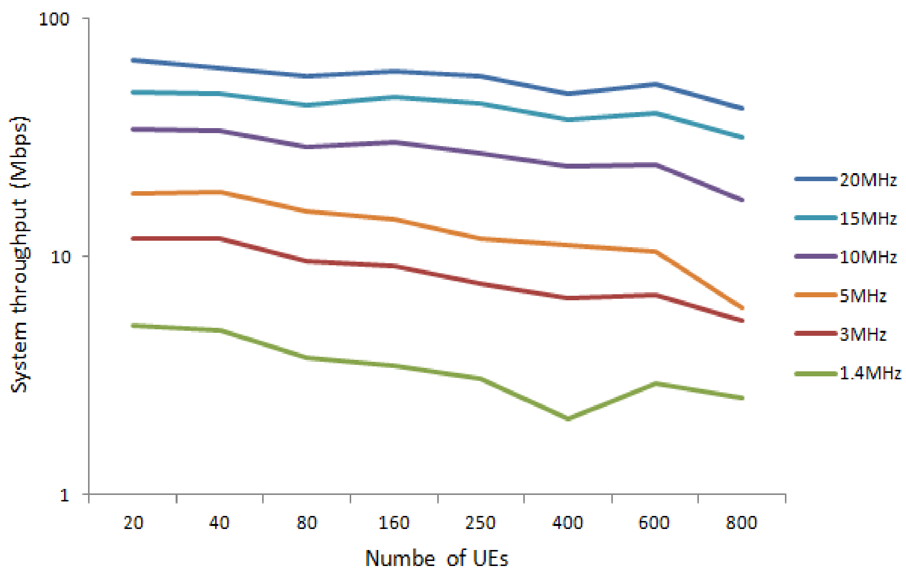

Figure 5.9: Effect of number of UEs on system throughput:SISO, 30dBm eNB transmission power, 3kmph UE speed and 80m UE-eNB distance.

Capacity vs Number of UEs

Figure 5.10 summarizes the serving capacity of each of the system bandwidths represented as a percentile of the carried traffic and offered traffic. The total carried traffic is represented with the bar while the percentile of carried traffic is represented by the lines. In this figure we can see that up until 400 offered traffic the carried traffic increases, this means the number of dropped UEs are less. As for UEs greater than 400, for 1.4 and 3MHz bandwidths the carried traffic is reduced, which indicates

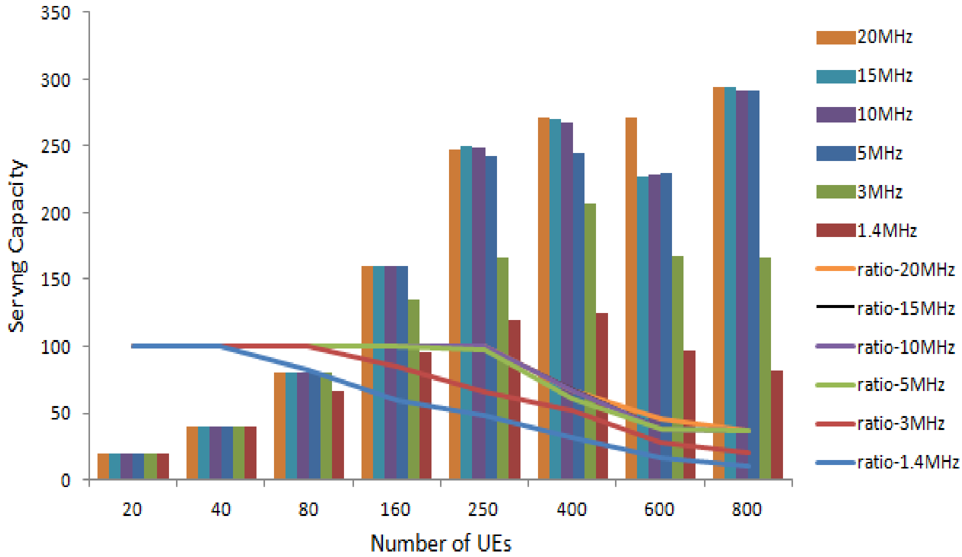

Figure 5.10: Serving capacity of LTE system bandwidths: SISO, 30dBm eNB transmission power, 3kmph UE speed and 80m UE-eNB distance.

that the maximum offered traffic for stable network operation to achieve an optimum serving capacity. For 5, 10, 15, 20 MHz the network can be stable as long as maximum offered traffic is 600. It is important to notice that with increasing number of total active UEs, the system throughput is much less and not all the UEs are served, as shown in Figure 5.10.

Total packet delay vs Number of UEs

Figure 5.11 shows the total packet delay for all the LTE defined bandwidths. As the figure shows, up to 400 offered load the delay increases as the number of UEs increase with corresponding system throughput. But for more than 400 UEs the eNBs start to drop the UEs due to spectrum congestion and the delay becomes low since packets are received by less number of UEs. In addition, the figure also shows that for more than 600 UEs the network is unstable, i.e. even for fewer number of UEs the delay is high. As we compare the total packet delay, it is less in the narrow bandwidths than in the wider bandwidths. This is because the carried traffic in narrow bandwidths is less which corresponds to the number of packets received in the network, which in turn determines the total packet delay.

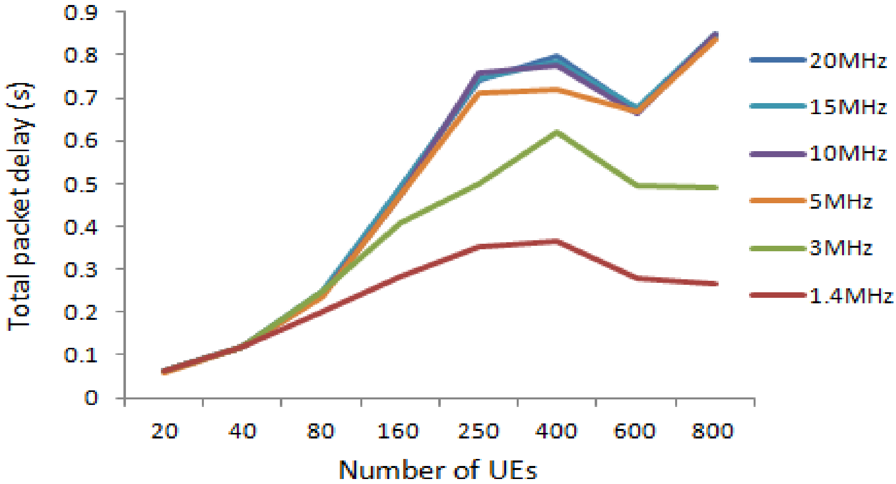

Figure 5.11: Effect of number of UEs on total packet delay : SISO, 30dBm eNB transmission power, 3kmph UE speed and 80m UE-eNB distance.

Chapter 6

LTE Network Experiment and Analysis with Network Coding

In this chapter we introduce NC into LTE networks, many theoretical and experimental analysis has been done on deploying NC for wireless networks. However, as LTE is a new trend in the mobile communication networks there has not been many practical works. Thus, in this chapter we simulate a simple scenario in ns-3 to evaluate the performance of LTE network with RLNC. In addition, comparison between deploying RLNC and other alternative options to attain improvements in system performance are made. The two variants of RLNC are also compared, systematic and non-systematic.

6.1 Experiment

The author previously conducted an AL-RLNC experiment on wifi X-topology and has proven system performance improvements. Especially the throughput, end-to-end packet delay and packet overhead. In addition, in many other works it has been shown that RLNC have their own disadvantages such as operation complexity and energy efficiency. Thus we wanted to see the effect of RLNC at the application layer for LTE networks.

In order to deploy RLNC in LTE networks, we use ns-3 and Kodo library to build the required simulation scenario and implement the encoding and decoding functionality respectively. We also make use of EPC model of LTE module in addition to the LTE model to analyse the QoS performance.

6.1.1 Topology

Figure 6.1 shows the topology of the LTE network used for simulation and QoS performance analysis. It consists of a remote host located in external PDN network, SGW/PDN-GW as one entity, an eNB and a UE.

The topology is chosen to include a remote host located in external network and only one UE and single eNB for two reasons. First the Kodo library allows to encode and decode packets at the application layer in which it demands to create TCP/UDP sockets in order to transmit and receive IP packets. In LTE access network radio communication is used to send traffic between UE and the eNB. UE is assigned an IP address the time it enters the EPS network and all the packets between a remote host and UE are carried by the SGW/PDN-GW which acts as a default gateway for the UE in order to make an analogy with the traditional IP world. Second the LTE-EPC simulation module in ns-3 only supports a point-to-point connection between remote host and UEs/eNBs located in different networks, it does not support broadcast topology. Therefore it is important to have a topology that includes the requirements from Kodo library and ns-3 module. However, for the LTE QoS performance in chapter 5 the LTE module supports the broadcast topology between eNBs and multiple UEs within the access network.

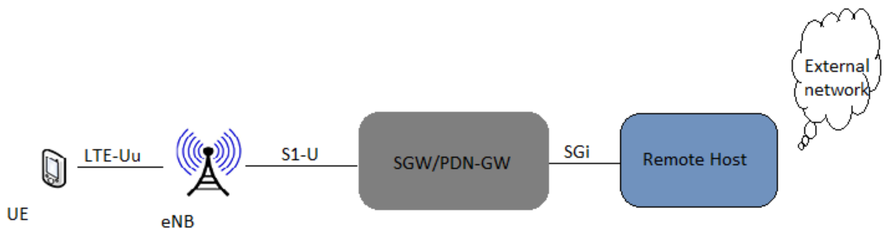

Figure 6.1: LTE-EPC simulation topology. The remote host sends coded packets to UE through the SGW/PDN-GW

The application module of ns-3 is used to create two applications that are used to encode packets at the remote host and decode packets at the UE. On each of these application a socket is created using a *socket factory* class in ns-3 and uses the IP addresses of respective devices to transmit and receive application layer IP packets. To enable end-to-end LTE-EPC simulation the ns-3 module provides a *ltehelper: enableEPC* helper so that the LTE access network can communicate with other external network that contains the remote host. In addition, the ns-3 module provides *point-to-point model* to create a point-to-point connection for the remote host through the SGW/PDN-GW. Figure 6.2 shows the overall steps to create and simulate the topology shown in Figure 6.1.

In this part of the thesis work the QoS performance is evaluated and analysed downlink end-to-end, from the remote host to the UE and is referred as QoS performance in the rest of the document.

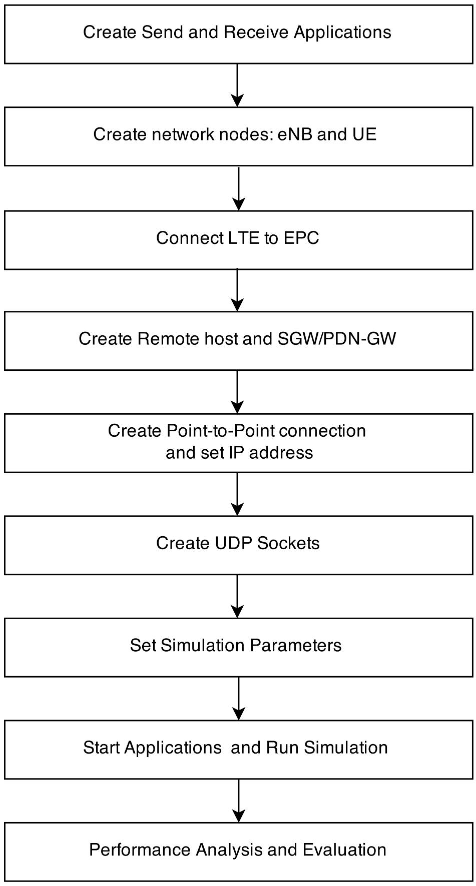

Figure 6.2: Overall system flow for creating the LTE-EPC topology

6.1.2 Architecture

Figure 6.3 shows the overall architecture of the topology in Figure 6.1. The AL-RLNC scheme is performed on the application layer on top of the existing HARQ scheme to enhance the performance of the LTE network, as briefly explained in section 3.2.1.

In the simulation, the S1-U interface is modeled in a realistic way to encapsulate IP packets, as done in real LTE-EPC systems. Downlink coded IP packets are generated from the application installed in the remote host, these packets are created from one large file. The packets are coded using the Kodo library's RLNC scheme and

APIs by specifying the generation size, packet size and Galois field, as explained in section 3.1.1 and 4.2.1. Internet routing will take care of forwarding the packet to the SGW/PDN-GW node which is connected to the Internet by the SGi interface. SGW/PDN-GW determines the eNB to which the UE is attached, by looking at the UE IP destination address. It classifies the packets using TFT to identify to which EPS bearer it belongs, each EPS bearer has a one-to-one mapping to S1-U interface. Following the mapping it sends the coded packet to the S1-U interface then to the intended eNB. Upon reception of the coded packets, the eNB forwards it to the UE over the LTE-Uu interface based on the TEID. Finally the UE will receive the encoded packets and the application installed on the UE checks whether the received packets are linearly independent or not. It drops linearly dependent packets and stores the linearly independent packets in a matrix form and decode them according to decoding algorithm discussed in section 3.1.1. The encoder/remote host keeps transmitting coded packets until it gets a notification from the decoder/UE that signals the reception of sufficient linear independent packets to recover the whole file. In parallel, the HARQ performs its functions independent of the application layer, treating the encoded packets as normal packets and sending a retransmission request, initiated from the channel condition CQI value, upon failure to receive packets.

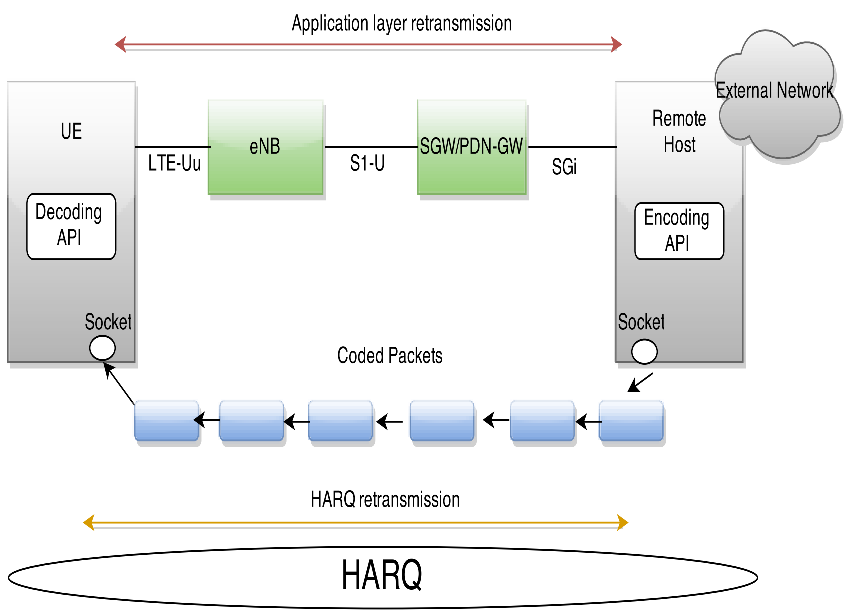

Figure 6.3: Architecture showing how coded packets are sent down to the UE using RLNC

6.1.3 Simulation Setup

In order to simulate and collect data for the performance analysis and evaluation of the defined network topology we have set the simulation parameters as shown in Table 6.1, which is similar to Table 5.3 with some changed parameters to fit the topology. As shown in the table an EPS bearer is established instead of radio bearer. In addition to radio bearers S1 bearer is necessary to classify the packets they cross the EPC core towards the UE in the LTE access network. Another changed parameter is the simulation time, as we can see from the table the simulation time is 20 seconds. As described above in 6.1.2 application are installed in the remote host and UE in order to transmit coded packets along the LTE network. For this purpose we set the run-time of the applications same as the simulation duration so that enough packets will be generated, coded and decoded with in the given duration with a packet generation interval of 100ms. Furthermore, the distance dependent propagation model used is COST231 propagation loss model which is one of the constituting components of Hybrid propagation loss model. Since we did not include any buildings in these topology we did not use the hybrid propagation loss model. However, all the fading and distance related factors are included as shown in the table. Table 6.1 also shows the parameters used for creating the coded packets. Galois Field GF(256) is used to generate the coding coefficients to increase the probability of successful decoding. Data is encoded with systematic RLNC and sent as a packet size of 1024 bytes with 100 generation size. We chose to use systematic RLNC because we only have one UE and one remote host.

6.2 Result and Analysis

The QoS performance is evaluated and analysed by executing the given topology with simulation parameters for two execution rounds, one for encoded packets and second for normal packets. Accordingly comparison is made between the two executions on the throughput and delay performance of the network. Additional comparison is made between systematic and non-systematic RLNC to evaluate the performance of the two schemes. Alternative system improvement method, MIMO, is also discussed by comparing it with AL-RLNC.

Since we only have one remote host and one UE the throughput is the rate of successful data delivery over the given channel. The delay is the sum of end-to-end delay for all the packets sent from remote host to UE until successful decoding of the packets. For this purpose we have used *flow monitor module* from the ns-3 library, additionally it is used to observe the total number of lost and received packets.

Table 6.1: LTE-EPC Performance Simulation Setup Parameters

Parameters	Value
Simulation time (s)	20
Application run-time (s)	20
Distance dependent pathloss model	COST231 propagation loss model
Antenna type	2x2 MIMO, SISO
Fading model	Fast fading
Shadowing deviation	7
UE speed of interest (Kmph)	0
Operating frequency band	2 GHz
Downlink EARFCN	100
Traffic model	video
Video packet generation interval (ms)	100
EPS bearer type	NGBR
MAC Scheduler Type	PSS
Encoding scheme	Systematic RLNC
Galois field	GF(256)
Generation Size	100
Packet Size (byte)	1024

6.2.1 Throughput Performance Analysis

To analyse the throughput performance with NC in LTE networks, 100 packets of size 1024 byte, which is 100x1024 byte data, is encoded and send down to the UE from the remote host. The simulation is performed for three different eNB transmission powers, UE-eNB distance of 0.3km to 4km, system bandwidth of 5MHz and SISO system antenna. This is done because in order to observe the effect of NC we want to consider a lossy link where the UE is located at a reasonable distance to make sure packets are lost. If the UE is close enough to the eNB there will not be packet loss since it is only one UE for the given system bandwidth. Since we only have single UE the throughput performance will not be affected by system bandwidth. As a result 5MHz is chosen to do the performance analysis.

Figure 6.4, Figure 6.5 and Figure 6.6 shows the difference in the throughput[1] performance obtained with and without NC, for 30, 46 and 60dBm eNB transmission power respectively. As we can observe from these figures the throughput is higher

[1] The throughput values are given in logarithmic scale

with NC. With NC, packets are sent as a combination using the RLNC, which means every linearly independent packet received at UE decoder have some portion of the transmitted data. This enables the decoder to recover lost packets. Besides it is performed on top HARQ which by itself recovers lost packets with retransmission requests. In these figures, the lowest value of the x-axis distance indicates that with the given power level all the packets sent are fully recovered up until that distance. However, as we increase the distance of the UE packets will be lost. For instance, for 30dBm power the throughput is same with and without NC up until 0.3km and as packets begin to lost the throughput drops but with better performance with NC.

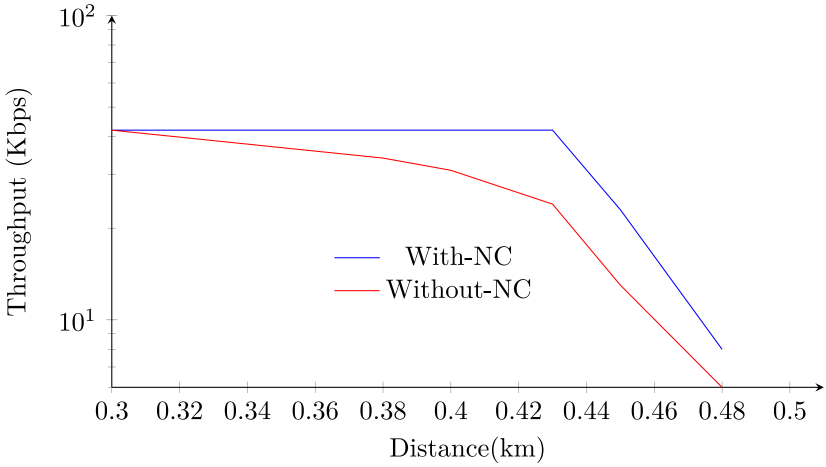

Figure 6.4: Throughput performance for 30dBm eNB transmission power with and without NC: SISO, 5MHz bandwidth

The reason to use different eNB transmission power levels is to see the effect of NC at different distance and to show how the performance can be improved not only with NC but also by increasing the transmission power. As we can see from these figures with increasing transmission power we can achieve better throughput performance in LTE networks. For example the throughput and coverage that can be obtained with 30dBm and NC can easily be achieved by increasing the transmission power to 46dB. However this is only for the downlink communication where eNB power transmission is not a big concern, but for UEs increasing the transmission power could have its own effect on the battery life of the UE. Thus, we can say that with out having to deal with the complexity of deploying new RLNC scheme on the existing HARQ scheme the required performance can be achieved with increasing the transmission power.

As shown in these figures the throughput obtained is in Kbps. This is because we only generated 100 packets of size 1024 bytes for 20s simulation and application run time using the application created on remote host and UE. Besides the goal of this part of the thesis work is to show how NC affects the LTE networks. Where as the system throughput obtained in section 5.4 are in Mbps, since the ns-3 LTE module is designed to generate packets and simulate the transmission as a real system would do without having the user to create the encoding or decoding applications in UE or remote host.

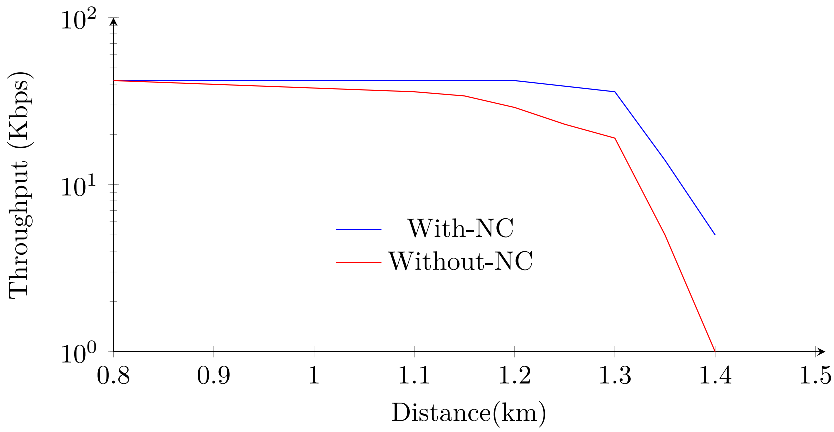

Figure 6.5: Throughput performance for 46dBm eNB transmission power with and without NC: SISO, 5MHz bandwidth

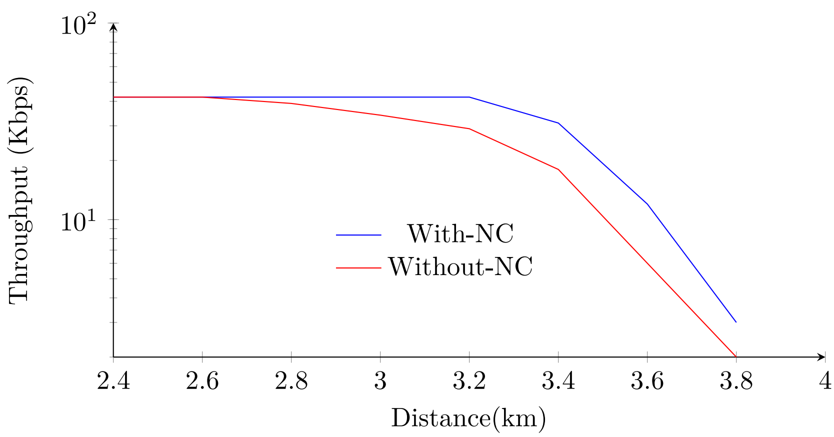

Figure 6.6: Throughput performance for 60dBm eNB transmission power with and without NC: SISO, 5MHz bandwidth

6.2.2 Delay Performance Analysis

Figure 6.7, Figure 6.8 and Figure 6.9, shows the delay[2] performance at 30, 46 and 60 dBm eNB transmission power levels. As shown the delay is higher with NC. This is because in NC in addition to the transmission delay, the delay includes the time to encode and decode a packet. From these figures we also see how the delay reduces for higher distance between the UE and eNB. This is in line with the decreasing throughput which indicates the total number of received packets are less so is the total delay.

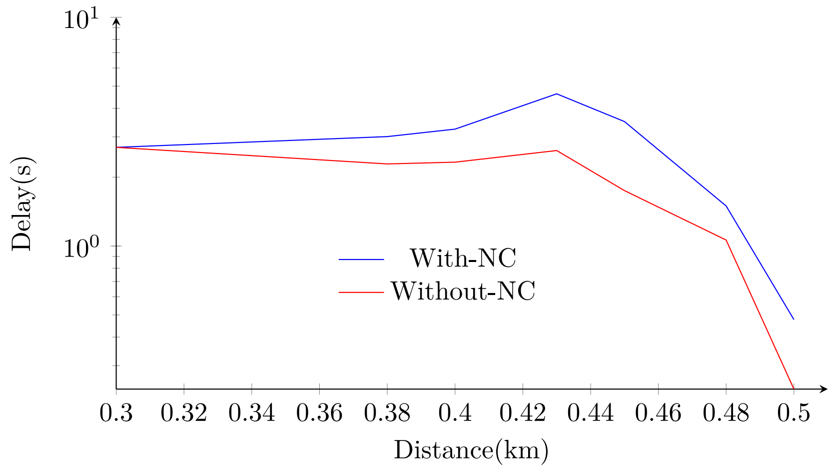

Figure 6.7: Delay performance for 30dBm eNB transmission power with and without NC: SISO, 5MHz bandwidth

[2]The delay values are given in logarithmic scale

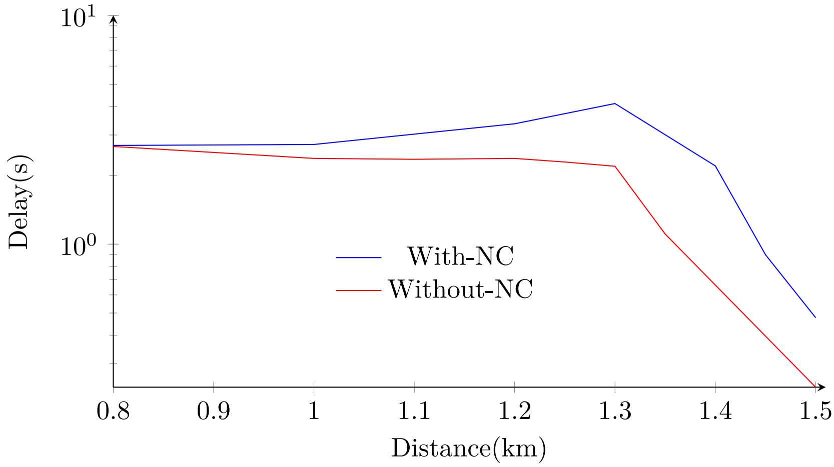

Figure 6.8: Delay performance for 46dBm eNB transmission power with and without NC: SISO, 5MHz bandwidth

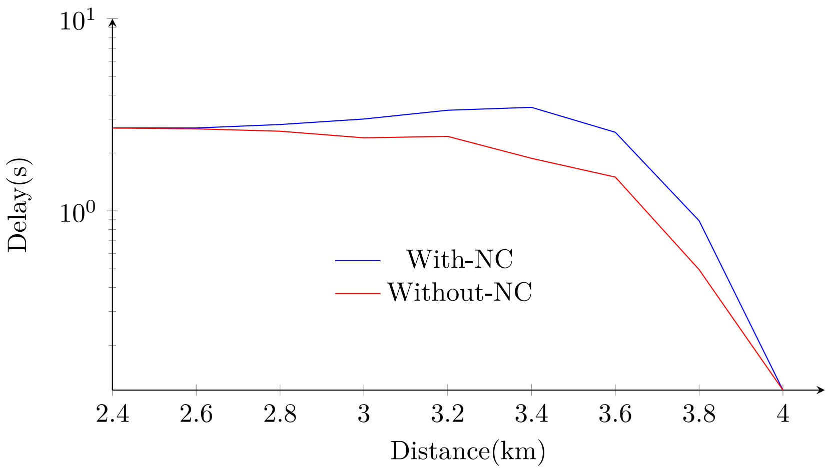

Figure 6.9: Delay performance for 60dBm eNB transmission power with and without NC: SISO, 5MHz bandwidth

6.2.3 MIMO vs NC

As discussed in previous chapters, MIMO improves the overall system performance with its own drawbacks. In this part it is shown that MIMO could be an alternative to deploying NC in LTE network. Figure 6.10 shows the throughput obtained for 60dBm eNB transmission power for MIMO and SISO systems with and without NC. As shown in the figure the throughput obtained with MIMO system without using NC is higher at any given distance as compared with the SISO with NC. From the result we can also see that the MIMO system increases the coverage of the network up until 5km. Thus, we can expect a better throughput performance of LTE networks by combining MIMO with NC.

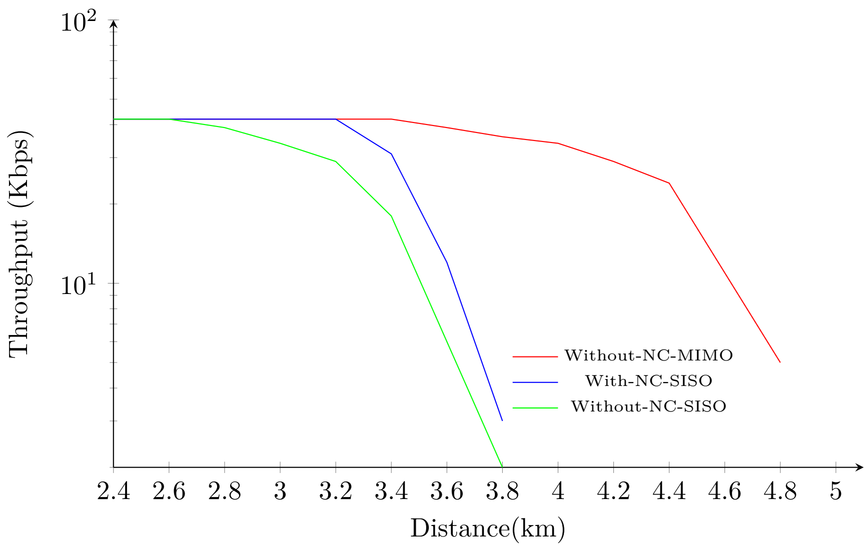

Figure 6.10: MIMO and SISO with NC performance comparison for 60dBm eNB transmission power and 5MHz bandwidth

6.2.4 Systematic vs Non-systematic Analysis

Figure 6.11 shows the performance of LTE network for the two variants of RLNC discussed in section 3.1.1. The result is obtained for 60dBm eNB power and 5MHz bandwidth. This figure shows the number of transmissions required to attain the corresponding throughput. Up until 3.2km we can see that systematic coding have less total number of transmitted packets than non-systematic coding to achieve the same throughput. This is because in the systematic scheme uncoded packets are transmitted and received in the UE without any loss. While in the non-systematic case the encoder/remote host sends coded packets, which are combination of different packets and each coded packets might not have all the information required. As a result more transmission are necessary to have the complete data even though there is no loss until this distance.

Moreover, as the distance increases, systematic RLNC still has a better throughput even though it is with similar total number of packet transmissions as non-systematic. This implies that both schemes achieve the corresponding throughput with more retransmissions but the systematic scheme benefited more from transmitting uncoded packets before the packets begins to loss. In the figure the maximum number of transmitted packets, including the retransmissions, is 200. This is due to the simulation time and the packet generation interval which are 20s and 100ms respectively.

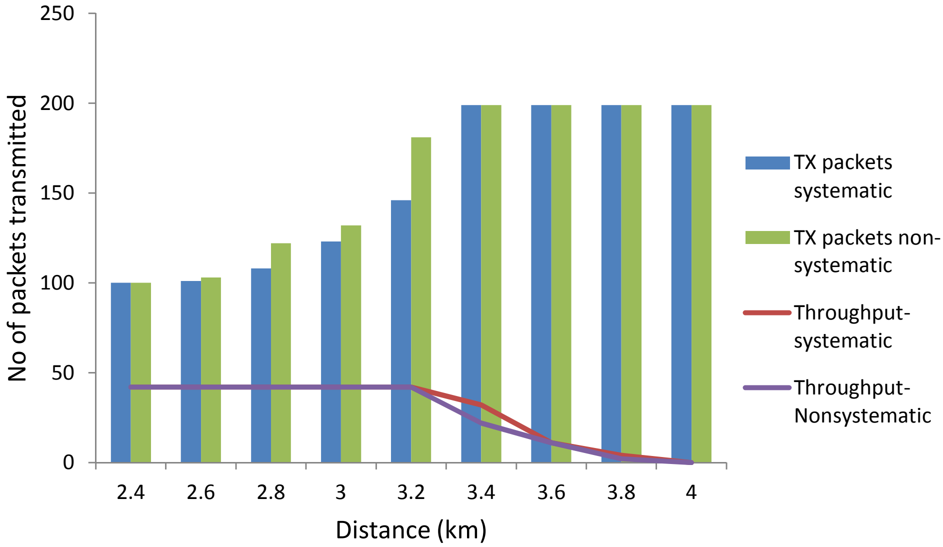

Figure 6.11: Systematic vs non-systematic: Throughput and total packet transmission number performance comparison for SISO, 60dBm eNB transmission power and 5MHz bandwidth.

Figure 6.12 shows the delay[3] and code rate performance of both of the schemes. As we can see from the figure, the delay is higher for non-systematic coding for distances until 3.2km and becomes lower than the systematic RLNC as the UE gets far. The respective reasons for this are; in the first case where the packets loss ratio is low , encoding packets adds unnecessary delay during non-systematic while in the second case as the packets loss ratio is higher the systematic RLNC result in higher throughput and we can see the corresponding end-to-end packet delay is also higher. The figure shows the code rate, which is the ration of the useful information obtained from the total number of packet transmissions. As we can expect due to the overhead

[3]Delay value is given in linear scale

for encoding and decoding the code rate is lower in non-systematic than systematic.

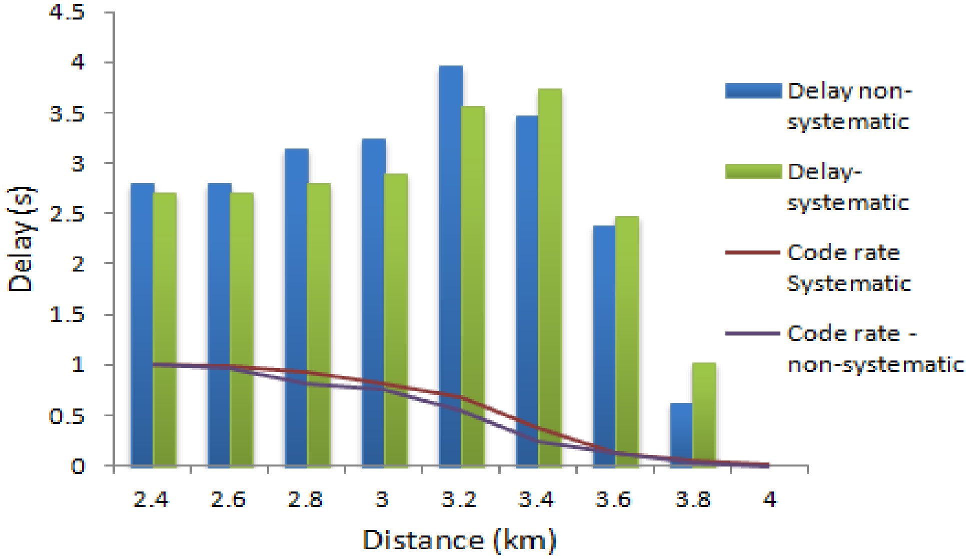

Figure 6.12: Systematic vs non-systematic: delay and code rate performance comparison for SISO, 60dBm eNB power and 5MHz bandwidth.

Chapter 7
Discussion

In this thesis work, we have presented the theoretical study on the architecture and performance of LTE networks and conducted an experiment in order to evaluate and determine the performance of LTE networks. We have also done the integration of NC, specifically the RLNC scheme on the application layer so as to observe and analyse the effect on the overall network performance. In this chapter, we discuss the results obtained in the previous chapters for LTE access network and the LTE-EPC network with RLNC.

7.1 LTE Access Network QoS Performance

In this section the main discussion points will focus on the key QoS performance parameter metrics of the LTE access network. These performance metrics include the system throughout and the total packet delay. In addition the most important performance aspects of LTE networks such as the spectral efficiency, capacity and coverage are explained. In the results we found, we have shown that the performance of the LTE networks depends on various factors, the nature of radio propagation being the most important one, regardless of the features that are considered in the 3GPP 4G network architecture. These factors include the fading, shadowing, distance dependent path losses, the distance and the speed at which a UE travels, the number of UEs and the eNB transmission power level.

Fading and Shadowing

The fading and shadowing from buildings, specially in urban areas, has been shown to be the main bottlenecks in the attempt to provide the theoretical data rates despite the flexible bandwidths used in LTE. As the results indicate these inevitable factors reduces the system throughput and increases the network latency by degrading the strength of the received signals at the receiving end. With the advantage of using OFDM in mind, it is shown that increasing the eNB transmission power and/or using a MIMO system antenna to enhance the transmission and reception capability are some of the ways to reduce the effect of fading and shadowing. Network operators may or may not fancy these methods because this comes with the cost of deploying eNB antennas with high efficiency and high energy consumption. However with the current technological advancement in electronic devices with high operational and processing efficiency at low energy consumption these methods could still be considered as optional approaches for downlink communication.

UE Speed

The speed at which a UE moves determines the fading profile. As indicated in the obtained results whenever the speed increases the impact of the corresponding fading profile has shown to affect the system throughput and the total packet delay. The results were obtained for 0, 3, 30 and 60 kmph, in reality the speed at which a UE travels may vary in these ranges considering a pedestrian and a vehicular speed for example. Thus we can say that the results obtained gives a fairly close performance with the real UE speed and can be used to predict the system performance by network operators. In addition, the results also show that the system throughput is similarly affected in all the standard system bandwidths while the main difference is on the total packet delay, which is very high in narrow bandwidths. However, it is possible to say that it is better to use the wider bandwidths than the narrow bandwidths since we can still achieve a reasonable system throughput with less total packet delay at any given UE speed.

Distance

Another factor considered is the distance between the UE and eNB. Theoretically it is proven that as the distance between transmitter and receiver varies the SINR also varies. The experimental results also show similar phenomenon in which the system throughput is reduced as the UEs are far from the eNBs. For the given simulation parameters, one can observe that the maximum distance achievable is 2km. Even though it is possible to improve the coverage with wider bandwidths and higher transmission power it may not be the performance users would expect form using LTE network. In addition, due to the distance factor the spectral efficiency also varies, the system bandwidths are efficiently used within distance of about 0.1km. The spectral efficiency decreases as expected with increasing distance independent of system bandwidth used. This indicates network operators need to consider other system performance factors to actually determine the coverage of LTE networks other than using wider system bandwidth, for example they should consider the number of UEs as explained next.

Number of UEs

The number of UEs is one of the key factors that determines QoS of a data service provided by a network operators. From the results, increasing the number of UEs affects the overall performance of the networks and inturn the services provided to a user. The result also indicates there is a limit to the total number of active UEs that a given system bandwidth can serve in order to provide the QoS at the desired level. For the given set of simulation parameters, the wider system bandwidths can serve up to 300 UEs simultaneously sharing the system bandwidth with reduced system throughput and increased total packet delay. While with the narrow bandwidths the maximum number of UEs is approximately 170 with much reduced system throughput and high total packet delay. In addition, the results also shows the maximum offered traffic in the area of coverage in order to have a stable network operation with the expected spectral efficiency. For instance in the wider bands, when the offered traffic is higher than 600 UEs the network tends to serve only a few users and drop most of them due to the congestion in the spectrum. As a result the spectrum efficiency is also low. Thus, when considering deploying an LTE network it is important to to first determine the capacity of the network in terms of offered and carried traffic for a specific eNB transmission power level and system bandwidth.

7.2 LTE-EPC Network Performance with Network Coding

Performance Comparison with and without Network coding

The results obtained from deploying AL-RLNC in LTE shows us that using NC in 4G networks improves the throughput and coverage of the network for different eNB transmission powers. However, the throughput improvement comes with an increment in network latency. With the high network latency, using AL-RLNC may not be an ideal solution specially for some real time services with low delay requirement. The higher network latency obtained by deploying the RLNC in the application layer is due to the fact that packets are encoded and decoded in the application layer and there is no functional combination between HARQ and RLNC. Our suggestion is if encoding and decoding the packets is done on the MAC layer where HARQ functionality is combined with the RLNC a better network latency performance can be obtained.

In many theoretical and practical works and the author's previous work, it has been shown that with NC high throughput can be achieved with low network latency in other wireless networks. In addition, it has been proven that in NC the broadcast nature of wireless networks and the recoding features of intermediate nodes are the main factors that result in overall system performance. Even though we were not able to exploit these benefits of wireless networks by setting up a broadcast LTE network topology, due to the limitation in simulation capability of ns-3. Similar conclusion can be made about LTE networks with remote host broadcasting to a number of UEs, since the nature of the LTE network is quite similar with the other wireless networks.

Performance Comparison MIMO vs NC

As explained in previous discussions the use of MIMO improves the throughput and coverage of 4G networks. Similarly we have shown that as an alternative for NC, performance of the network can be improved with MIMO system antennas. In the downlink communication, network operators and vendors can choose to implement NC at each end devices, i.e. UE and remote host, and the total power requirement will be shared among these devices or they can have the MIMO system antenna in eNBs, remote host and UEs with additional power requirement in each device for MIMO operation. In terms of feasibility implementing encoding and decoding functionality at each end device may not be easier due to the complexity and energy consumption of coding techniques. On the other hand, deploying MIMO in each end device might not be battery efficient but with the current rapid evolution of digital electronic chips this might be a feasible solution. In addition combination of both techniques can be considered for QoS performance optimization of LTE networks.

Performance of RLNC variants

In practical deployment of NC it is important to not only implement the functionality but also to choose the the better coding technique depending on the network topology of interest. In our case for the given topology the systematic RLNC performed better in terms of the throughput, the delay and with low overhead than the non-systematic due to the simplicity of the network. In many other theoretical works it has been shown that, in wireless networks with multiple nodes non-systematic performs better. This is because sending encoded packets at the beginning of transmission can actually increase the decoding probability and the intermediate nodes can also decode, recode and send to a nearby node with out having to interact with the source node. Thus, for LTE simple topologies systematic RLNC performs better and for large and complex networks with multiple nodes non-systematic could perform better.

Chapter 8 Conclusion and Further Work

8.1 Conclusion

In this thesis work we presented a theoretical background study on the architecture and the main features of 4G/LTE access network. The EPS architecture and entities including the functional split is briefly discussed. In addition, LTE networks QoS provisioning and enforcement, protocol stack and the concept behind the NC and implementation options are explained. An experimental analysis on the QoS performance of LTE networks in the RAN and the performance with RLNC in the LTE-EPC core network is given. ns-3 and Kodo libraries has been used to create and simulate an experimental scenarios to represent a real LTE network and to collect and analyse the results obtained.

The conducted experiments achieved valuable results for performance assessment and analysis which can be used to predict the performance aspects of LTE networks before deployment. The results obtained demonstrated the key performance parameters in LTE networks to fulfill the goal of 3GPP in 4G networks. The QoS performance is measured and evaluated in terms of system throughput, total packet delay, spectral efficiency, coverage and serving capacity by varying parameters such as the number, distance and speed of UEs. In addition to these parameters, changes in the network component and resource such as using MIMO and different system bandwidth or transmission power are considered. Moreover, main factors such as fading, buildings shadowing and path loss that can affect the SINR of radio signals have been considered in the experiment and analysed under different assumptions.

The present thesis introduced the concept of NC into LTE networks by integrating them in the application layer on top of HARQ. Results are analysed and evaluated to compare the performance improvements that can be achieved with this new scheme and to consider other system improvement techniques such as deployment of MIMO systems. In addition two variants of RLNC are implemented and the result of each variant have been compared in regard to different network topology.

8.2 Further work

This section presents some points to consider in future works that can be done in continuation of this work.

In the thesis work all the eNBs used are the typical outdoor-mount-type. Recently the deployment of indoor eNBs is becoming a new thread and are considered to be a replacement for wifi networks, thus conducting the performance analysis with these kind of eNBs would be a good idea.

In this work NC is introduced at the application layer and another scheme at the MAC layer is discussed, the MAC-RLNC. This scheme requires a modification to the LTE protocol stack in order to implement RLNC in the MAC layer. There are some theoretical works related to this scheme but not sufficient practically proven ones, thus this can be included in future work in roder to compare with the AL-RNC. In addition the topology used in the second part of the work is a very simple one due to the simulation library limitation. In reality multiple remote hosts broadcast to multiple UEs are implemented with in the LTE-EPC network. Thus analysing the performance with the broadcasting feature can show the performance in more detail and realistic way. For this, simulation library other than ns-3 that can support broadcasting feature and network coding libraries and coding schemes that can be deployed in layers other than application layer will be a requirement.

In the present work the performance is done only for the downlink communication, and uplink communication can also be included in the future. In addition, in wireless network nodes can benefit by sending and transmitting from nearby nodes not only from source to destination, a phenomena called user cooperation. With user cooperation and NC better performance can be achieved by exploiting user diversity and by decoding and recoding the combined packets at the intermediate nodes.

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A Standard LTE Bands

E-UTRA Operating Band	Uplink (UL) operating band BS receive UE transmit F_{UL_low} – F_{UL_high}	Downlink (DL) operating band BS transmit UE receive F_{DL_low} – F_{DL_high}	Duplex Mode
1	1920 MHz – 1980 MHz	2110 MHz – 2170 MHz	FDD
2	1850 MHz – 1910 MHz	1930 MHz – 1990 MHz	FDD
3	1710 MHz – 1785 MHz	1805 MHz – 1880 MHz	FDD
4	1710 MHz – 1755 MHz	2110 MHz – 2155 MHz	FDD
5	824 MHz – 849 MHz	869 MHz – 894MHz	FDD
6[1]	830 MHz – 840 MHz	875 MHz – 885 MHz	FDD
7	2500 MHz – 2570 MHz	2620 MHz – 2690 MHz	FDD
8	880 MHz – 915 MHz	925 MHz – 960 MHz	FDD
9	1749.9 MHz – 1784.9 MHz	1844.9 MHz – 1879.9 MHz	FDD
10	1710 MHz – 1770 MHz	2110 MHz – 2170 MHz	FDD
11	1427.9 MHz – 1447.9 MHz	1475.9 MHz – 1495.9 MHz	FDD
12	699 MHz – 716 MHz	729 MHz – 746 MHz	FDD
13	777 MHz – 787 MHz	746 MHz – 756 MHz	FDD
14	788 MHz – 798 MHz	758 MHz – 768 MHz	FDD
15	Reserved	Reserved	FDD
16	Reserved	Reserved	FDD
17	704 MHz – 716 MHz	734 MHz – 746 MHz	FDD
18	815 MHz – 830 MHz	860 MHz – 875 MHz	FDD
19	830 MHz – 845 MHz	875 MHz – 890 MHz	FDD
20	832 MHz – 862 MHz	791 MHz – 821 MHz	FDD
21	1447.9 MHz – 1462.9 MHz	1495.9 MHz – 1510.9 MHz	FDD
...			
23	2000 MHz – 2020 MHz	2180 MHz – 2200 MHz	FDD
24	1626.5 MHz – 1660.5 MHz	1525 MHz – 1559 MHz	FDD
25	1850 MHz – 1915 MHz	1930 MHz – 1995 MHz	FDD
...			
33	1900 MHz – 1920 MHz	1900 MHz – 1920 MHz	TDD
34	2010 MHz – 2025 MHz	2010 MHz – 2025 MHz	TDD
35	1850 MHz – 1910 MHz	1850 MHz – 1910 MHz	TDD
36	1930 MHz – 1990 MHz	1930 MHz – 1990 MHz	TDD
37	1910 MHz – 1930 MHz	1910 MHz – 1930 MHz	TDD
38	2570 MHz – 2620 MHz	2570 MHz – 2620 MHz	TDD
39	1880 MHz – 1920 MHz	1880 MHz – 1920 MHz	TDD
40	2300 MHz – 2400 MHz	2300 MHz – 2400 MHz	TDD
41	2496 MHz 2690 MHz	2496 MHz 2690 MHz	TDD
42	3400 MHz – 3600 MHz	3400 MHz – 3600 MHz	TDD
43	3600 MHz – 3800 MHz	3600 MHz – 3800 MHz	TDD
Note 1: Band 6 is not applicable			

Figure A.1: Standard LTE operating bands

B ns-3 UML class diagram

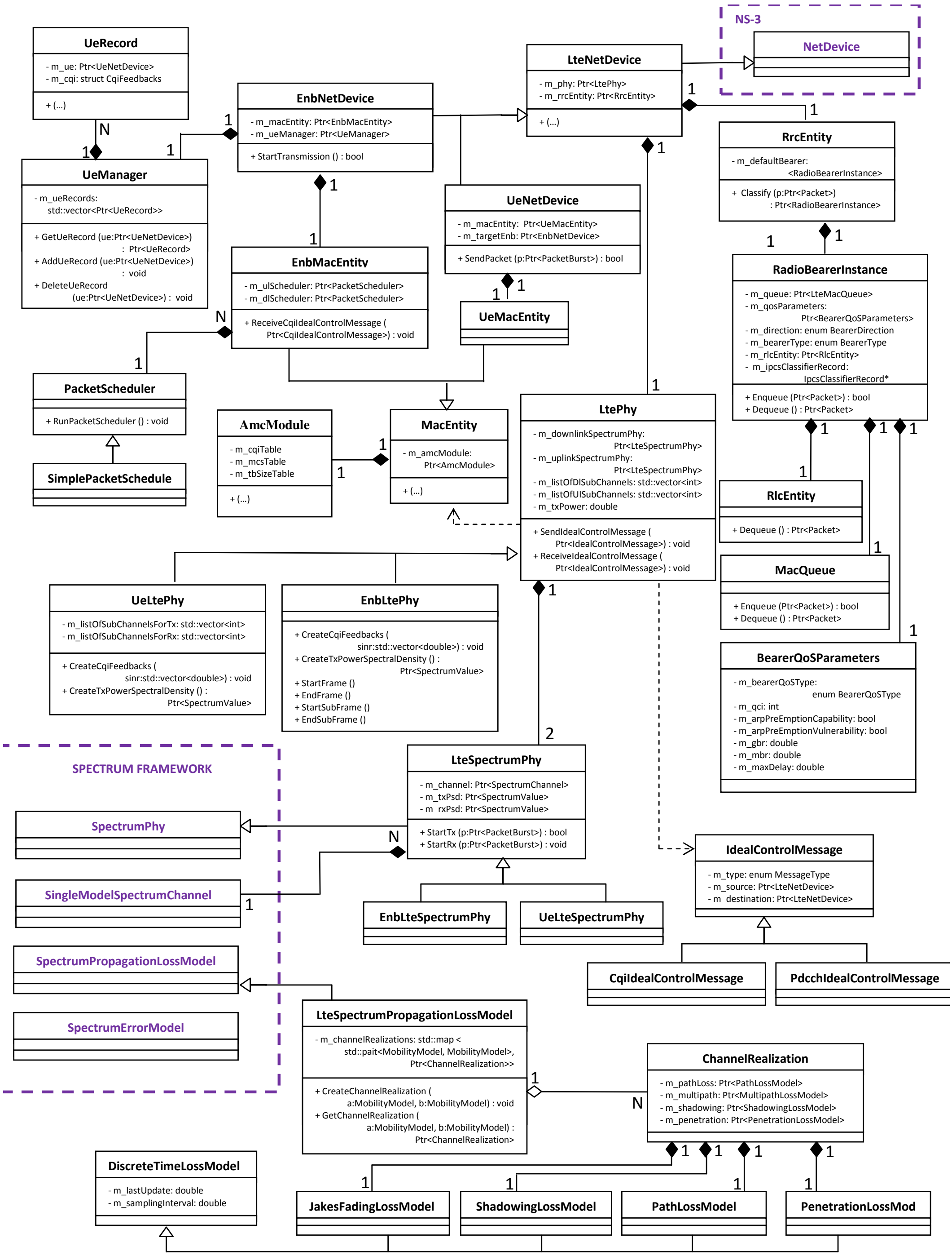

Figure B.1: ns-3 UML class diagram

C
Conference paper draft

Performance Analysis of LTE Networks with Random Linear Network Coding

Tewelde Degefa Assefa, katina Kralevska and Yuming Jiang, *Professor*
Norwegian University of Science and Technology, NTNU, 2015.

***Abstract*—Random Linear Network coding (RLNC) has recently been investigated as a promising solution for reliable multimedia delivery over wireless networks. As the demand for vast multimedia service delivery increases over the 4G wireless cellular standards such as Long Term Evolution (LTE), novel content aware transmission techniques are needed. In this paper, we investigate Application Layer-RLNC (AL-RLNC) as one such promising technique, building upon the author's recent work on AL-RLNC integration in Wifi X-topology. A simple implementation scenario composed of a User Equipment (UE), an eNB and a remote host is considered for integration on top of the state-of-the-art Hybrid Automatic Repeat Request (HARQ) in LTE network. Our result shows that AL-RLNC improves throughput and coverage at a cost of higher packet delay. In addition, we compare AL-RLNC performance with advanced LTE system antenna technique called Multiple-Input Multiple-Output (MIMO).**

***Keywords*—RLNC, AL-RLNC, LTE, MIMO, HARQ**

I. Introduction

The rapid increase of mobile data usage and emergence of new applications such as online gaming, mobile TV, streaming contents have motivated the 3GPP to work on the LTE on the way towards 4G mobile networks. LTE is the air-interface for the 4G wireless systems promising high data rates and small transmission delays to meet the increasing users demands. It supports scalable carrier bandwidths, from 1.4 MHz to 20 MHz and supports both Frequency Division Duplexing (FDD) and Time division Duplexing (TDD). Moreover, it has the ability to manage fast-moving mobiles and supports multicast and broadcast streams. It is based on Orthogonal Frequency Division Multiplexing (OFDM) in combination with higher order modulation techniques. In LTE large bandwidths and spatial multiplexing techniques are used in the downlink to provide downlink peak rates of 300 Mbps and uplink peak rates of 75 Mbps with Quality of Service (QoS) provisioning capability and Adaptive Modulation and Coding (AMC) capabilities.

Despite the fact that the LTE physical layer represents the state-of-the-art in communications technology, novel solutions for adaptation to mutimedia services are needed. Recently RLNC has emerged as a new paradigm to exploit broadcasting nature of the wireless medium to improve the throughput in the wireless networks. It is becoming a promising approach for a low complexity, adaptive and content aware packet level error resilience solution. Given the unique flexibility of RLNC to efficiently bridge the upper layer media packetization and the lower layer wireless packet transmission, RLNC is considered as a powerful cross layer solution for reliable mutimedia delivery over the LTE [1] [2] [3].

The paper is organized as follows. In section II, we provide a short introduction to AL-RLNC. Section III describes the experimental topology, architecture, simulation setup and system flow. In section IV we discuss the results obtained in the experiment and we present the comparison between RLNC and MIMO system deployments. Conclusion and further work are given in section V.

A. Related work and Paper Contributions

This work is influenced by recent theoretical and practical investigation of RLNC integration in LTE networks in [6] [4], especially the practical work on integration of RLNC in the MAC layer in [7]. To the author's knowledge there are no many practical works done on integration of AL-RLNC into LTE networks. This paper implements AL-RLNC in LTE networks and evaluates the performance improvements that can be obtained.

II. Application layer - RLNC

In AL-RLNC the standard LTE protocol layer stack remains unchanged, shown in figure 1. It is deployed on top of the existing MAC layer HARQ based packet transmission process. In this solution, RLNC encoded IP packets enters the eNB PDCP layer. PDCP layer performs header compression and ciphering then the PDCP encapsulated IP packets are delivered to the RLC layer. The RLC layer performs segmentation/ concatenation of IP packets into RLC packets to fit the MAC frame size requirements. Each MAC frame is allocated a single physical layer transport block for transmission over the eNB/UE interface. The physical layer carries all the information from the MAC transport channels over the air interface. In addition takes care of the link adaptation for realizing link adaptation functionality in the MAC layer to provide a matching of the modulation and coding techniques to the radio link interface condition [1][3]. In this work application layer packets are coded using RLNC scheme from Kodo network coding library.

III. Experiment

In order to deploy RLNC in LTE networks, we use ns-3[1] and Kodo[2] library to build the required simulation scenario

[1]A C++ based open source simulation library for networking research. At the time of writing ns-3 version 3.22

[2]A C++ library for implementing erasure correcting codes. At the time of writing Kodo version 19.0.0

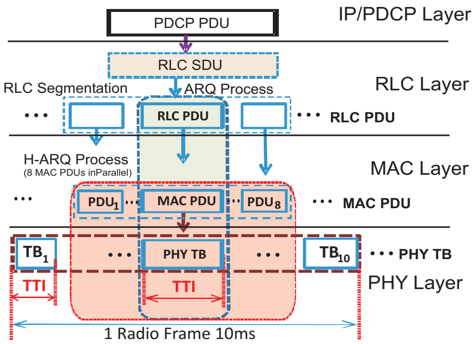

Fig. 1. eNB RAN protocol : AL-RLNC on top of MAC-HARQ solution

and implement the encoding and decoding functionality.

A. Topology and Architecture

Figure 2 shows the topology of the LTE network used for the performance analysis. It consists of a remote host located in external network, SGW/PDN-GW as one entity, eNB and UE. Two applications are implemented at the remote host and the UE for encoding and decoding packets respectively. On each of the applications a UDP socket is created to transmit and receive the encoded IP packets over the LTE network. Coded packets are generated from a large file with a specified generation size, packet size and Galois field using the RLNC scheme in Kodo library.

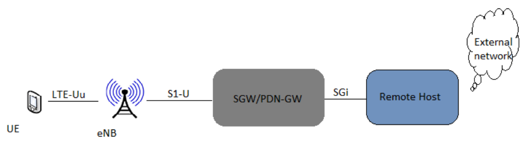

Fig. 2. LTE-EPC simulation topology. The remote host sends coded packets to UE through the SGW/PDN-GW

The architecture shown in Figure 3 indicates how the encoded packets are transmitted from the remote host and decoded at the receiver, UE. As shown in this figure Internet routing will take care of forwarding the packet to the SGW/PDN-GW node which is connected to the Internet by the SGi interface. SGW/PDN-GW determines the eNB to which the UE is attached, by looking at the UE IP destination address. It classifies the packets using traffic flow templates to identify to which Evolved packet system (EPS) bearer it belongs, each EPS bearer has a one-to-one mapping to S1-U interface. Following the mapping it sends the coded packet to the S1-U interface then to the intended eNB. Upon reception of the coded packets, the eNB forwards it to the UE over the LTE-Uu interface based on the bearer ID. Finally the UE will receive the encoded packets and the application installed on the UE checks whether the received packets are linearly independent or not. It drops linearly dependent packets and stores the linearly independent packets in a matrix form for decoding. The encoder/remote host keeps transmitting encoded packets until it gets a notification from the decoder/UE that signals the reception of sufficient linear independent packets to recover the whole file. In this process the HARQ performs its functions independent of the application layer, treating the encoded packets as normal packets and retransmission upon failure to receive packets is initiated from the channel condition.

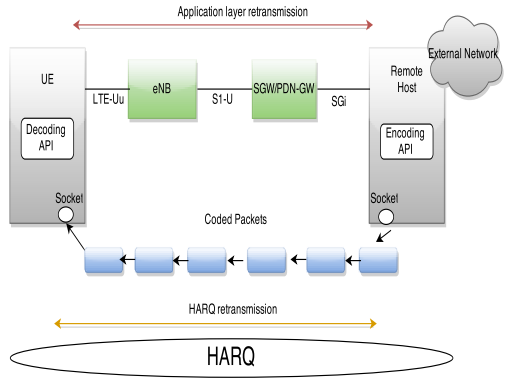

Fig. 3. Architecture showing how coded packets are sent down to the UE using RLNC

Figure 4 shows the overall steps for creating the simulation architecture.

B. Simulation Setup and System Flow

In order to simulate and collect data for the performance analysis and evaluation of the defined network topology we have set the simulation parameters as shown in table I. As shown in the table an EPS bearer is established to classify the packets according to TFTs and the defined QoS class, as they cross the EPC core towards the UE in the LTE access network. The simulation time, is 20 seconds, we set the run-time of the application on remote host and UE to be same as the simulation duration so that enough packets will be generated and encoded with in the given duration at a packet generation interval of 100ms. Furthermore, the distance dependent propagation model used is COST231 propagation loss model[3]. In addition, a fast fading model generated with a Rayleigh channel is included in order to consider the impact of fading on radio signals. The table also shows the parameters used for

[3]COST231 propagation loss model model is applicable to urban areas to evaluate path loss of radio signals in frequency range 1500 MHz to 2000 MHz and link distance of up to 20 km.

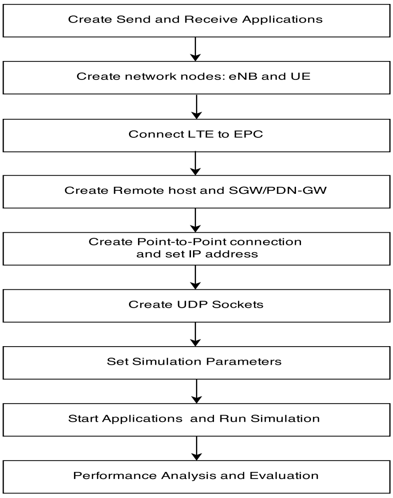

Fig. 4. Overall system flow for creating the LTE-EPC topology

TABLE I. LTE-EPC Performance Simulation Setup Parameters

Parameters	Value
Simulation time (s)	20
Application run-time (s)	20
Downlink Bandwidth (MHz)	5
Resource Block	25
Distance dependent pathloss model	COST231 propagation loss model
Antenna type	2x2 MIMO, SISO
eNB Tx power (dBm)	30, 46 and 60
Fading model	Fast fading
Shadowing deviation	7
UE speed of interest (Kmph)	0
Operating frequency band	2GHz
Downlink EARFCN	100
Traffic model	video
Video packet generation interval (ms)	100
EPS bearer type	NGBR
MAC Scheduler Type	PSS
Encoding scheme	Systematic RLNC
Galois field	Binary8
Generation Size	100
Packet Size (byte)	1024

encoding application IP packets. Galois Field GF(256) is used to generate the coding coefficients to increase the probability of successful decoding. Data is encoded with systematic RLNC and sent as a packet size of 1024 bytes with 100 generation size.

IV. Result and Analysis

Execution of the simulation scenario is done in order to observe the effect of RLNC in LTE networks. The performance is evaluated and analysed by executing two execution rounds,

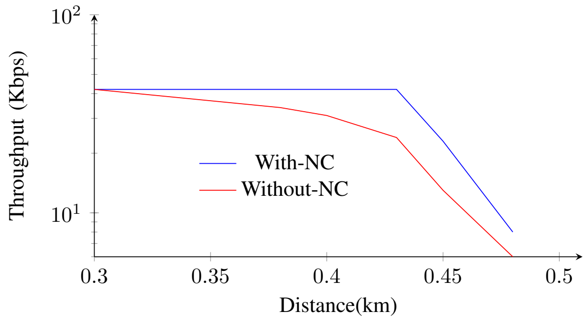

Fig. 5. Throughput performance for 30dBm eNB transmission power: SISO, 5MHz bandwidth.

one for encoded packets and second for normal packets. Accordingly comparison is made between the throughput and the delay performances. The throughput is defined as the rate of successful data delivery while the delay is defines as the end-to-end delay for all the packets sent from remote host to UE until successful decoding of the packets.

A. *Throughput performance analysis*

To analyse the throughput performance with RLNC, 100 packets of size 1024 byte, which is 100x1024 byte data, is encoded and send down to the UE from the remote host. The simulation is performed for three different eNB transmission powers and UE-eNB distance of 0.3km to 4km, system bandwidth of 5MHz. This is done because in order to observe the effect of RLNC we want to consider a lossy link where the UE is located at a reasonable distance to make sure packets are lost, unless if the UE is close enough to the eNB there will not be packet loss since it is only one UE for the given system bandwidth.

Figures 5, 6 and 7 shows the difference in the throughput performance obtained with and without RLNC, for 30, 46 and 60dBm eNB transmission power respectively. As we can observe from these figures the throughput is improved with RLNC. With RLNC, packets are sent as a combination, which means every linearly independent packet received at decoder have some portion of the data transmitted which enables the decoder to recover lost packets. Besides it is performed over HARQ which by itself is an efficient way of recovering lost packets with retransmission requests. The results shown are obtained for the given range of eNB transmission power at corresponding distances. In these figures, the lowest value of the x-axis distance indicates that with the given power level all the packets sent are fully recovered up until that distance. However, as we increase the distance of the UE packets will be lost. For instance, for 30dBm power the throughput is same with and without RLNC up until 0.3km and as packets begin to lost the throughput drops but with better performance with RLNC. .

The reason to use different eNB transmission power levels is to see the effect of NC at different distance and to show

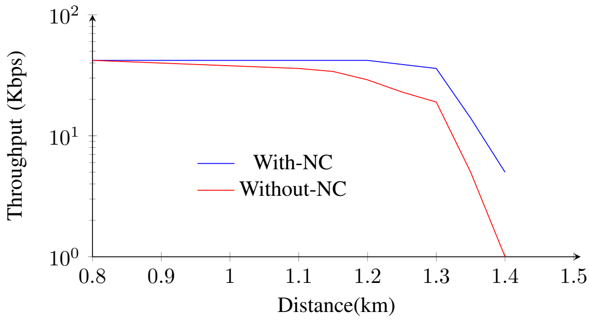

Fig. 6. Throughput performance for 46dBm eNB transmission power: SISO, 5MHz bandwidth.

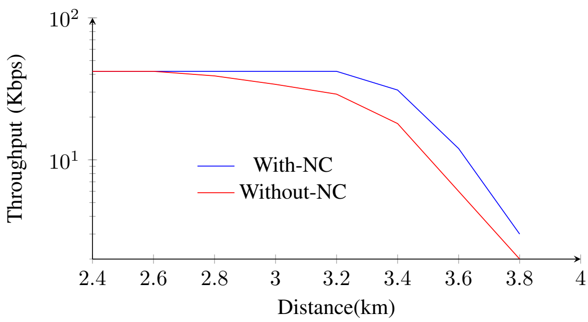

Fig. 7. Throughput performance for 60dBm eNB transmission power : SISO, 5MHz bandwidth.

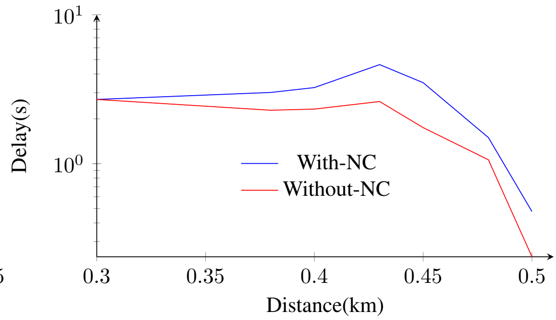

Fig. 8. Delay performance for 30dBm eNB transmission power: SISO, 5MHz bandwidth.

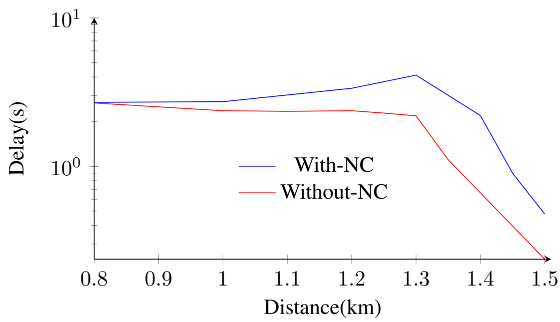

Fig. 9. Delay performance for 46dBm eNB transmission power: SISO, 5MHz bandwidth.

how the performance can be improved not only with NC but also by increasing the transmission power. As we can see from these figures with increasing transmission power we can achieve better throughput performance in LTE networks. For example the throughput and the coverage that can be obtained with 30dBm and RLNC can easily be achieved by increasing the transmission power to 46dB. However this is only for the downlink communication where eNB power transmission is not a big concern, but for UEs increasing the transmission power could have its own effect on the battery life of the UE. Thus, we can say that with out having to deal with the complexity of deploying new RLNC scheme on the existing HARQ scheme the required performance can be achieved with increasing the transmission power.

As shown in these figures the throughput obtained is in Kbps. This is because we only generated 100 packets of size 1024 bytes for 20s simulation and application run time using the application created on remote host and UE.

B. *Delay performance analysis*

Figures 8, 9 and 10, shows the delay performance at 30, 46 and 60 dBm eNB transmission power levels. As shown the delay is higher with RLNC. This is because in RLNC in addition to the transmission delay, the delay includes the time to encode and decode a packet. From these figures we also see how the delay reduces for higher distance between the UE and eNB. This is in line with the decreasing throughput which indicates the total number of received packets are less so is the total delay.

C. *MIMO vs AL-RLNC*

In this part it is shown that MIMO system could be an alternative to deploying RLNC in LTE network. MIMO systems are one of the major enabling technologies for LTE. They allows higher data rate transmission through the use of multiple antennas at the transmitter and receiver to provide simultaneous transmission of multiple parallel data streams over a single radio link. Figure 11 shows the throughput obtained for 60dBm eNB transmission power for MIMO and Single-input Single-output (SISO) systems with and without NC. As shown in the figure the throughput obtained with MIMO system without using RLNC is higher at any given distance as compared with the SISO with NC. From the result we can also see that the MIMO system increases the coverage

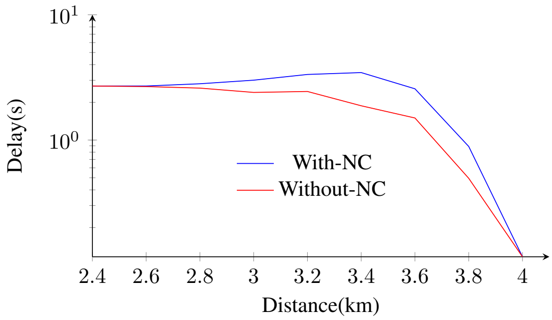

Fig. 10. Delay performance for 60dBm eNB transmission power: SISO, 5MHz bandwidth.

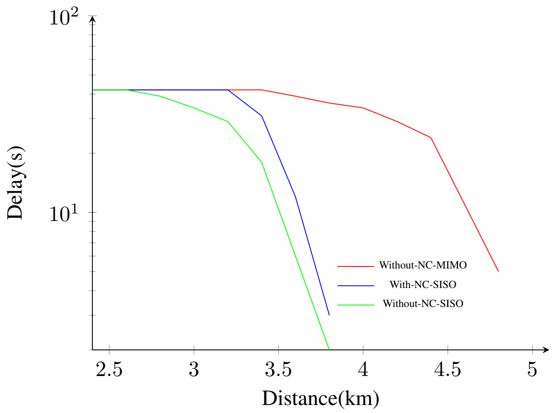

Fig. 11. MIMO, SISO and RLNC performance comparison for 60dBm eNB transmission power.

of the network up until 5km. Thus, we can expect a better throughput performance than the graph shows if we combine MIMO with NC.

V. Conclusion and Future work

This paper presented an AL-RLNC integration in LTE networks deployed on top of HARQ with a goal of achieving an efficient and flexible multimedia delivery mechanism. As the obtained results indicated the throughput performance of LTE network is improved by RLNC but at a cost of higher packet delay. We have also shown that the performance improvement with RLNC can also be obtained either with increasing the eNB transmission power or using MIMO systems. The obtained results are obtained for a simple topology and the performance can further be analysed and evaluated with multiple UEs and remote hosts in the future.

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