End-to-End Delay Performance Evaluation for VoIP in the LTE network

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This thesis is presented as part of Degree of
Master of Science in Electrical Engineering

Blekinge Institute of Technology
June 2011

Blekinge Institute of Technology
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Dedicated to our parents

ABSTRACT

Long Term Evolution (LTE) is the last step towards the 4th generation of cellular networks. This revolution is necessitated by the unceasing increase in demand for high speed connection on LTE networks. This thesis mainly focuses on performance evaluation of end-to end delay (E2E) for VoIP in the LTE networks. In the course of E2E performance evaluation, simulation approach is realized using simulation tool OPNET 16.0. Three scenarios have been created. The first one is the baseline network while among other two, one consists of VoIP traffic solely and the other consisted of FTP along with VoIP. E2E delay has been measured for both scenarios in various cases under the varying mobility speed of the node. Furthermore, packet loss for two network scenarios has been studied and presented in the same cases as for E2E delay measurement. Comparative performance analysis of the two networks has been done by the simulation output graphs. In light of the result analysis, the performance quality of a VoIP network (with and without the presence of additional network traffic) in LTE has been determined and discussed. The default parameters in OPNET 16.0 for LTE have been used during simulation.

Keywords: LTE, VoIP, E2E delay, Throughput and OPNET.

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Acknowledgement

In the name of Allah, the most Merciful & Beneficent

First of all, we would like to thanks to Almighty ALLAH for blessing us with the ability and patience to finish this thesis work.

We would like to thank our advisor and examiner, Dr. Jörgen Nordberg for his excellent support and guidance during this thesis work. Without his suggestions this work would not have been possible.

We want to extend our gratitude to our beloved parents & family for their heartless love, support and encouragement during our thesis work, in particular our mother for her kind love and encouragement advice.

Last but not least, we are grateful to our friends for their supports, discussions, comments and entertaining fun during our thesis work that helped us to feel relax.

Masum & Jewel

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List of Acronyms

3GPP	Third Generation Partnership Project
AMBR	Aggregate MBR
ARP	Allocation and Retention Priority
CDMA	Code Division Multiple Access
CS-ACELP	Algebraic-Code-Excited-Linear-Prediction
DES	Discrete Event System
DFT	Discrete Fourier Transform
DwPTS	Downlink Pilot Timeslot
E2E Delay	End-to-End Delay
eNB/eNodeB	Evolved Node-B
EPC	Evolved Packet Core
EPS	Evolved Packet System
E-UTRAN	Evolved Universal Terrestrial Radio Access Network
FDD	Frequency Division Duplex
FDMA	Frequency Division Multiple Access
GBR	Guaranteed Bit Rate
GERAN	GSM EDGE Radio Access Network
GP	Guard Period
HSPA	High Speed Packet Access
IDFT	Inverse Discrete Fourier Transform
ITU	International Telecommunication Union
LTE	Long Term Evolution
MAC	Medium Access Control
MBR	Maximum Bit Rate
MIMO	Multiple Input Multiple Output
MME	Mobility Management Entity
non-GBR	non-Guaranteed Bit Rate
OFDM	Orthogonal Frequency Division Multiplexing
OFDMA	Orthogonal Frequency Division Multiple Access
OPNET	Optimized Network Engineering Tool
PAPR	Peak-to-Average Power Ratio
PCM	Pulse Code Modulation
PCRF	Policy and Charging Rules Function
PDCP	Packet Data Control Protocol
PDN-GW	Packet Data Network Gateway

PSTN	Public Switched Telephone Network
QCI	QoS Class Identifier
QoS	Quality of Service
RAN	Radio Access Network
RB	Resource Block
RLC	Radio Link Control
ROHC	Robust Header Compression
RRC	Radio Resource Control
RTP	Real-time Transport Protocol
SC-FDMA	Single Carrier-FDMA
SDFs	Service Data Flows
S-GW	Serving-Gateway
SID	Silence Description
SM	Spatial Multiplexing
TCP	Transmission Control Protocol
TDD	Time Division Duplex
TDMA	Time Division Multiple Access
UDP	User Datagram Protocol
UE	User Terminal
UMTS	Universal Mobile Telecommunication System
UpPTS	Uplink Pilot Timeslot
UTRA	Universal Terrestrial Radio Access
UTRAN	Universal Terrestrial Radio Access Network
VoIP	Voice over Internet Protocol
WiMAX	Worldwide Interoperability for Microwave Access

CHAPTER 1

INTRODUCTION

1.1 Introduction

The trend of the modern society is as the days go by, time is getting more expensive and commodity is getting cheaper. To create a world compatible for this, it is necessary to create a network backbone for the whole world so the information along with communication, is instantaneous. As internet is the main information database, cellular technology is required to merge with the core internet structure, with all its bandwidth and fast trafficking facility in the cheapest way possible. This has been the fundamental premise behind the development of LTE. The study of the performance of Voice-over-IP (VoIP) over LTE thus has a great significance. Nowadays, communication and network technology have expanded significantly. As LTE is relatively a new technology, there are not enough technical documents to get a deeper knowledge of LTE for real time application. Introduction of Long Term Evolution (LTE), the 4^{th} Generation (4G) network technology release 8 specifications are being finalized in 3GPP have developed and planning to globalize extensively compared to 3^{rd}Generation (3G) and 2^{nd} Generation (2G) networks [1]. LTE determines goals peak data rate for Downlink (DL) 100 Mbps and Uplink (UL) data rate for 50Mbps, increased cell edge user throughput, improved spectral efficiency and scalable bandwidth 1.4 MHz to 20 MHz [2]. VoIP capacity of LTE has to show better performance as Circuit Switch voice of UMTS. LTE should be at least as good as the High Speed Packet Access (HSPA) evolution track also in voice traffic. The core network of LTE is purely packet switched and optimized for packet data transfer, thus speech is also transmitted purely with VoIP protocols. Simultaneously, demand for the higher quality of wireless communications has increased as well. Use of demand driven applications and services have been growing rapidly to satisfy users. Meeting such demand poses a challenge for the researchers to solve till now. Among such demands, enhance quality of voice and data transfer rates are one of the main aspects to improve. Thus, to improve the performance of such important aspects, performance evaluation of VoIP can point out the issues which can be resolved to improve the overall performance of LTE networks. In this paper, VoIP application is used to represent the class of inelastic, real-time interactive applications that is sensitive to end-to-end delay but may tolerate packet loss. This need is much more expedient in real-time application such as voice has enormous importance in providing efficient services in order to fulfill the users expectation, and hence to the researchers to improve the technology to meet the ever growing demand of efficient use of the system. It is expected that LTE should support a significantly higher number of VoIP users. The important factor is now the quality of service (QoS) of VoIP. To measure QoS of VoIP in a LTE network, the first basic evaluation can be done in terms of maximum end to end delay and acceptable packet loss [3].

1.2 Aims and Objectives

The main objective of this thesis work is to evaluate the End-to-End Delay performance in terms of application such as VoIP and FTP server in the LTE networks. OPNET Modeler 16.0 is used for doing the simulation. In order to achieve the goal the followings have been done:

- Applying both qualitative and quantitative research methods that will guide the study in suitable direction.
- Doing literature study about LTE and real time application.
- Setting up a platform for performing the simulation in OPNET and becoming familiar with different tools of OPNET software.
- Creating different scenarios and analyzing the way of running the simulation in OPNET platform.
- Studying the individuality of voice and FTP server over LTE networks. To understanding how way we can do the configuration in the LTE environment and set their networks attributes into the OPNET Modeler 16.0.
- To select the quantitative metrics such as end-to-delay and throughput.
- Discussing the different constraints that affect the E2E delay performance of VoIP in LTE network and critically examine various approaches that are suggested in the literature for improving the E2E delay performance.
- Developing, testing and evaluating strategic scenario in OPNET.
- Verifying the way of how to minimize the effects of network congestion using FTP server in LTE platform.
- Construing the simulation result and predicting which technology is the best our network modeling objectives.
- Simulating different network scenarios with different network load and analyzing the simulation results.
- Drawing conclusions by presenting and interpreting the outcomes.

1.3 Scope of the Thesis

This thesis covers the technical issues and factors that need to be considered for the implementation of VoIP in the computer networks. It discusses the challenging issues that need to be faced by computer networks to transmit the VoIP applications. It gives the description idea about the VoIP over LTE and their functionality and design parameters of the LTE networks. In this thesis, qualitative and quantitative analysis of E2E delay performance over LTE networks have been done in a simple and understandable fashion so that it might be helpful for those who have some intention to do further research.

1.4 Research Questions

After determining the problems it is necessary to indentify the research questions that lead the research process to be in the scope. The formulated questions are described as follows.

> **Q1**. How much the maximum throughput is support in the different bandwidth (e.g. 1.4 MHz, 3 MHz, 5 MHz, 10 MHz, 15 MHz and 20 MHz)?
>
> **Q2**. What is the impact on the VoIP quality in terms of E2E delay when the network is congested with VoIP only or VoIP with FTP?
>
> **Q3**. To what extent do the performances of packet loss for interactive voice vary from Congested VoIP to VoIP congested with FTP network?

1.5 Research Methodology

The research methodology presented in this thesis is based on both Qualitative and Quantitative approach suggested by John W. Creswell [4]. In this Qualitative approach, three steps are considered:

1. Identify the key factors influencing VoIP performance in LTE networks by considering the existing research and knowledge based on famous scholars, relevant articles and journals i.e. IEEE Xplore, Inspec, Google and Google Scholar.

2. Determine the suitable VoIP model to support real-time application.

3. Justify the VoIP performance thresholds on the basis of strong facts and figures.

In this Quantitative approach, following four steps are considered:

1. Develop a network model based on qualitative approach and experimental research. Experimental research typically starts with the formulation of hypothesis. Design and analysis of the network model need to validate or invalidate the hypothesis [5]. With respect to the present study, the LTE network models are designed in the OPNET simulator based on different network entities. The OPNET simulator is well-known for network design and attractive features. In the OPNET simulator, different network entities are needed to accurately configure support selective

application services of the network.

2. Evaluate the performance of different simulation scenarios in terms of VoIP when LTE is deployed.

3. Collect quantitative data regarding throughput, end-to-end delay and packet loss for analysis of the network performance.

4. The simulation results are collected using OPNET in terms of different statistical graphs and tables as furnished in chapter 5.

1.6 Motivation

The 3GPP LTE is a new standard with comprehensive performance targets, therefore it is necessary to evaluate the performance and stability of this new system at an early stage to promote its smooth and cost-efficient introduction and deployment. The motivation behind the design models presented in this report is to discuss issues related to traffic behavior for VoIP alone as well as along with other traffic in the LTE network. E2E delay for VoIP is a matter of fact for performing real-time application efficiently over the Internet. Today, emergence of the real-time application demands more resources. The main motivation of our thesis work is to ensure fast and reliable voice communication for huge number of users in wireless network. In our framework, the evaluate and analyze the E2E delay performance of voice based on the performance metrics such as throughput, E2E delay and packet loss over LTE network.

1.7 Contribution

This thesis is focused on the comprehensively analyzed for VoIP performance metrics such as end-to-end delay and throughput for real-time applications over the LTE networks. In our dissertation, a number of important system parameters such as network load and fixed node speed are taken into consideration. OPNET Modeler 16.0 is used to design the model for simulation (Baseline VoIP network scenario, congested VoIP network scenario and VoIP congested with FTP network scenario) to realize different realistic LTE scenarios as well as to determine the extent of their impact on network and VoIP performance.

1.8 Thesis Outline

The outline of this thesis paper is organized as following structure:

Chapter 1 provides the introduction, aims and objectives, research methodology, motivation and contribution of this research are discussed, and also discusses about the research question and scope of the thesis.

Chapter 2 covers the general overview of the 3GPP LTE technology standard, multiple access technique, frame structure and network architecture. Furthermore, Multiple-antenna technique is described in briefly at the end of this chapter.

Chapter 3 presents the Quality of Service (QoS) of LTE where guaranteed bit rate (GBR), non-guaranteed bit rate (non-GBR) and characteristics of QCI are described in briefly. Moreover, Voice over IP (VoIP) principle, codec and characteristic are focused in this chapter.

Chapter 4 dedicates a discussion the experiment setup, network scenarios and the parameters required to configure them.

Chapter 5 explains the simulation result followed by chapter 4.

Chapter 6 concludes the entire thesis work.

CHAPTER 2

Theoretical Knowledge

2.1 Background

Lately, the demand for high data rates to support the Internet services and the wide range of multimedia has received a substantial attraction around the globe from mobile researchers and industries. An international collaboration project, known as, Third Generation Partnership Project (3GPP), takes a host of members into account, specially, from both mobile industries and research institutes in a bid to delivering a globally applicable third generation (3G) mobile phone system specification [6]. The organization started their journey on December 1998 and was initially based on 2nd generation (2G) mobile system, i.e. Global System for Mobile Communications, which is nowadays known as Universal Mobile Telecommunications System (UMTS). The key function of 3GPP involves improving the UMTS standard to cope with the ever-evolving future requirements such as services boosting, exploiting the spectrum facilities, lowering costs, efficiency improvement and better integration with other standards. The following table (Table 2.1) demonstrates few complete-sets of technical specifications produced by 3GPP.

Table 2.1: Technical specifications published by the 3GPP group

Release	Specification	Date	Downlink Data Rate	Uplink Data Rate	Round Trip Time
Release 99	WCDMA	March, 2000	384 kbps	128 kbps	150 ms
Release 4	TD-SCDMA	March, 2001	384 kbps	128 kbps	150 ms
Release 5	HSDPA	March to June, 2002	14 Mbps	5.7 Mbps	<100ms
Release 6	HSUPA	December, 2004 to March, 2005	14 Mbps	5.7 Mbps	<100ms
Release 7	HSPA	December, 2007	28 Mbps	11 Mbps	< 50 ms
Release 8	LTE	December, 2008	100 Mbps	50 Mbps	10 ms
Release 10	LTE-Advanced	Published 2012	1 Gbps in a low mobility	375 Mbps	

2.2 Requirements for Long Term Evolution (LTE)

The radio access of Long term Evolution (LTE) is often known as Evolved UMTS Terrestrial Radio Access Network (E-UTRAN) and is likely to support various types of services such as

video streaming, FTP, web browsing, VoIP, real time video, online gaming, push-to-talk, push-to-view and so on. Consequently, it has been immensely important for the LTE to be designed as a high data rate and low latency system as pointed out by the key performance criteria in Table 2.2. For both transmission and reception, the bandwidth capability of a UE is required to be 20MHz [15]. Though, the service provider can deploy the cells with any of the bandwidths specified in given table. This eventually allows the service providers to alter their offering dependent on the amount of available spectrum or the ability to initiate with fixed spectrum for lower upfront cost and grow the spectrum for additional capacity.

In LTE, the interworking with existing UTRAN/GERAN systems and non-3GPP systems should be ensured. Multimode terminals need to support handover from and to UTRAN and GERAN and also the inter-RAT measurements. In real time services, the interruption time of handover between E-UTRAN and UTRAN/GERAN should be less than 300 ms, and in-case of no real time services, the time should be less than 500 ms. Ability of cost effective migration from release 6 UTRA radio interface and architecture should be available. Cost and power consumption, reasonable system and terminal complexity are to be provided. It is mandatory for all the interfaces to be open for multi-vendor equipment interoperability.

Table 2.2: LTE performance requirements

Metric	**Requirement**
Peak Data Rate	DL: 100Mbps UL: 50Mbps (for 20MHz Spectrum)
Mobility support	Up to 500kmph but optimized for low speeds from 0 to 15 km/h
Control plane latency (Transition time to active state)	< 100ms (for idle to active)
User plane latency	< 5ms
Control plane capacity	> 200 users per cell (for 5MHz spectrum)
Coverage (Cell sizes)	5 – 100km with slight degradation after 30km
Spectrum flexibility	1.4, 3, 5, 10, 15 and20MHz

2.3 Multiple Access Techniques

A Multiple access technique effectively utilizes the expensive transmission resources among multiple users to minimize the communication interferences. The conventional multiple access strategies are based on dividing the available resources through implementing the frequency, time, or code division multiplexing techniques. For instance, in Time Division Multiple Access (TDMA), each user is allocated to a unique time slot, either on demand or in a fixed rotation. But in Frequency Division Multiple Access (FDMA), each user is assigned to a unique carrier frequency and bandwidth. And in Code Division Multiple Access (CDMA), the

users will belong to the unique code for transmission, allowing each user to share the entire bandwidth and the time slots [7].

Meanwhile, two candidate standards of IMT-Advanced (i.e., mobile WiMAX and LTE) make use of OFDMA as the multiple access technique in the downlink direction. However, with different resource grouping, frame structures, and allocation. On the other hand, the two systems implement different techniques in the uplink direction, for instance, the mobile WiMAX uses OFDMA and 3GPP standardization group uses SC-FDMA in LTE [8]. The SCFDMA technique explores a modified version of OFDM scheme (also known as DFT-spread orthogonal frequency division multiple access) to mitigate the high PAPR problem [9]. The SC-FDMA becomes more attractive for uplink transmission due to having its low PAPR property especially when this is a case of low-cost device with limited energy resources.

2.3.1 OFDMA for DL

In general, OFDMA is an OFDM-based multiple access scheme that is utilized in the downlink direction for both LTE and WiMAX standards. The OFDMA can be considered as a combination of the FDMA and TDMA techniques as shown in Figure 2.1. In such a technique, each user is given a unique fraction of the system bandwidth (OFDM subcarriers) in each specific time slot [7, 9]. The most of the OFDMA advantages are inherited from OFDM technique (i.e. better spectral efficiency). In addition, OFDMA is capable of managing the resource scheduling based on the frequency responses and channel time, which eventually allow allocation of different subcarriers to the individual users based on the channel condition. This technique is often known as multiuser diversity [10].

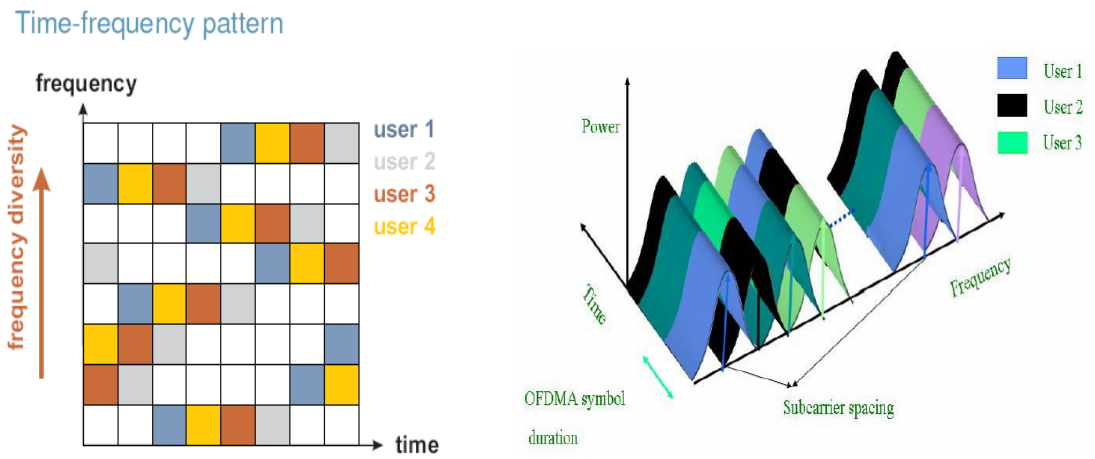


Figure 2.1 OFDMA basic operations [7]

2.3.2 SC-FDMA for UL

In SC-FDMA, prior to performing the OFDMA modulation technique, the time domain data symbols are often transformed to frequency domain by applying DFT method as illustrated in Figure 2.2. Similar to the OFDMA, the orthogonality of the users in such a technique is obtained by assuming the fact that each source/user occupies different subcarriers in frequency domain. The key difference between OFDMA and SC-FDMA includes the introduction of an additional DFT and IDFT module at the transmitter and the receiver side, respectively. In both

cases, equalization technique is implemented in frequency domain though OFDMA performs modulation and demodulation operations in the frequency domain while SC-FDMA performs these operations in the time domain.

SC-FDMA spreads each modulated symbol very efficiently across the total channel bandwidth and thus makes it relatively less sensitive to the channel frequency-selective fading effect than that of OFDMA. However, in OFDMA, the utilization of narrower bandwidth makes it advantageous over SC-FDMA thorough allowing potential adaptation of the modulation techniques and power resource per individual subcarrier [9, 11]. The most important advantage and difference between the OFDMA and the SC-FDMA is the low PAPR of SC-FDMA [11]. Figure 2.2 shows OFDMA and SC-FDMA transceiver comparison [12].

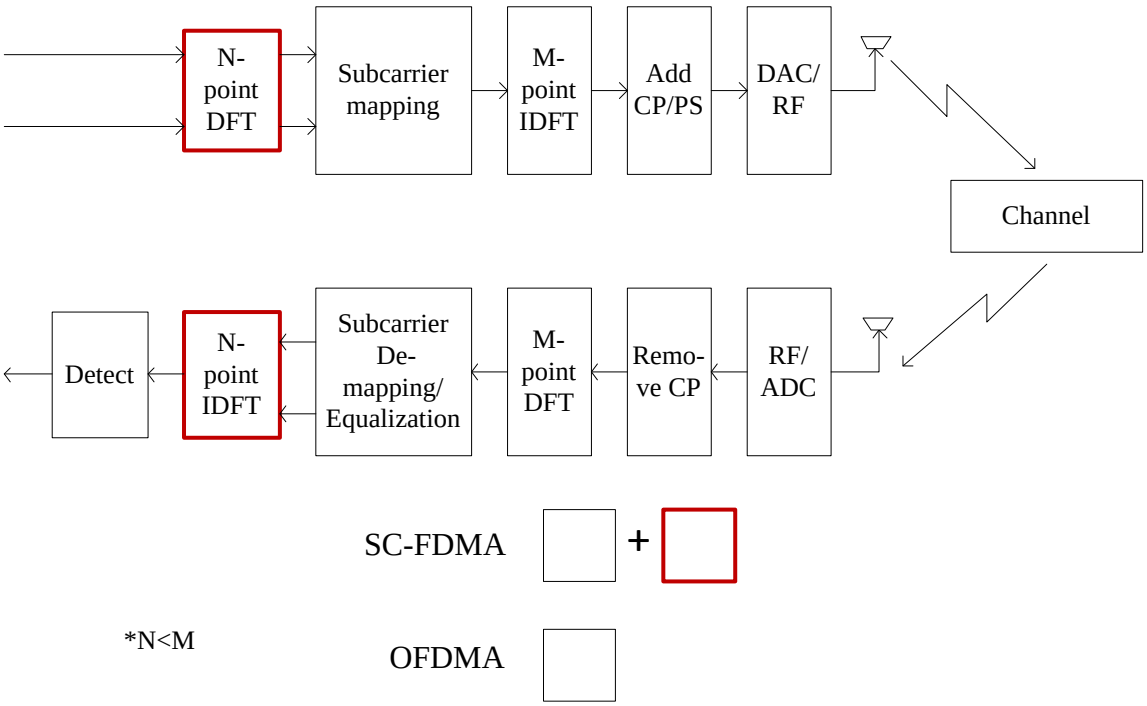


Figure 2.2 OFDMA and SC-FDMA transceiver comparison [10]

2.4 Generic Frame Structure

Two types of generic frame structures are designed for radio access network of LTE [13], as Type 1 and Type 2. Type 1 and 2 frame structures are applicable for Frequency Division Duplex (FDD) and Time Division Duplex (TDD), respectively.

2.4.1 Type 1 LTE Frame Structure

Type 1 frame structure supports on both half duplex and full duplex FDD modes. This kind of radio frame has period of 10ms and each slot equal to 0.5ms; each radio frame consists of 20 slots. A sub-frame belong to two slots, hence one radio frame has 10 sub-frames as depicted in figure 2.3. In FDD mode, there are two carrier frequencies domain, one for uplink direction (f_{UL}) and another for downlink direction (f_{DL}). The frames of uplink and downlink are transmitted simultaneously.

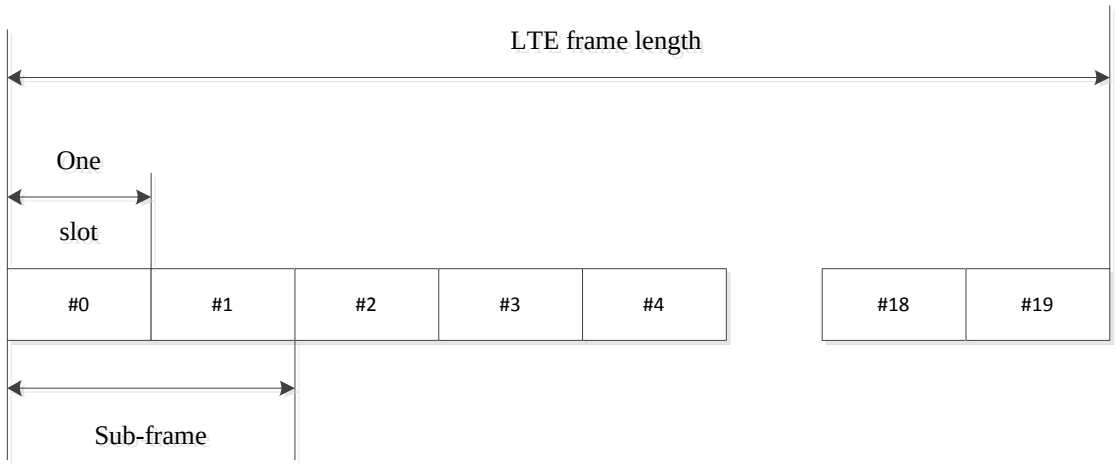


Figure 2.3 Type-1 LTE Frame Structure

2.4.2 Type 2 LTE Frame Structure

Type 2 frame structures are applicable to TDD; the radio frame is composed of two identical half-frames and the duration of each half-frame is 5ms. Both half-frame have further divided into 5 sub-frames which is equal duration of 1 ms as illustrated in figure 2.4. Every sub-frame consists of two slots and each slot has time of 0.5ms. There are three special sub-frames field namely; Guard Period (GP), Downlink Pilot Timeslot (DwPTS) and Uplink Pilot Timeslot (UpPTS). The length of these three fields must be equal to 1ms.

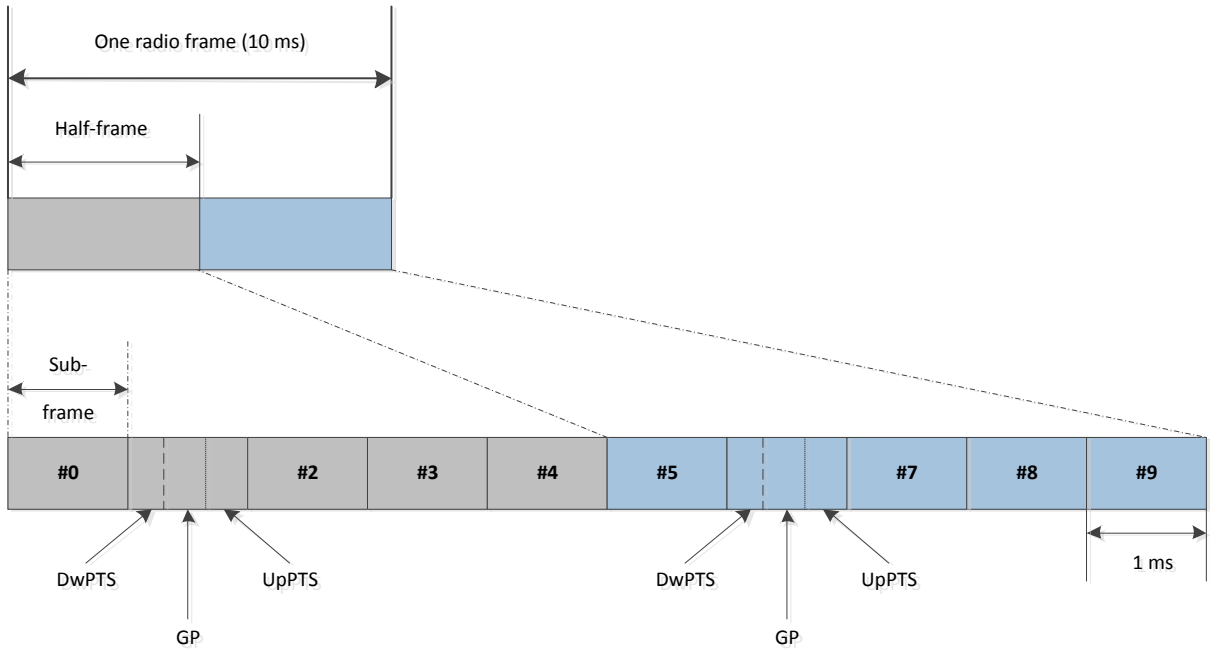


Figure 2.4 Type-2 LTE Frame Structure

2.5 Physical Resource Block Parameters

In LTE, the radio resources are structured into time-frequency grid, which is collected of *Nsc* successive subcarriers in frequency domain and time-slots in time domain as demonstrated in figure 2.5. The smallest radio resource unit is called by the resource element. It corresponds to the task of one subcarrier per one time-slot. These resource elements are collected in form of resource blocks as illustrated in figure 2.5.

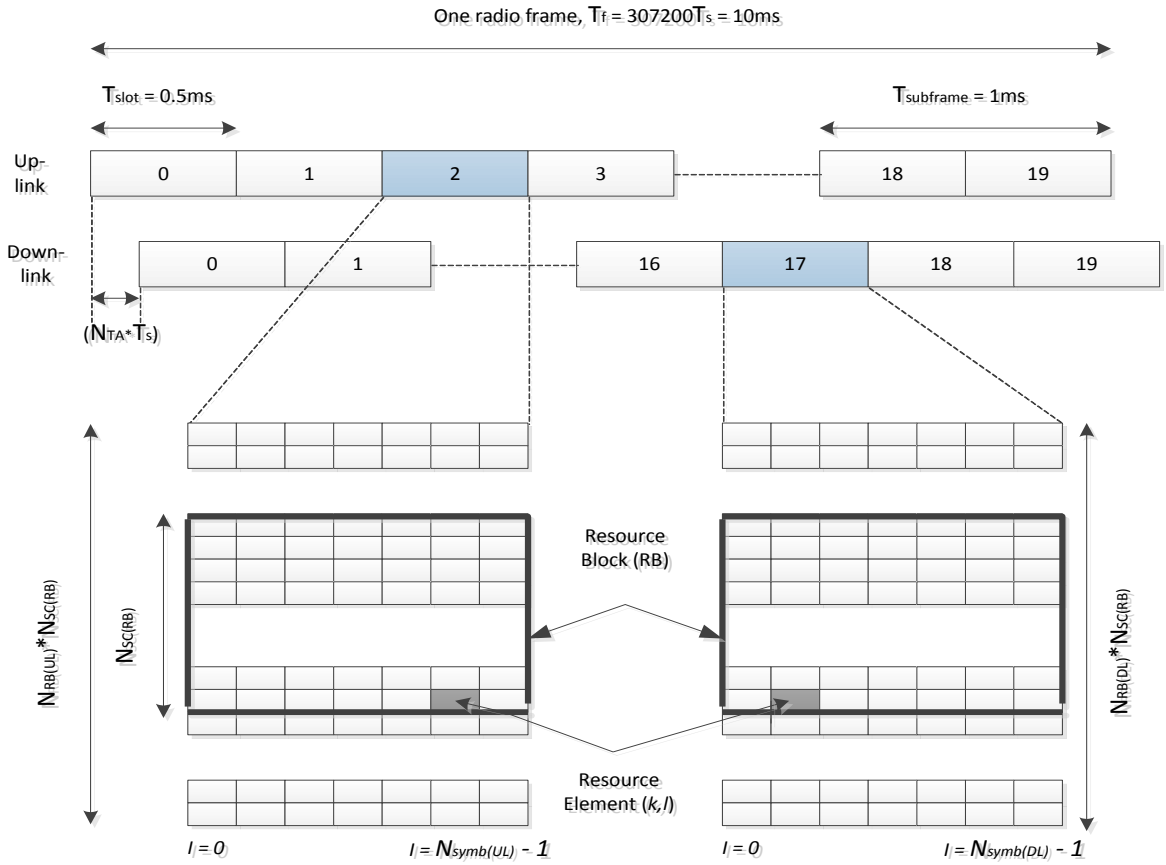


Figure 2.5 Frame structure and physical resource block in LTE uplink & downlink [9]

The Resource Block (RB) shows the smallest radio resources which can be allocated to any user for every time slot. The bandwidth of one RB is 180 kHz (15 kHz or 7.5 kHz frequency spacing used by sub-carriers 12 or 24, respectively) [9, 11].The numbers of RBs in the resource grid differ corresponding to the used from 1.4 MHz to 20 MHz as presented in Table 2.3 [9].

Table 2.3 Resources Block Number per Channel Bandwidth [9]

Channel Bandwidth [MHz]	1.4	3	5	10	15	20
Number of resource Block (Nrb)	6	15	25	50	75	100
Number of occupied subcarrier	72	180	300	600	900	1200
IDFT (Tx)/DFT (Rx) size	128	256	512	1024	1536	2048
Sample Rate [MHz]	1.92	3.84	7.68	15.36	23.04	30.72
Samples per slot	960	1920	3840	7680	11520	15360

2.6 LTE Radio Access Network Architecture

The fundamental architecture of LTE system is presented in Figure 2.6. All the network interfaces are based on internet protocols (IP). The LTE system as depicted in Figure 2.6 comprised of the core network and radio access network which represent the IP connectivity layer of LTE system [8, 11].

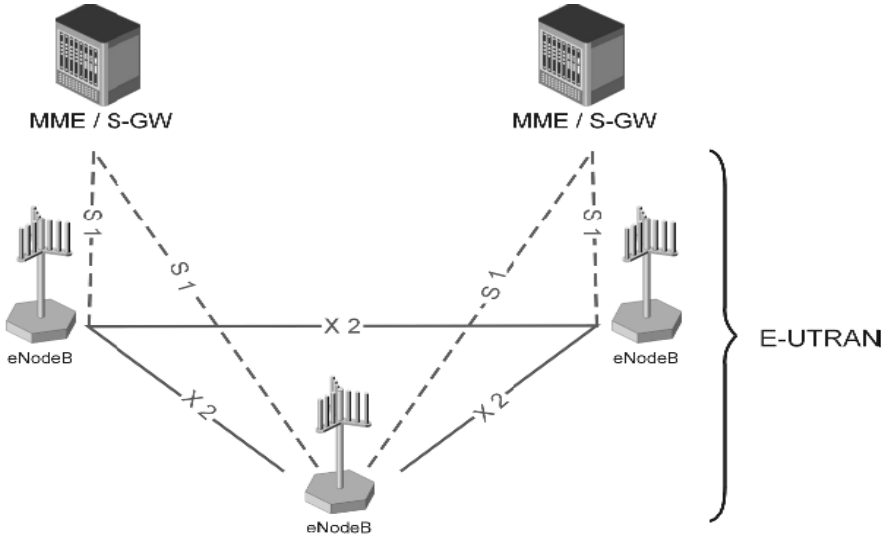


Figure 2.6 Networks Architecture of LTE

2.6.1 Core Network

In LTE, the core network operations are completely based on packet switching domain, i.e., all the network interfaces are dependent on IP protocols, and hence it is known as Evolved Packet Core (EPC) [11]. The essence of EPC is to keep the number of operating nodes and interfaces as minimum as possible. The EPC divides the network components into control-plane objects such as data/barer-plane entity (i.e. a Serving Gateway) and the Mobility Management Entity (MME). The major entities in the EPC are described briefly in the following subsections [11, 14].

2.6.1.1 Mobility Management Entity (MME)

The MME is considered as a signaling entity and used to represent the control plane function of the EPC. Such control functions include, among others, location function, the subscribers' equipments paging, and the bearer establishment, the connections establishment, roaming management, UE location update, controlling the UE authentication and authorization procedures, and security negation [11, 14].

2.6.1.2 The Serving Gateway (S-GW)

S-GW functions as switching as well as routing node to route and forward the data packets to and from the BS or Evolved-Universal Terrestrial Radio Access Network NodeB (eNB). For

instance, the S-GW produces a tunnel during the connection mode (i.e. UE is connected) to transmit data traffic between the P-GW (Packet Data Network Gateway) and UE (via specific BS) [14, 15].

2.6.1. 3 Packet Data Network Gateway (PDN-GW)

Between the EPC and the external packet data network, a PDN-GW is often used as an interface point or an edge router. It is also possible that a UE has synchronized connectivity with more than one PDN GW [14, 15].

The responsibilities of the PDN-GW include establishment, maintenance, and deletion of GTP tunnels to S-GW or SGSN in the case of inter-RAT mobility scenarios. The PDN-GW routes the user plane packets by allocating the user's dynamic IP addresses. Apart from that, it provides functions for lawful interception, policy/QoS control, and charging.

2.6.1.4 The Policy and Charging Rules Function (PCRF)

The PCRF mainly performs the Policy and Charging Control (PCC) functions. It is used to control the QoS configuration and tariff making of each individual user. The specified tariff and QoS policies for each UE are given to the P-GW and the S-GW [14, 15].

2.6.2Radio Access Network

The radio access network of LTE is termed as Evolved Universal Terrestrial Radio Access Network (E-UTRAN). The evolved RAN for LTE comprises of a single node, i.e., BS or eNB, which often involves with the UEs. The BS or e-NB involves controlling all the radio interface related functions. Between UE and EPC, the eNB acts as a gateway, and manipulates all the communications towards the UE and forwards radio connection to core network (EPC) by using the related radio protocols and the corresponding IP based connectivity, respectively. To gain its function as interface between the core and the radio parts of the network, the eNB hosts two bunches of protocols, namely, the control plane protocols and the Evolved Universal Terrestrial Radio Access (E-UTRA) user plane protocols. The first bunch, i.e., the user plane contains Radio Link Control (RLC), the Physical (PHY), Medium Access Control (MAC), and Packet Data Control Protocol (PDCP) layers protocols where it is necessary to relay the data traffic to and from the UE. The other protocol i.e. the control plane is associated with the Radio Resource Control (RRC) and it manipulates functions such as the radio resource management, admission control and resource scheduling [15, 16].

2.7 Multi-Antenna Technique

In LTE system, multi-antenna transmission techniques can be realized to establish better system performance (i.e. increasing the capacity and providing higher data rate per user) [17]. Three schemes regarding this technique are described as follows [8]:

- Spatial Diversity: the reason of implementing this technique is to achieve transmission or reception diversity by minimizing the instantaneous fading effects caused due to the multipath propagation. The spatial diversity technique creates a host of independent

paths. To receive higher gain at the receiver side, it transmits and receives with low fading correlation.

- Beam-forming: The purpose of this technique is to allow the base station to conduct a direct transmission, or to allow the radiation beam to move toward the specific user in for boosting the received signal power.
- Spatial multiplexing (SM) or multiple-input and multiple-output (MIMO): Through employing this technique, a high data transmission rate is achieved by transmitting various data streams over independent parallel channels. This is done by utilizing multiple transmitting and receiving antennas, without increasing the channel bandwidth or the total transmitted power.

LTE realizes different multi-antenna techniques such as single user (SU)-MIMO, multiuser (MU)-MIMO, transmit diversity and dedicated beam-forming [8, 18].

CHAPTER 3

LTE QoS and Voice over IP

3.1 LTE QoS Framework

LTE evolved packet system (EPS) is the bearer of the QoS level of granularity. This system also establishes the packet flow between the user terminal (UE or MS) and the packet data network gateway (PDN-GW). The traffic running between a particular client application and the service can be wrecked into split service data flows (SDFs). Mapping the same bearer, SDFs receive common QoS activities (e.g., scheduling policy, queue management policy, rate shaping policy, and radio link control (RLC) configuration) [19, 20]. A scalar value referred to as a QoS class identifier (QCI) with the help of bearer, specifies the class to which the bearer belongs. Set of packet forwarding treatments referred by QCI (e.g., weights scheduling, admission thresholds, configuration of link layer protocol and queue management thresholds) preconfigured through the operator on behalf of each network element [21]. The class-based technique applies in the LTE system to improve the scalability of the QoS framework. LTE bearers are illustrated in Figure 3.1.

In the LTE framework, bearer management and control follows the network-initiated QoS control paradigm that initiated network establishment, modification, and deletion of the bearers.

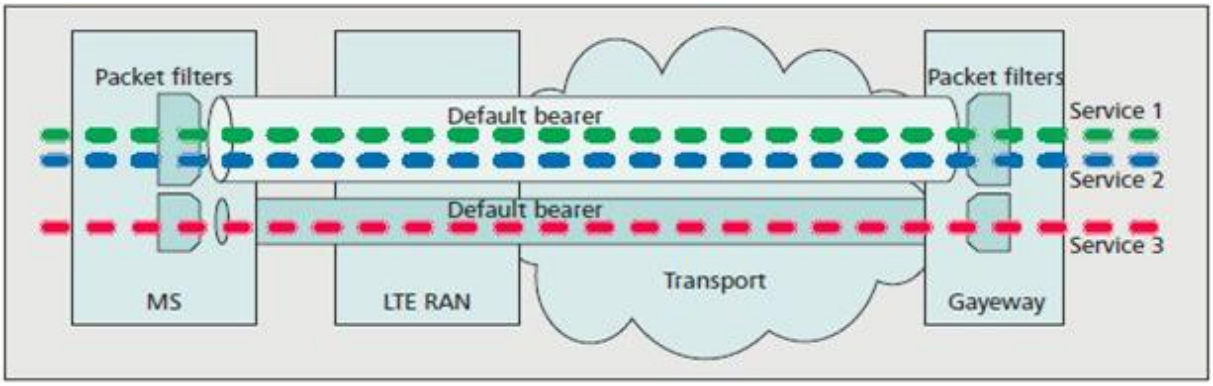


Figure 3.1 Default and dedicated bearers of a terminal (MS) in the LTE QoS framework [20]

Two types of bearers in LTE:

- **Guaranteed bit rate (GBR)**: Dedicated network resources correlated to a GBR value connected with the bearer and permanently allocated when a bearer becomes established or modified [20].
- **Non-guaranteed bit rate (non-GBR)**: In the LTE system, non-GBR bearer is assigned as the default bearer, similar to the preliminary SF in WiMAX, used to estab-

lish the IP connectivity. A non-GBR bearer has enough knowledge about congestion-related packet loss. In the framework, additional bearer is assigned as a dedicated bearer which is GBR or non-GBR.

Dedicated bearer is classified by IP five-tuple based packet filter moreover provisioned in PCRF or defined by the application layer signaling in the mapping of SDFs in LTE. In the mapping, SDF is not equivalent to the existing dedicated bearer packet filters. As a result, traffic is rerouted to the default bearers, if the dedicated bearer packet is dropped [21].

LTE ensures the multivendor deployments and roaming because of a number of standardized QCI values with homogeneous characteristics which reorganizes the network elements. Table 2 shows the mapping of standardized QCI values to standardized characteristics [22].

Table 3.1 LTE standardized QCI characteristics [22]

QCI	Resource type	Priority	Packet delay budget	Packet error loss rate	Example services
1	GBR	2	100 ms	10^{-2}	Conversational voice
2		4	150 ms	10^{-3}	Conversational video (live streaming)
3		3	50 ms	10^{-3}	Real time gaming
4		5	300 ms	10^{-6}	Non-Conversational video (buffered streaming)
5	Non-GBR	1	100 ms	10^{-6}	IMS signaling
6		7	100 ms	10^{-3}	Voice, Video (live streaming), Interactive gaming
7		6	300 ms	10^{-6}	Video (buffered streaming), TCP-based (e.g., www, e-mail, chat, ftp, p2p file, sharing, progressive video, etc.)
8		8			
9		9			

Following QoS attributes associated with the LTE bearer:

- **QCI**: A set of packet forwarding treatments represented by the scalar (e.g., scheduling weights, admission thresholds, queue management thresholds, and link layer protocol configuration)
- **Allocation and retention priority (ARP)**: Call admission control and overload control plane treatment of a bearer uses a restriction. To decide then whether a bearer establishment or modification, call admission control uses the ARP, is to be accepted or rejected. Similarly, the overload control uses the ARP to decide which bearer to release during overload situations [21]
- **Maximum bit rate (MBR)**: Bearer may not exceed the maximum sustained traffic rate; it is only valid for GBR bearers
- **GBR**: The minimum reserved traffic rate the network guarantees; it is only valid for GBR bearers
- **Aggregate MBR (AMBR)**: A group of non-GBR bearers is the total amount of bit rate. It distinguishes between its subscribers by transmitting higher values of AMBR

to its higher-priority customers compared to lower-priority ones by the help of AMBR. 3GPP releases 8 number of MBR which is equal to the GBR and another 3GPP releases an MBR that are greater than a GBR

3.2 Real-time Transport Protocol

IEFT developed many standardized network protocols; Real-time Transport Protocol (RTP) is one of them for audio and video transmission [23]. It was originally designed for multicast protocol published in 1996, although, this protocol is now widely used in unicast applications. RTP can independently carry any type of real-time data without help of underlying protocol. The most popular protocol is the Transmission Control Protocol (TCP) or the User Datagram Protocol (UDP). RTP applied above them is intended for real-time applications and such applications normally are more sensitive to delay than packet-loss. RTP usually chooses the UDP as an underlying protocol.

RTP is the basic protocol in Voice over Internet Protocol (VoIP) engineering, which is, not only for transporting media streams but also to initialize the media session in concord with SIP. It is also used for media stream supervision and intended to provide out-of-band control information for the RTP flow. In response to the media quality that supplies to the other members in the media session via separate UDP port, there are many additional functionalities of RTP. Audio and video synchronization and quality improvements through low compression instead of high compression are a few of them.

3.3 VoIP Principle

VoIP is a technology that delivers voice communications over computer networks like the Internet or any other IP-based network. Using the Internet's packet-switching capabilities, VoIP technology has been implemented to provide telephone services and offers substantial cost savings over traditional long distance telephone calls. VoIP transmissions are deployed through traditional routing [24]. A typical VoIP structural design is showed on the Figure 3.2, though many "possible" modifications of this architecture are implemented in existing systems.

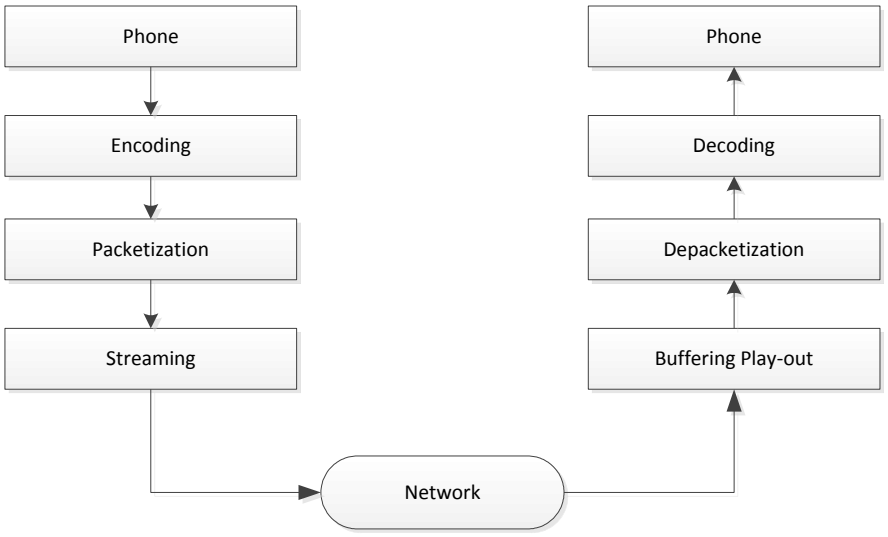


Figure 3.2 VoIP architecture

In VoIP engineering, original voice signal is sampled and is encoded to a constant bit rate digital stream at the end of the sending process. This compressed digital stream data is then encapsulated into equal sized packets to broadcast it easily over the Internet.

Every packet contains the compress voice data along with the information of the packet's origin, projected destination address and have the packet stream to be reconstructed in the correct order with the help of timestamp. In place of circuit-switched voice transmission and traditional dedicated lines, these packets flow over a general-purpose packet-switched digital to analog signal in the receiving end for it to be easily detected by human ear.

Generally, voice data information is sent in digital form in discrete packets rather than using the traditional circuit-committed protocols of the Public Switched Telephone Network (PSTN). In addition, VoIP technology ensures the precise time packet delivery with the help of RTP. In the last few years, VoIP took the place of existing telephone networks and is progressively gaining more popularity for voice quality and the cost. It has the potential to completely substitute for the world's current phone systems.

3.4 VoIP Codec

Human voices are analog. In modern technology, transmit the digital signal for better communication. For that case, a codec (coder/decoder) is used during the voice communication. In the transmitting end, a codec converts the analog signal to a compress digital bitstream, and at the receiving end, another codec converts the digital bitstream back into analog signal. For RTP packet, codec used the payload type or the encoding method in the VoIP technology. Generally, codec provides a compression capability to save network bandwidth and also supports silence containment, where silence is not encoded or transmitted. Compression capabilities of the codecs save the network bandwidth and support the silence suppression. Size of the resulting encoded data stream, speed of the encoding/decoding operations and the quality and fidelity of sound and/or video signal are the three most important factors to be optimized by codecs. Basic characteristic of ITU standard codecs are illustrated in Table 3.2.

Table 3.2 VoIP Codec Comparison

Codec	Algorithm	Data Rate (Kbps)	Packetization Delay (ms)
G.711	PCM	64.0	1.0
G.723	Multi-rate Coder	5.3 and 6.3	67.5
G.729	CS-ACELP	8.0	25.0

G.711 codec technologies apply the Pulse-Code Modulation (PCM) samples method for signals of voice frequencies sampled at 8000 samples/second i.e 8 binary digits per sample. G.711 encoder will create a 64Kbps bitstream.

G.723 speech coder is developed for multimedia ground, and is specified by the H.32x series recommendations. Two types of compressed bit rates are provides by G.723 codec. 6.3 Kbps bit rate is the greater quality to optimize, to represent high quality speech and has the limited amount of complexity with the low bandwidth requirement. 5.3 Kbps is the smallest bit rate [25].

G.729 codec technologies apply the Conjugate-Structure Algebraic-Code-Excited-Linear-Prediction (CS-ACELP) speech compression algorithm, approved by ITU-T. It is an 8Kpbs bit rate and offers tax quality speech at low bit rate and also allows reasonable transmission delays. It will be perfect for teleconferencing or visual telephony where quality, delay and bandwidth are important [26].

3.5 Characteristics of VoIP

The major characteristics of VoIP traffic is authoritarian delay requirements. AMR codec provides the VoIP traffic along with the Voice Activity Detector, Relieve Noise Generation and Discontinuous Transmission. Depending on the speed activity of the traffic, AMR provides a constant rate of small packets transmission. During the active period, one VoIP packet took at 20 ms intervals and 160 ms interval for one Silence Description (SID) packet during silent period. To improve the spectral efficiency of the VoIP traffic, UDP, IP and RTP headers in LTE are also compressed with Robust Header Compression (ROHC). According to [27], for voice signal, 250 ms is the maximum tolerable mouth-to-ear delay and around 100 ms delay for the Core Network and also less than 150 ms acceptable delay for Medium Access Control (MAC) buffering and Radio Link Control (RLC). Both end users are LTE users and assume less than 80 ms acceptable delay for buffering and scheduling. For 3 GPP performance evaluations 50 ms delay has been bound for variability in network end-to-end delays.

The outage limit of maximal VoIP capacity for LTE is limited in TR 25.814 [28] and R1-070674 is updated in contribution. Based on the above limitation, VoIP capacity can be defined as follows:

- The system capacity is defined as the number of users in the cell when more than 95 % of the users are satisfied.
- A VoIP user is satisfied if more than 98 % of its speech packets are delivered successfully.
- It is required for VoIP user that the packet End-to-end delay shouldn't exceed 150 milliseconds [29].

3.6 End-to-End Delay

End-to-end delay means the time required for a packet to be traversed from source to destination in the network and is measured in seconds. Generally, in VoIP network there are three types of delays occurring during the packet transverse. They are: sender delays when packets are transverse from source node, network delay and receiver delay.

In one direction from sender to receiver for VoIP stream flow, end-to-end delay can be calculated by the equation [30]:

$$D = D_{pack} + \sum_{h \in Path}(T_h + Q_h + P_h) + D_{play} \quad (3.1)$$

where, D is the end-to-end delay and D_{pack} is the delay due to packetization at the source. During the packet encoding in the source site, there is also D_{enc} and $D_{process}$. Encoding delay D_{enc} occurs while conversion of A/D signal into samples. PC of IP phone processing is

defined by $D_{process}$ including encapsulation. D_{pack} and D_{enc} , are 20 ms and 1ms respectively in G.711 technology.

By using the equation 3.1, in the worst case scenario, an approximate delay of 25 ms is being introduced at the source. At the end of the transmission, D_{play} is the playback delay together with jitter buffer delay where jitter delay is at most 40 ms because of two packets.

Similarly, at the receiving site, total fixed delay is 45 ms including $D_{process}$. Due to transmission, propagation and queuing in the packet network through each hop h, the path from the sender to the receiver, the total delays are $T_h + Q_h + P_h . where$, T_h is the transmission delay, Q_h is the queuing delay and P_h is the propagation delay. We apply the queuing theory to calculate the transmission and the queuing delay are expressed as $\sum_{h \in Path}(T_h + Q_h)$ and propagation delay will be added for WAN, which is typically ignored for LAN.

CHAPTER 4

Simulation Design and Implementation

4.1 Evaluation Platform

In real world scenarios, performance evaluation of a well designed network model and the model itself carries significant importance. Though, the performance evaluation process is a complex and challenging task in a real scenario. In-order to cope with the challenge, different simulators is being used in practice to simulate the network model from different perspectives. For example, well known open source simulator such as NS-2, gives simulators the flexibility to extend the simulation environment. Nevertheless, modeling in real world scenarios are too complex to model in NS-2.

On the other hand, OPNET (Optimized Network Engineering Tool), introduced by OPNET Technologies [31], is a commercial simulator where the kernel source code is not open. However it has a rich and comprehensive development features built in, which eases the process of designing the real world scenario and simulating the network models [32]. It adds comprehensive options as being both an object oriented and Discrete Event System (DES) based network simulator. In our studies, we used OPNET modeler 16.0 for its reliable and efficiency for simulation. The motivation of choosing OPNET will discuss in below section.

4.1.1 Why OPNET?

Through DES, OPNET models the system behavior by each event in the system effectively. It's efficiency can be measured from the below mentioned features:

- Provides more features than any other simulator in practice.
- Allows modelers to directly include models in it with a wide range of available standard and vendor specific communication networks. It also helps to reduce the development time greatly.
- Has a dynamic development environment with rich features that support both distributed systems and modeling of communication networks.
- Has a large and user friendly documentation to guide users.
- Provides easy graphical interface to work and view the results.
- OPNET results are flexibly interpretable (i.e. exported to spreadsheets), and have comprehensive tools to support display, plot and analyze time series, histograms, probability, parametric curves, and confidence intervals.

4.2 Network Model Configuration

4.2.1 Network Components

This section briefly describes about the following network elements used in our study network models running on OPNET [33].

- The lte_access_gw_atm8_ethernet8_slip8_adv node models are used to represent an IP-based gateway running LTE and supporting up to 8 Ethernet interfaces and up to 8 serial line interfaces at a selectable data.
- The lte_enodeb_4ethernet_4atm_4slip_adv node model is used to represent a base station which is called eNodeB in LTE. This type of base station is maintained up to 4 Ethernet interfaces and up to 4 serial line interfaces at a selectable data.
- The lte_wkstn_adv node model is used to represent a workstation with source-destination application running over TCP/IP and UDP/IP.
- The PPP_DS3 link is used to represent the Ethernet connection operating 44.736 Mbps. This node is connected to two nodes in running IP. The type of this link is duplex.
- The Application_Config comprises a name and a description table which is specified different parameters for the various applications (i.e. voice and FTP applications). The individual application name is used while inventing user profile on "Profile_Config" object.
- The Profile_Config node can be used to create user profiles. These user profiles can be precise on various nodes in the network to generate application layer traffic. The applications distinct in the Application_Config are applied by this object to configure profiles. Traffic patterns can be precise followed by the application as well as the configured profiles.
- The Lte_attr_definer_adv node is used to store PHY configurations and EPS bearer definition which can be referenced by all LTE nodes in the network.
- The Mobility_Config node is used to define mobility profiles that individual nodes reference to model mobility. This node controls the movement of nodes based on the configuration parameters.

4.2.2 Network traffic Generation

In order to create an application in OPNET, an object is presented which is called *application definition attribute.* This attribute consists of predefined applications that can be customized as per the demands of the user. In application definition attribute, there are several predefined applications i.e. HTTP, E-mail, Video, FTP, Voice, Database etc.

The Application Definition attribute shown in figure 4.1 is used in the simulation model. There are two applications (FTP and VoIP) that are defined in the simulation by using the applications attributes. FTP application is modeled for set up background traffic in the simulation. The configuration parameter, as defined in Table 4.1 is used during the configuration of FTP. Since this application transfers file at a fixed interval, constant (60) is set to generate the FTP traffic. The Voice application is designed by configuring the (Voice) Table (see right part in Figure 4.1). The VoIP application uses G.711 encoder scheme and Interactive Voice (6) as the type of service for creating the VoIP calls.

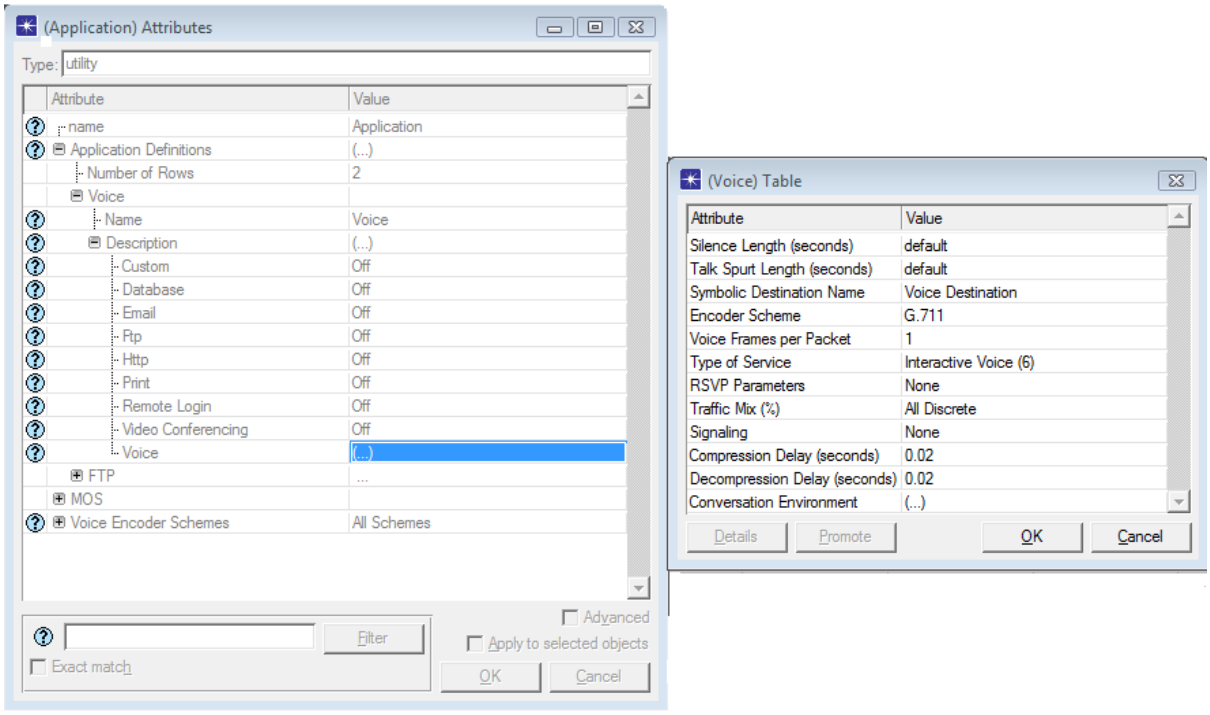


Figure 4.1 Application Definition

After configuring the application, it is necessary to configure the Profile definition where the behavior of the application is set. Figure 4.2 illustrates the Profile Definition attribute that is used in our study models. It shows the start time of the simulation, which is set to 100 (off-set‘60‘+start time‘40‘) seconds and the VoIP application is repeated continuously till the end of the simulation. It refers that VoIP calls are established between source and destination starting at 100 seconds and the calls are added continuously till the end of simulation.

Table 4.1 FTP Parameters Application

Attribute	**Value**
Command Mix	50%
Inter Request Time (Seconds)	Constant(60)
File Size (Bytes)	Constant(1000000000)
Symbolic Server Name	FTP Server
Type of Service	Best Effort(0)
RSVP Parameters	None
Back-End Custom Application	Not Used

The profile Definition is configured in a way that every VoIP call is added after a fixed time interval and the process of adding the call is continuous till the end of simulation. The first VoIP call is established at the 100^{th} second of the simulation, after that for each 1 second a VoIP call is added to the simulation. The addition of VoIP calls are prepared by repeating the voice application for every 1 second in the profile definition. This procedure is continuous till the end of simulation. In this approach the VoIP calls are increased continuously at fixed interval to the network model.

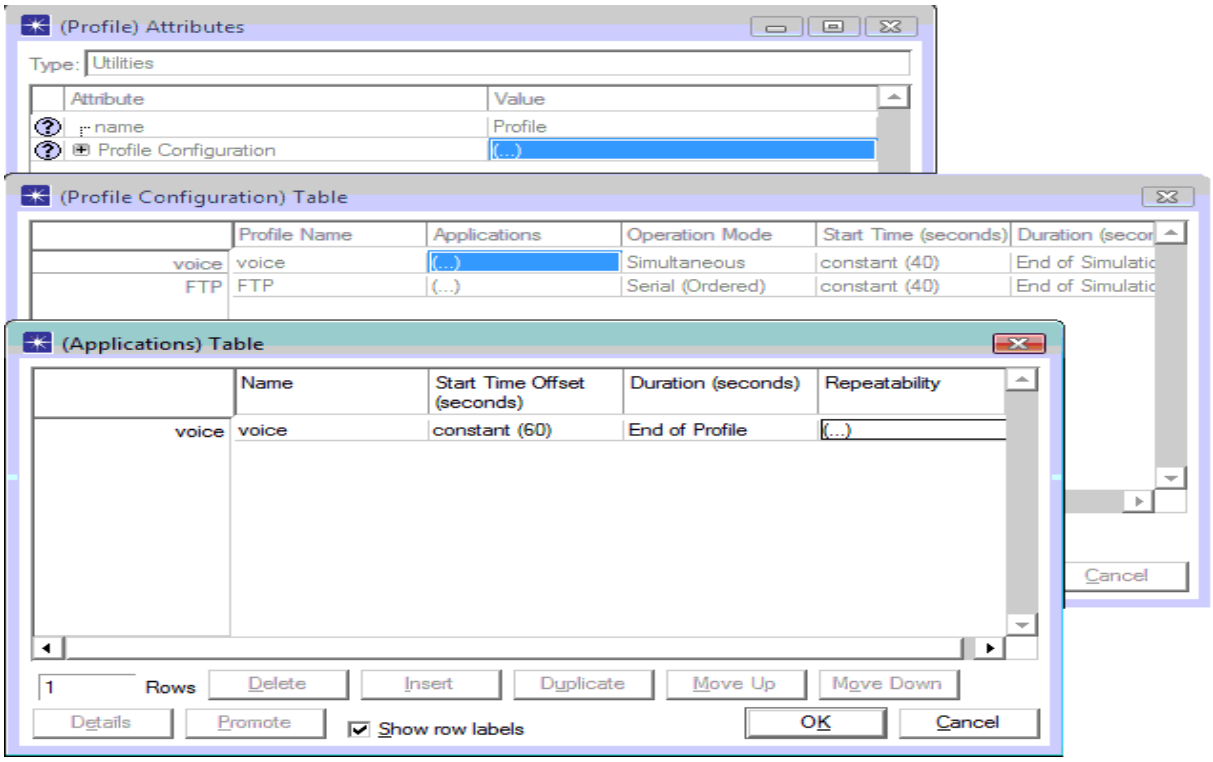


Figure 4.2 Profile Definition

4.2.3 Simulation General Parameters

Table 4.2 demonstrates the LTE general parameters used in the process of all simulation models of the study. One of the other important entities is the mobility configuration, which is used to determine the mobility model of the workstations.

Table 4.2 LTE Parameter

LTE Parameter	Value
QoS Class Identifier (Voice)	1(GBR)
QoS Class Identifier (FTP)	6(Non-GBR)
Uplink Guaranteed Bit Rate (bps)	1 Mbps
Downlink Guaranteed Bit Rate (bps)	1 Mbps
Uplink Maximum Bit Rate (bps)	1 Mbps
Downlink Maximum Bit Rate (bps)	1 Mbps
UL Base Frequency (GHz)	1920 MHz
UL Bandwidth (MHz)	20 MHz
UL Cyclic Prefix Type	7 symbols per slot
DL Base Frequency (GHz)	2110 MHz
DL Bandwidth (MHz)	20 MHz
DL Cyclic Prefix Type	7 symbols per slot

There are several appropriate parameters such as speed start time, stop time and pause time which properly control the movement of the workstations. The random waypoint mobility is used for all simulation purpose in the study. In the simulation, the speed of start time and pause time is set as constant (10) and constant (100) in seconds, respectively. On the other hand, the speed of stop time is defined at the end of simulation.

4.3 Simulation Scenarios for Throughput Performance

The following network is deployed with the help of OPNET Modeler 16.0 [31]. As mentioned in chapter 2, LTE supports scalable bandwidth i.e. 1.4, 3, 5, 10, 15 and 20 MHz. To evaluate the throughput, six different simulation scenarios are designed for scalable bandwidth. Figure 4.3 illustrates the simulation scenario based on LTE 1.3 MHz. In this scenario, two eNodeB namely eNB_1 and eNB_2 are connected to EPC (Evolved Packet Core) via PPP_DS3 links. The speed of DS3 links is 44.736 Mbps. Each eNodeB has three nodes where the nodes of eNB_1 perform as a source while the nodes of eNB_2 perform as a destination. Unlimited numbers of VoIP calls are generated in this scenario to evaluate the maximum throughput of the simulation scenario.

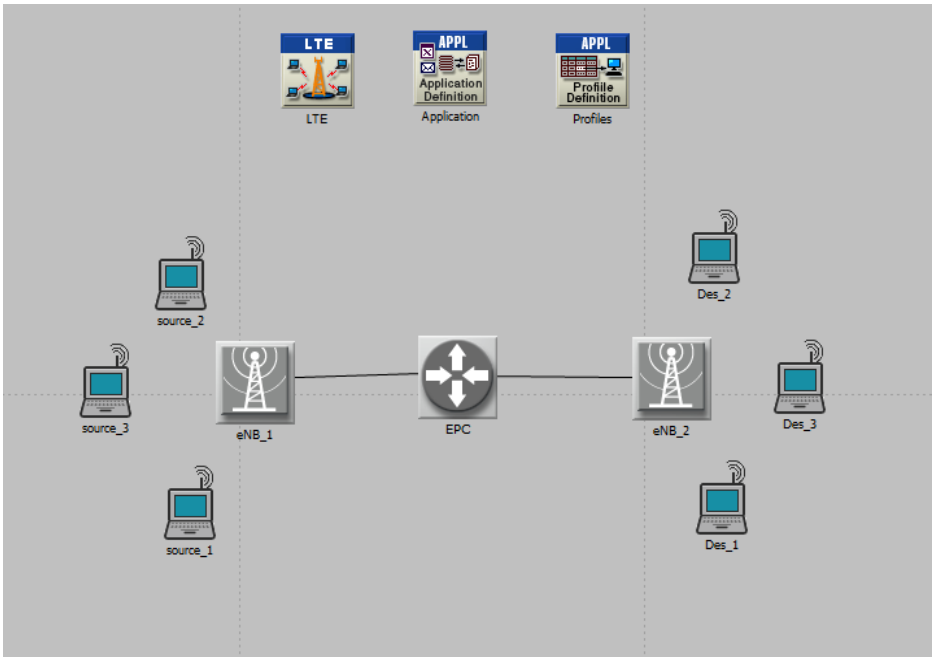


Figure 4.3 Network Scenario for LTE 1.3MHz

4.3.1 Throughput performance

The amount of data packet is delivered successfully from source node to destination over a communication network is known as throughput. The unit of throughput is bits per second (bits/sec). A throughput with a maximum rate is more often an absolute preference in each system. The efficiency of a specific method can be predicted by observing the overall throughput achieved by the network.

4.3.2 Throughput performance in scenarios

As mentioned in previous section, the start times of profile and application definition are set to 40 sec and 60 sec, respectively. No traffic is created for that reason up to 100 sec of the simulation time. This phase is known as the warm up time. A warm up time allows the queues and other aspects in the simulation to get into condition which are typical of normal running conditions in the systems [34].

Figure 4.4 demonstrates the throughput of 1.4, 3, 5 MHz bandwidth under various simulation scenarios. The X axis shows the simulation time in seconds while the Y axis represents the throughput in bits/sec. During the bandwidth of 1.3 MHz, the received maximum throughput for downlink and uplink are approximately 1,507,200 bits/sec and 1,889,920 bits/sec, respectively. Meanwhile, the received maximum throughput for 3 MHz and 5 MHz bandwidth are about 4,266,240 bits/sec and 7,808,640 bits/sec, respectively. It is important to note that the throughput of uplink and downlink are same for 3 and 5 MHz bandwidth.

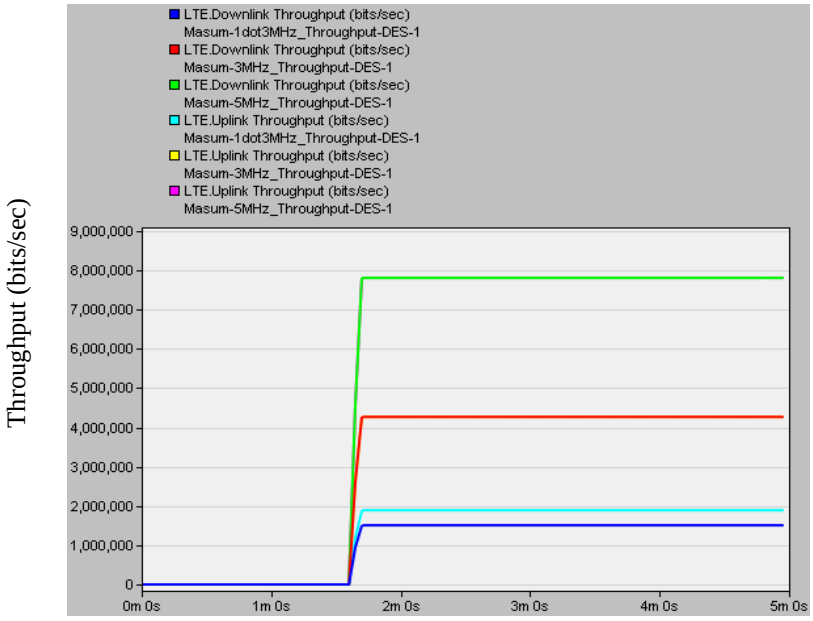


Fig. 4.4 Downlink and Uplink Throughput of 1.3, 3 and 5 MHz Scenarios

Figure 4.5 shows the throughput of 10, 15, 20 MHz bandwidth under different simulation scenarios. The uplink and downlink throughput are also equal for 10, 15 and 20 MHz bandwidth. The throughput for 10 and 15 MHz are found about 16,407,680 bits/sec and 25,145,600 bits/sec, respectively. On the other hand, the received throughput for 20 MHz bandwidth is approximately 35,887,680 bits/sec [35]. Out of the six different scenarios, 20 MHz bandwidth achieves the maximum throughput. For that reason, the bandwidth of 20 MHz is used in the upcoming simulation scenarios.

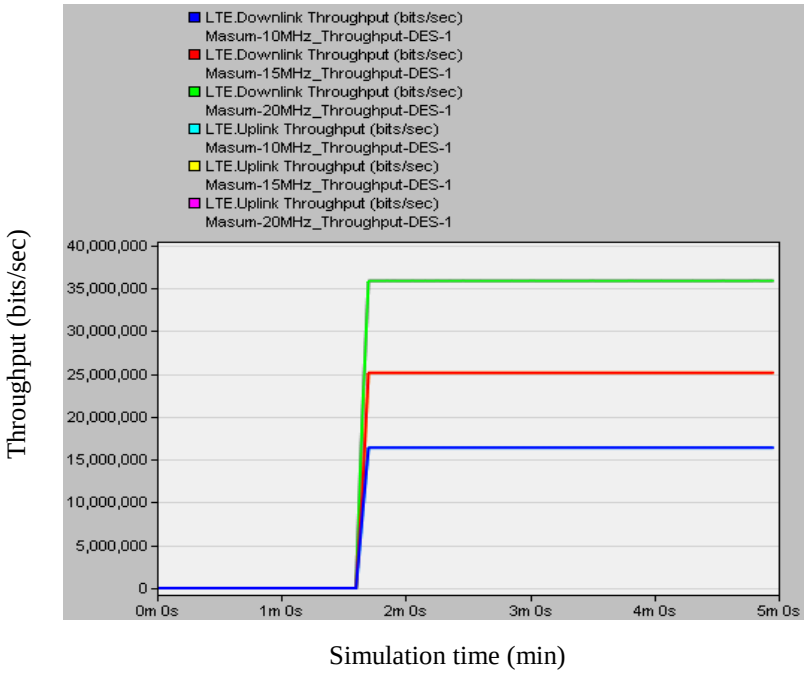


Fig. 4.5 Downlink and Uplink Throughput of 10, 15 and 20 MHz Scenarios

4.4 Simulation Design

This sections explains the network model used in this study. Three simulation scenarios have been prototyped as follows which will be briefly described in the up-coming sections. The same network topology is considered for the different simulation scenarios.

1. Scenario 1: Baseline VoIP Network
2. Scenario 2: Congested VoIP network
3. Scenario 3: VoIP congested with FTP Network

4.4.1 Scenario 1: Baseline VoIP Network

The baseline VoIP network is designed, in the OPNET simulator, by taking help of different network entities. An example of this network models is depicted in Figure 4.6 where the area of one cell radius is confined in 1 kilometer. The network elements used during the design of the simulation model are LTE configuration, application configuration, profile configuration and mobility configuration which are described in previous section. The other network elements are eNodeB, Evolved Packet Core and Workstations. This scenario objects are basically a series of network components that allow attribute definition and tuning.

In the baseline scenario, two eNodeB namely eNodeB_1 and eNodeB_2 are used which are connected with EPC by PPP_DS3 links. The speed of this link is 44.736 Mbps. Each eNodeB have five VoIP workstations where the nodes of eNodeB_1 are proceed as source node while the nodes of eNodeB_2 are served as destination node. As mentioned earlier, 20 MHz bandwidth is used in the simulaiton scenario where maximum throughput is about 35.88 Mbps. 50% traffic is generated in the baseline scenario by creating the VoIP calls in order to obtain packet end-to-end delay, traffic sent and received values. In this scenario, first call is generated at 100^{th} second of the simulation because warmup time set as 100 seconds. After

that in every 1 second a VoIP call is added in the simulation. There are four simulation cases consider based on the movement of the nodes which are listed in Table 4.3.

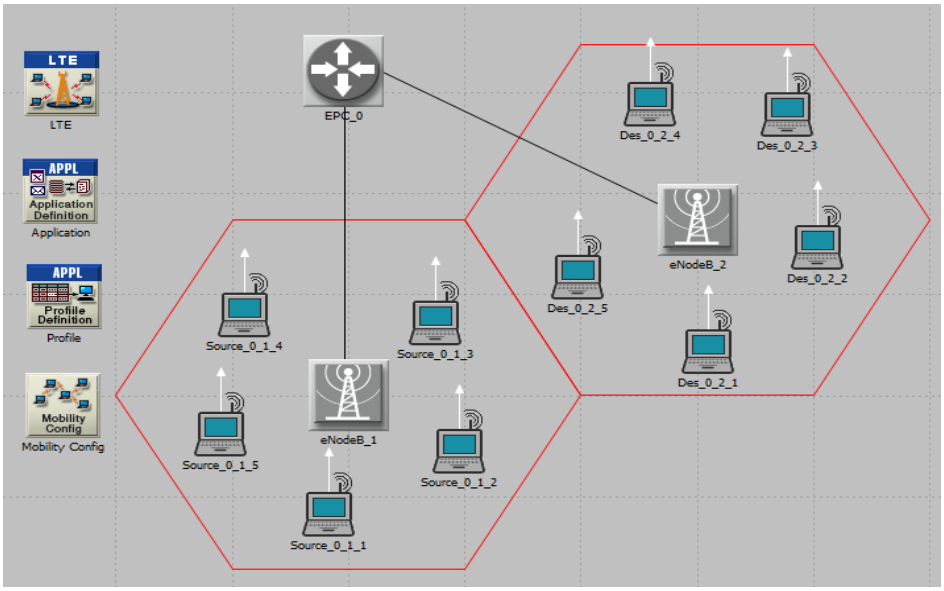


Figure 4.6 Baseline VoIP and Congested VoIP Networks

Table 4.3 Simulation case definition of Baseline VoIP Network

Case	Bandwidth (MHz)	VoIP Traffic Load (%)	Cell Radius (Km)	Speed (m/s)
1	20	50	1	Fixed(0)
2	20	50	1	10
3	20	50	1	20
4	20	50	1	50

4.4.2 Scenario 2: Congested VoIP network

Congested VoIP network is similar to what is shown in Figure 4.6. About 95% traffic is generated in this scenario which is only voice traffic. The main objective of this scenario is to see the effect on voice when the traffic is about 95%. The VoIP calls are created for 95% traffic, in order to obtain the end-to-end delay, traffic sent and traffic received values. Table 4.4 represents the simulation cases of Congested VoIP Network that are based on the mobility of the workstations.

Table 4.4 Simulation case definition of Congested VoIP Network

Case	Bandwidth (MHz)	VoIP Traffic Load (%)	Cell Radius (Km)	Speed (m/s)
1	20	95	1	Fixed (0)
2	20	95	1	10
3	20	95	1	20
4	20	95	1	50

4.4.3 Scenario 3: VoIP congested with FTP Network

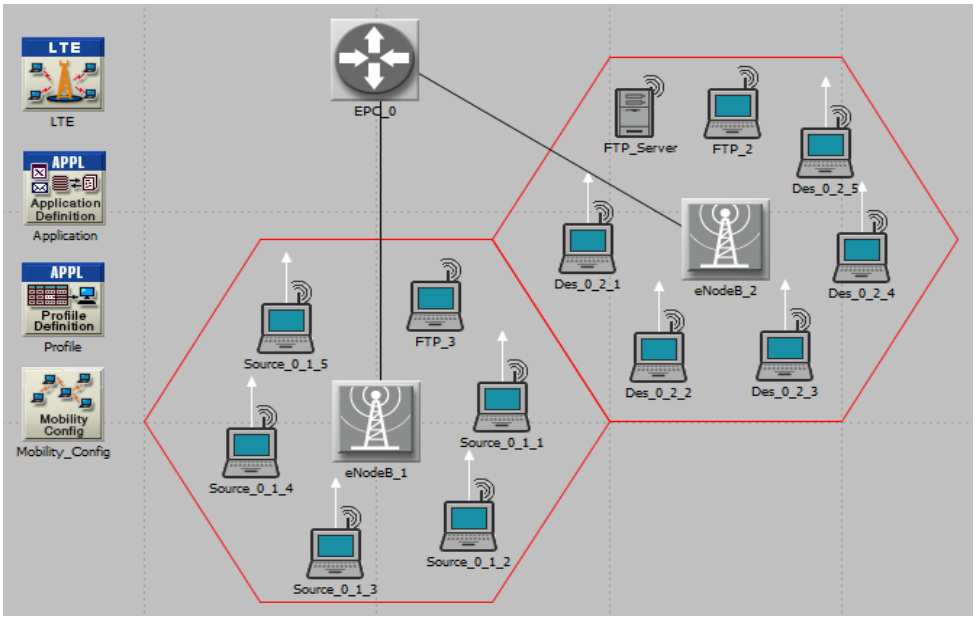


Figure 4.7 VoIP Congested with FTP Network

VoIP congested with FTP scenario is represented in Figure 4.7. Mixed traffic (VoIP and FTP) is generated in this scenario where FTP traffic is modeled in order to introduce background traffic in the simulation. Guaranteed Bit Rate (GBR) is used for voice traffic, while Non-Guaranteed Bit Rate (NGBR) is used for FTP traffic in the simulation scenario. One FTP workstation is created in each cell. The FTP application parameters are shown in previous section. It is important to mention that FTP workstations are fixed in the all the simulation case. Table 4.5 shows the simulation cases of VoIP congested with FTP Network which are considering two types of criteria i.e. the mobility of the workstations and various background traffic loads.

Table 4.5 Simulation case definition of VoIP Congested with FTP Network

Case	Bandwidth (MHz)	VoIP Traffic Load (%)	FTP File Size (Bytes)	Speed (m/s)
1	20	80	Constant (1000000000)	Fixed
2	20	80	Constant (1000000000)	20
3	20	80	Constant (2000000000)	Fixed
4	20	80	Constant (2000000000)	20

4.5 Simulation Run-Time

All the simulations run time is set as 600 seconds and all applications generate the traffic (voice) at 100 seconds of the simulated time where the calls are started in simultaneously. The simulation is implemented in OPNET Modeler 16.0 on HP laptop with Windows Vista, Pentium Duel core T4200 GHz Processor with 3GB of RAM.

CHAPTER 5

Results and Analysis

5.1 End-to-End (E2E) Delay Performance

End-to-end (E2E) delay refers to the time needed for a packet to be transmitted from one node to another node in a network. The unit of E2E delay is second. Generally, in VoIP a network, three types of delay occur during the packet transverse. Firstly, senders delay, when packets are transverse from source node. The other two are network delay and receiver delay. For VoIP applications, the packet E2E delay should not exceed 150 ms to evaluate that the quality of the created VoIP calls are accepted [29]. In this section, the packet E2E delays result for different scenarios are presented in various statistical plots. Scenario 1, 2 and 3 correspond to the Baseline VoIP, Congested with VoIP and VoIP congested with FTP, respectively. In the 1st and 2nd scenarios, voice application is designed between source and destination, while voice and FTP application are designed in the 3rd scenario. In all the scenarios, the sources and destinations happen to be started at 100 seconds as the start time of the profile and application configuration has been set to 40 seconds and 60 seconds, respectively. In all of the figures presented in following sections, X axis represents simulation time in seconds; on the other hand, Y axis represents End-to-end delay in seconds. Voice and FTP application and profile configuration are discussed in details in Chapter 4.

5.1.1 E2E Delay performance for Baseline VoIP Network

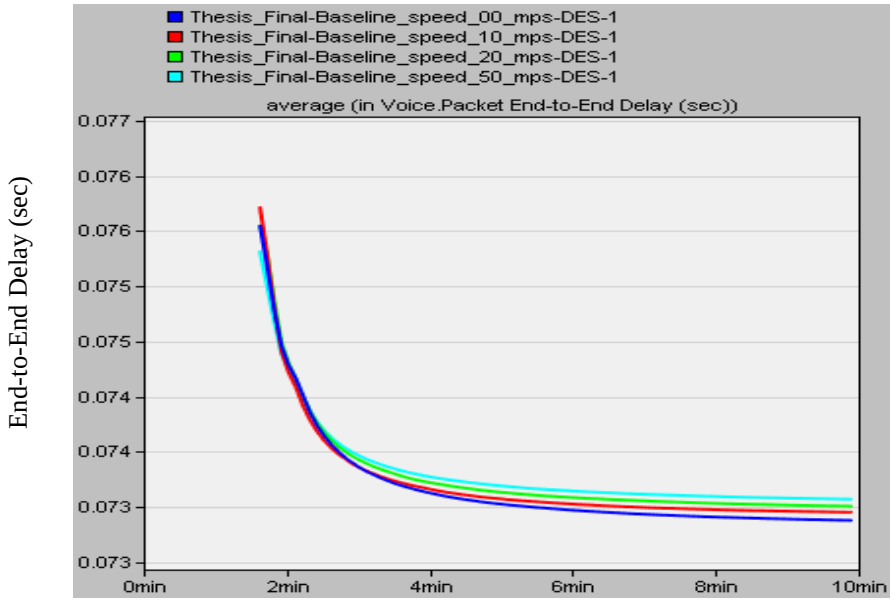


Figure 5.1 End-to-End Delay of Baseline Network

Figure 5.1 illustrates the comparable performance of the E2E delay under the different case scenarios of Baseline VoIP network. E2E delay is measured for the VoIP traffic flows between source node and destination node through Baseline network with various node speeds. The comparison against different case scenarios in terms of end-to-end delay is referred to Table 5.1 and observed in Figure 5.1.

In the Figure 5.1, the blue line shows the end-to-end delay of the scenario where the node speed is fixed(0 mps) while red, green and turquoise lines show the E2E delay of scenarios where the node speeds are 10 mps, 20 mps and 50 mps , respectively. As can be seen in Figure 5.1, the case 1 has the lowest end-to-end delay of about 73.68 milliseconds (ms) on average and varies from 73.38 ms to 76.06 ms. On the other hand, the E2E delay for the case 4 is about 73.82 ms on average, displaying a maximum delay among all the case scenarios. However, at the initial stage of the simulation time, the E2E delay for case 2 is found approximately 76.062 ms, which is even higher than case 4. With the increasing simulation time, it settles around 73.38 ms, and remains there for the rest of the time. In the Baseline scenario, the average delay for case 4 (speed 50 mps) is approximately 0.25% higher than that of case 1 scenario. All VoIP calls are accepted due to the packet E2E delay doesn't exceed the threshold value of 150 milliseconds.

Table 5.1 Summary statistics of E2E delay of Baseline Network

Scenario	Min. [s]	Avg. [s]	Max. [s]	Std Dev [s]
Case 1	7.338E-02	7.368E-02	7.606E-02	5.10E-04
Case 2	7.345E-02	7.372E-02	7.622E-02	4.97E-04
Case 3	7.351E-02	7.378E-02	7.606E-02	4.85E-02
Case 4	7.357E-02	7.382E-02	7.582E-02	4.26E-02

5.1.2 E2E Delay performance for Congested VoIP network

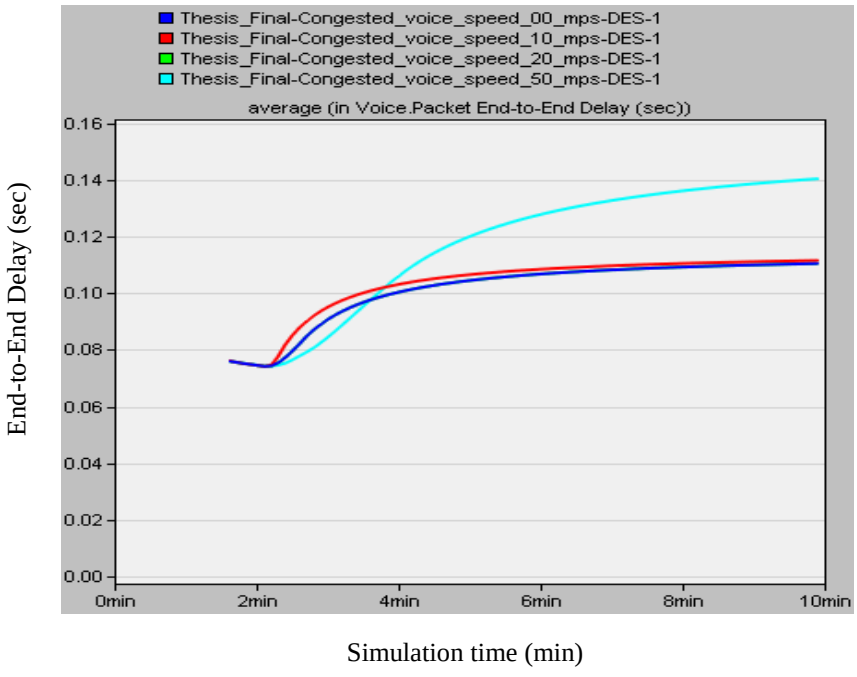


Figure 5.2 End-to-End Delay of Congested VoIP Network

Figure 5.2 displays a graphical representation of a comparative analysis on the E2E delay under the various mobility scenarios of Congested VoIP network. As mentioned earlier, there are four different scenarios based on the movement of the node where case 1 is designed with fixed node while case 2, 3 and 4 are designed with node speed 10, 20 and 50 mps, respectively. The comparison against different case scenarios in terms of end-to-end delay is observed in Figure 5.2 and referred to Table 5.2.

In the Figure 5.2, the blue line shows the end-to-end delay of the case 1 scenario while red, green and turquoise lines show the E2E delay of case 2, case 3 and case 4 scenarios, respectively. As can be seen in Figure 5.2, the congested VoIP network, with a fixed node speed, maintain a lower E2E delay level of about 101.283 milliseconds (ms) on average and varies from 74.232 ms to 110.566 ms. The topmost curve represents the case 4 scenarios, displaying a maximum delay among all the case scenarios. The E2E delay for this case is about 116.788 ms on average. Case 4 E2E delay is lower than other cases between simulation times 135 seconds to 230 seconds. The E2E delays are overlapped in case 2 and case 3. Though, at the initial stage of the simulation time, the E2E delay is higher than case 4. With the increasing simulation time, the both curves settle at 111.639 ms. In the Congested VoIP network, the average delay of case 4 is approximately 13% higher than case 1. Meanwhile, the average E2E delay of case 2 and 3 is approximately 11 % lower than that of case 4. In all scenarios, all VoIP calls are accepted due to the packet E2E delay doesn't exceed the threshold value of 150 milliseconds.

Table 5.2 Summary statistics of E2E delay of Congested VoIP Network

Scenario	Min. [s]	Avg. [s]	Max. [s]	Std Dev [s]
Case 1	7.423E-02	1.012E-01	1.105E-01	1.128E-02
Case 2	7.423E-02	1.032E-01	1.116E-01	1.103E-02
Case 3	7.423E-02	1.032E-01	1.116E-01	1.103E-02
Case 4	7.429E-02	1.167E-01	1.405E-01	2.287E-02

5.1.3 E2E Delay performance for VoIP Congested with FTP network

Figure 5.3 illustrates the comparable performance of the E2E delay under the different case scenarios of VoIP Congested with FTP network. E2E delay is measured for the VoIP traffic flows between source node and destination node through this network. In this network, our observation is to determine the effect of voice data when the FTP traffic is used to increase the load of different scenarios. The comparison against different case scenarios in terms of end-to-end delay is referred to Table 5.3 and observed in Figure 5.3.

In the Figure 5.3, the blue line shows the end-to-end delay of the case 1 scenario, while red, green and turquoise lines show the E2E delay of case 2, case 3 and case 4 scenarios, respectively. In all the cases, FTP workstations are fixed in our study. For VoIP workstations, the speed of case 1 and case 3 are fixed (0 mps), while, the mobility rate is 20 mps in case 2 and case 4. In the case 1 scenario of VoIP congested with FTP network, the E2E delay varies from 73.667 milliseconds (ms) to 105.466 ms and its average is 90.675 ms. For the case 2 scenario, the E2E delay is appeared to be higher than that of case 1 scenario. From the Table 5.3, it can be seen that the minimum E2E delay value in case 2 scenario is 73.31 ms while the maximum value is found to be 181.267 ms. The average E2E delay in such a scenario is about 143.311 ms, which is about 36% higher than case 1 scenario.

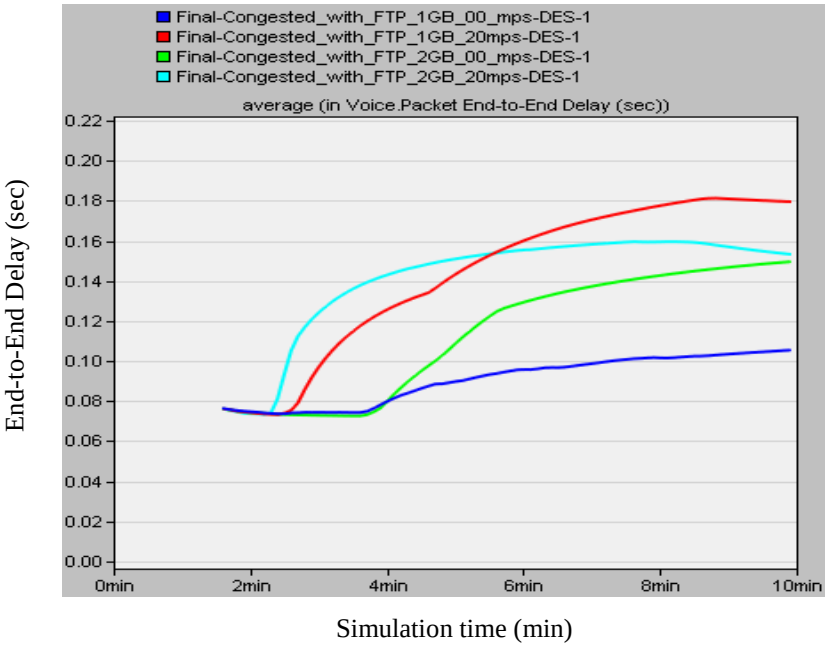


Figure 5.3 End-to-End Delay of VoIP Congested with FTP Network

For the case 3 scenario, a lower E2E delay is maintained between 100 seconds to 240 seconds, after that the curve rises gradually till end of the simulation. It can be observed that the minimum E2E delay is about 72 milliseconds, while, the maximum E2E delay is found to be approximately 149 milliseconds. The average E2E delay is about 113 ms in this scenario. In case 4, the end-to-end delay varies from 73.402 milliseconds to 159.555 milliseconds and The E2E delay for this case is 141.328 ms in average. In the VoIP congested with FTP Network, the average delay of case 4 is approximately 19% higher than that of case 3 scenario.

Table 5.3 Summary statistics of E2E delay of VoIP Congested with FTP Network

Scenario	Min. [s]	Avg. [s]	Max. [s]	Std Dev [s]
Case 1	7.367E-02	9.067E-02	1.054E-01	1.161E-02
Case 2	7.331E-02	1.433E-01	1.812E-01	3.737E-02
Case 3	7.265E-02	1.139E-01	1.495E-01	3.031E-02
Case 4	7.341E-02	1.413E-01	1.595E-01	2.642E-02

5.1.4 Summary of E2E Delay Performance

In figure 5.4, X-axis represents the four different cases while Y-axis represents the end-to-end delay in milliseconds. From the obtained results and observing in figure 5.4 the following conclusion can be drawn in terms of end-to-end delay performance between the two different network scenarios.

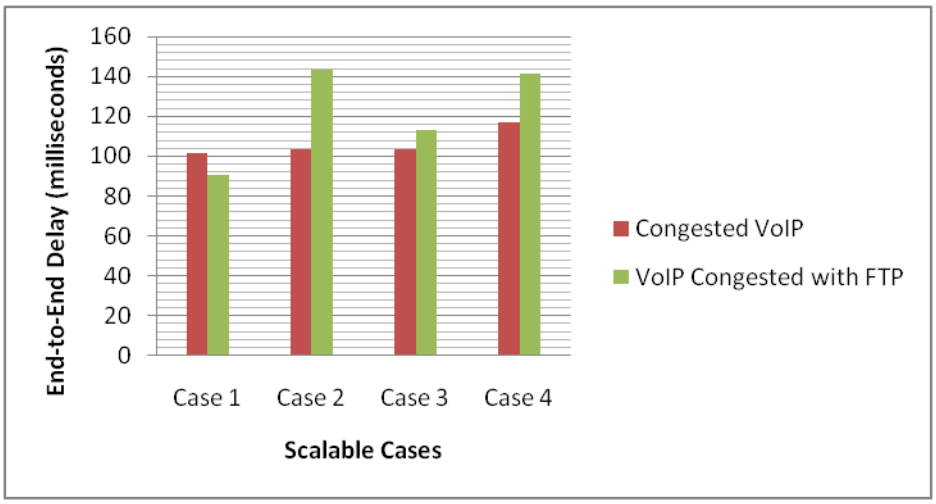


Figure 5.4 Average E2E delay between two different scenarios

For the Congested VoIP and VoIP congested with FTP network scenario (which is known in our study as scenario 2 and 3, respectively), on an average, E2E delay in the Congested VoIP network for case 1 is about 10.5% higher than that of VoIP Congested with FTP network. On the other hand, the average E2E delay in VoIP Congested with FTP network is approximately 28%, 8.5% and 17.5% higher than Congested VoIP network of the corresponding case scenarios 2, 3 and 4. In conclusion, Congested VoIP network has outperformed VoIP Congested with FTP in terms of E2E delay when the node is been moved and for the fixed node, VoIP Congested with FTP has performed well in terms of E2E delay.

5.2 Packet Loss Performance

Packet loss refers to the failure of one or more transmitted packets to reach their destination across a network. A VoIP user is satisfied if more than 98 % of its voice packets are delivered successfully. Packet loss is determined using the following equation 5.1:

$$Packet\ loss\ = \left(\frac{Send\ packet - Receive\ packet}{Send\ packet}\right) * 100 \tag{5.1}$$

In this section, the packet loss ratios for different scenarios are presented in various statistical plots. In all of the figures presented in following sections, X axis represents simulation time in seconds; on the other hand, Y axis represents the sent and received traffic in bytes per seconds.

5.2.1 Packet Loss Performance for Baseline VoIP Network

Figure 5.5 shows the sent and received traffic of different case scenarios in baseline VoIP network where X axis represents the simulation time in seconds and Y axis represents the sent and received traffic in bytes per seconds. As can be seen in figure, sent and received data are overlapped in all the cases in the simulation scenarios. It can be observed in Table 5.5 that the average sent traffic is approximately 1463285 bytes per second in all the case scenarios, while the average received traffic is about 1463236, 1463239, 1463237, 1463236 bytes per second

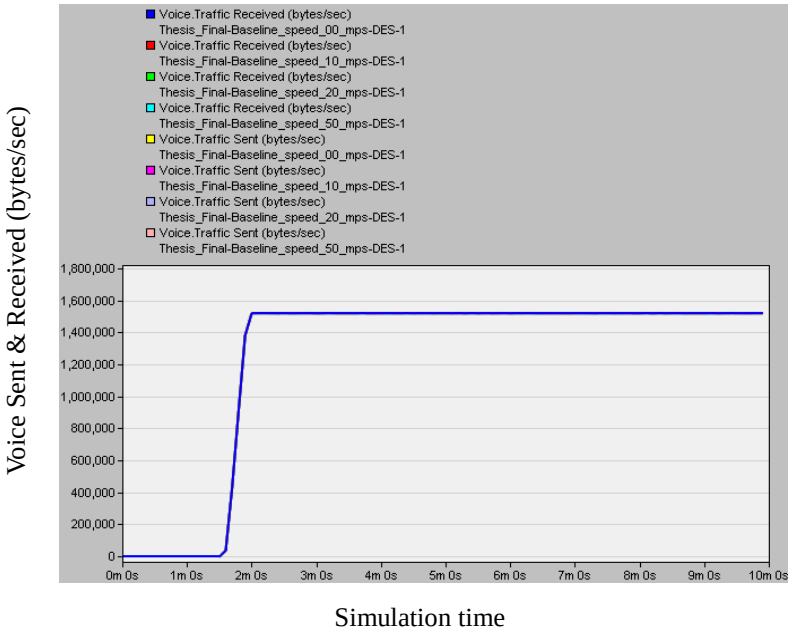


Figure 5.5 Sent and Received traffic of Baseline VoIP Network

in case scenarios 1, 2, 3 and 4, respectively. As a matter of fact, Packet loss is an important parameter to measure the quality of voice traffic. A VoIP user is satisfied if more than 98% of its speech packets are delivered successfully. From the Table 5.5, the packet loss is found about 0.003 % in all the simulation cases of Baseline VoIP Network which is very negligible.

Table5.4 Summary statistics of VoIP traffic sent of Baseline VoIP Network

Sent traffic in the case scenarios 1, 2, 3 and 4	**Min. [Bytes]**	**Avg. [Bytes]**	**Max. [Bytes]**	**Std Dev [Bytes]**
	0	1463285	1520667	262420

5.5 Table Summary statistics of VoIP traffic Received of Baseline VoIP Network

Scenario	**Min. [Bytes]**	**Avg. [Bytes]**	**Max. [Bytes]**	**Std Dev [Bytes]**	**Avg. Packet Loss (%)**
Case 1	0	1463236	1520653	262508	0.003
Case 2	0	1463239	1520573	262499	0.003
Case 3	0	1463237	1520707	262512	0.003
Case 4	0	1463236	1520587	262421	0.003

5.2.2 Packet Loss Performance for Congested VoIP Network

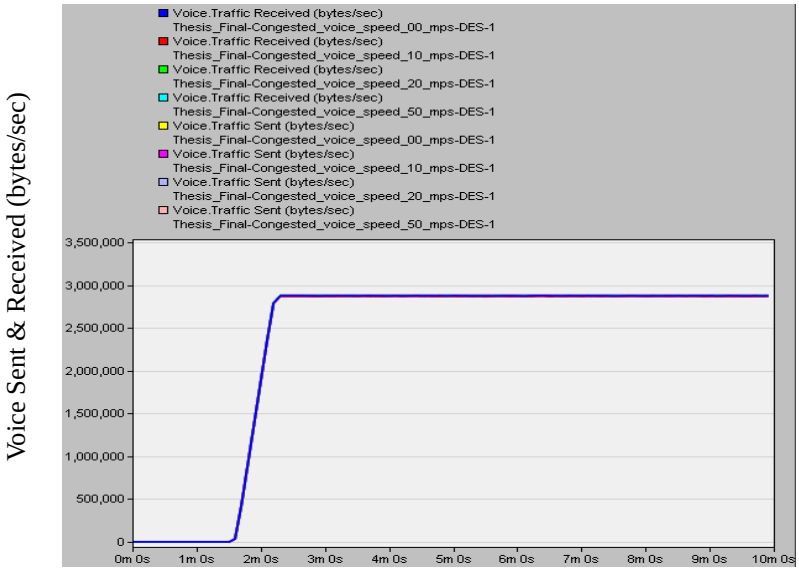


Figure 5.6 Voice Traffic Sent and Received of Congested VoIP Network

Figure 5.6 demonstrates the sent and received traffic of different case scenarios in Congested VoIP network. As can be observed in figure, sent and received data are overlapped between 100 seconds to 140 seconds in all the case scenarios. After 140 seconds when the load is around 95%, the packet loss is found in all the cases that are shown in Table 5.7. The case 1, with a node speed of 0 mps, sustains a less packet loss of about 0.07% on average; the packet loss of 0.16% is achieved when the node is moving at 10 mps and 20 mps. It increases further to 0.42% when the node speed changes to 50 mps.

Table 5.6 Summary statistics of VoIP traffic sent of Congested VoIP Network

Sent traffic in the case scenarios 1, 2, 3 and 4	**Min. [Bytes]**	**Avg. [Bytes]**	**Max. [Bytes]**	**Std Dev [Bytes]**
	0	2724546	2880933	573358

Table 5.7 Summary statistics of VoIP traffic Received of Congested VoIP Network

Scenario	**Min. [Bytes]**	**Avg. [Bytes]**	**Max. [Bytes]**	**Std Dev [Bytes]**	**Avg. Packet Loss (%)**
Case 1	0	2722538	2879627	572960	0.07
Case 2	0	2720056	2877187	572283	0.16
Case 3	0	2720056	2877187	572283	0.16
Case 4	0	2713011	2869027	570354	0.42

5.2.3 Packet Loss Performance for VoIP Congested with FTP Network

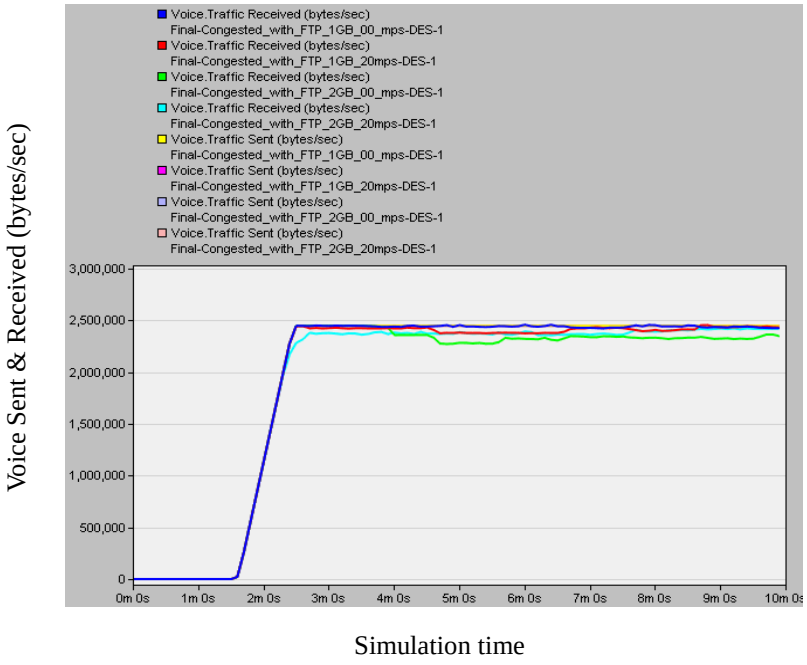


Figure 5.7 Voice Traffic Sent and Received of VoIP Congested with FTP Network

Figure 5.7 illustrates the sent and received traffic of different case scenarios in VoIP Congested with FTP network. As observed in figure 5.7, sent and received data are overlapped between 100 seconds to 130 seconds in all the case scenarios. The packet starts to drop from 130 seconds in case 4 simulation scenario. Similarly for case 2 and 3, the packet starts to drop from about 140 seconds and 230 seconds, respectively. The value of sent and received traffic is referred in Tables 5.8 and 5.9. The case 1 maintains a less packet loss of about 0.27% on average. It increases further to 1.37% when the node speed reaches 20 mps. On the other hand, the background load is increased in case 3 and 4, the packet loss observed is about 3.80% and 2.63%, respectively.

Table 5.8 Summary statistics of VoIP traffic sent of VoIP Congested with FTP Network

Sent traffic in the case scenarios 1, 2, 3 and 4	**Min. [Bytes]**	**Avg. [Bytes]**	**Max. [Bytes]**	**Std Dev [Bytes]**
	0	2279862	2448600	535033

Table 5.9 Summary statistics of VoIP traffic Received of VoIP Congested with FTP Network

Scenario	**Min. [Bytes]**	**Avg. [Bytes]**	**Max. [Bytes]**	**Std Dev [Bytes]**	**Avg. Packet Loss (%)**
Case 1	0	2273600	2457627	533217	0.27
Case 2	0	2248567	2454507	525683	1.37
Case 3	0	2193173	2448680	510259	3.80
Case 4	0	2219851	2422413	517177	2.63

5.2.4 Summary of Packet Loss Performance

The goal of this section is to evaluate the performance in terms of packet loss for the different case scenarios between Congested VoIP and VoIP Congested with FTP network. In figure 5.8, X-axis represents the four different cases and Y-axis represents the packet loss (%). From the obtained results and observing figure 5.8 the following conclusion can be drawn in terms of packet loss performance between the two different network scenarios.

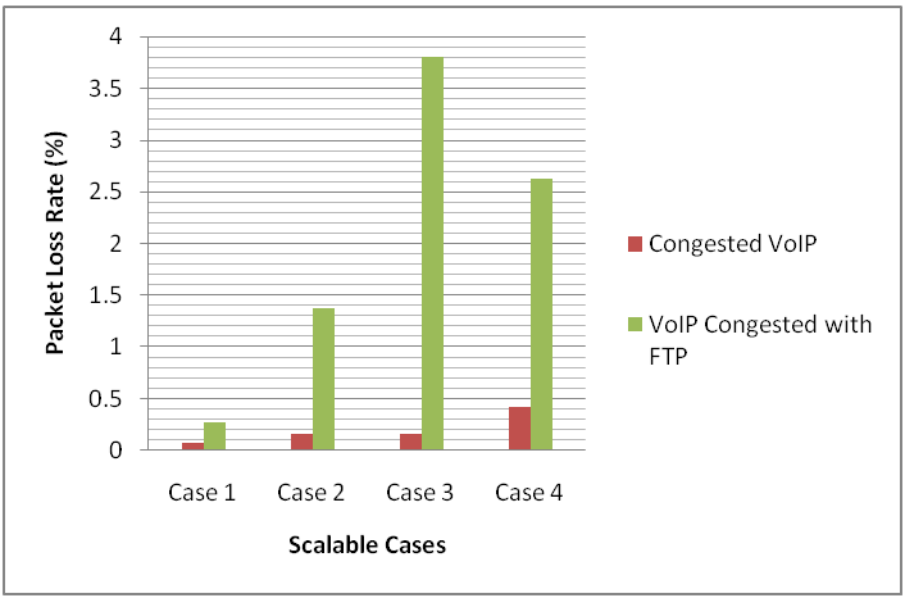


Figure 5.8 Average Packet Loss between two different scenarios

In the Congested VoIP and VoIP congested with FTP network scenario, on an average, packet loss in the VoIP Congested with FTP network for case 1 is about 74% higher than that of Congested VoIP network. In other three cases, average packet losses are 88%, 95% and 84% higher respectively. To wrap it, all of the voice traffic corresponding to four different cases in VoIP Congested with FTP experience higher packet loss than Congested VoIP network.

CHAPTER 6

Conclusion

In this thesis work, an effective study, analysis and evaluation of the end-to-end delay performance evaluation for VoIP in the LTE network have been done. The evaluation is made by simulating in OPNET Modeler 16.0 based on some performance metrics such as end-to-end delay and throughput. Three network scenarios have been simulated: Baseline VoIP network scenario, congested VoIP network scenario and VoIP congested with FTP network scenario. During simulation, answers for the research questions of this thesis have been attained in terms of the graphical results. It has been found that maximum throughput increased as the bandwidth incremented. Out of the six scenarios created for throughput measurement, the scenario with highest bandwidth (i.e. 20 MHz) exhibited maximum throughput. After that, quality measurement of VoIP has been done with respect to E2E delay, both for a network congested exclusively with VoIP and VoIP with FTP. Four scenarios have been created for this evaluation, one case with stationary node and other three cases with mobile nodes (gradually increasing the node speed). The simulation results showed that when the node is not moving, E2E delay is slightly higher for network congested with VoIP only. In other cases, better E2E is obtained for this network due to the presence of moving node. Finally, rate of packet loss for the two networks in the previously mentioned four cases have been analyzed. Packet loss remains quite minimal for congested with VoIP network regardless of the node speed. Meanwhile, for VoIP with FTP network, packet loss rate is also quite insignificant for fixed node case but the rate upshots as the node starts moving.

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