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Throughput Performance Analysis VOIP over LTE

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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 throughput performance analysis VoIP in the LTE networks. In the course of throughput performance analysis and performance evaluation, simulation approach is realized using simulation tool OPNET 16.*
Keywords: *Long Term Evolution (LTE), Voice over Internet Protocol (VoIP), Quality of Service (QoS).*

I. Introduction

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. Nowadays, communication and network technology have expanded significantly.

The of Long Term Evolution (LTE), the 4th Generation (4G) network technology release 8 specifications are being finalized in 3GPP have developed and planning to globalize extensively compared to 3rd Generation (3G) and 2nd Generation (2G) networks [2]. LTE determines goals peak data rate for Downlink (DL) 100 Mbps and Uplink (UL) data rate for 50 Mbps, increased cell edge user throughput, improved spectral efficiency and scalable bandwidth 1.4 MHz to 20 MHz [6]. 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 IP application is used to represent the class of inelastic, real-time interactive applications that is sensitive to 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 throughput [7].

II. Generic Frame Structure:

Two types of generic frame structures are designed for radio access network of LTE [3], 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.

2.1 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 (1). In FDD mode, there are two carrier frequencies domain, one for uplink direction (f_{HL}) and another for downlink direction (f_{DL}). The frames of uplink and downlink are transmitted simultaneously.

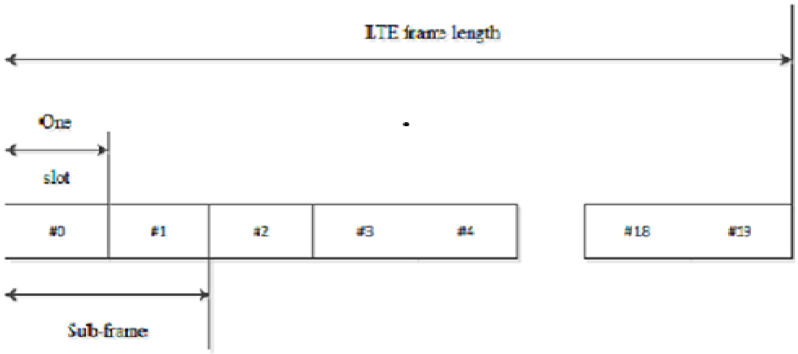


Figure 1 Type-1 LTE Frame Structure

2.2 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 frames have further divided into 5 sub frames which is equal duration of 1 ms as illustrated in figure (2). 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.

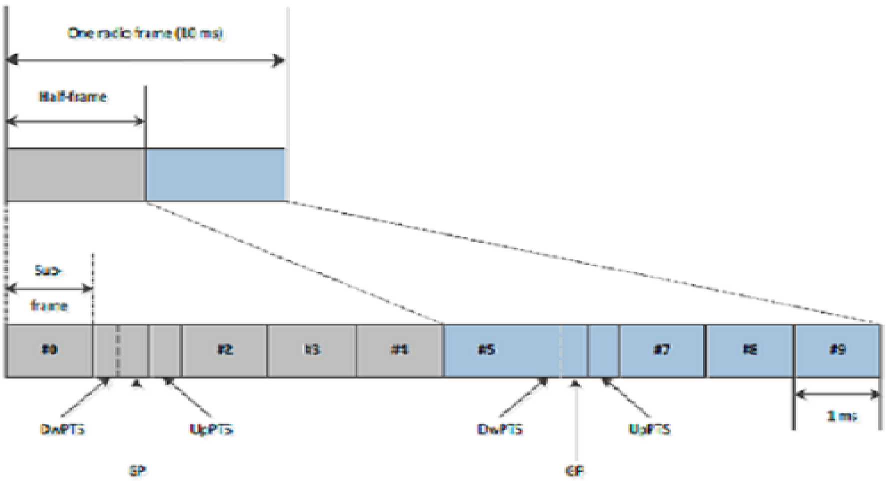


Figure 2 Type-2 LTE Frame Structure

2.3 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) [1,2] . 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, ad-mission thresholds, configuration of link layer protocol and queue management thresholds) preconfigured through the operator on behalf of each network element [4]. The class-based technique applies in the LTE system to improve the scalability of the QoS frame work. 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.

Table 1 LTE standardized QCI characteristics

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			

2.4 Real time Transport Protocol:

IEFT developed many standardized network protocols; Real-time Transport Protocol (RTP) is one of them for audio and video transmission [5]. It was originally designed for multicast protocol published in 1996, although, this protocol is now widely used in uncast 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 [5].

III. 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 [9].

Table 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

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 packets origin, projected destination address and have the packet stream to be reconstructed in the correct order with the help of times tamp. 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 worlds current phone systems.

3.1 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, the 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) [8]. 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 and R1-070674 is updated in contribution [10]. 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 total users are satisfied. A VoIP user is satisfied if more than (98) of total speech packets are delivered successfully. It is required for VoIP user that the packet delay should not exceed 150 milliseconds [6].

IV. Methodology:

The main objective of this thesis work is to evaluate the Throughput performance in terms of application such as VoIP in the LTE networks. OPNET 16 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 over LTE networks. To understanding how way we can do the configuration in the LTE environment and set their networks attributes into the OPNET 16.
- To select the quantitative metrics such as throughput.
- Discussing the different constraints that affect the throughput performance of VoIP in LTE network and critically examine various approaches that are suggested in the literature for decreased the throughput performance.
- Developing, testing and evaluating strategic scenario in OPNET.
- 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.

4.1 The Problems:

This thesis covers the technical issues and factors that need to be considered for the implementation of VoIP over LTE and their functionality and design parameters of the LTE networks. In this thesis, qualitative and quantitative analysis of throughput performance over LTE networks have been done in a simple and understandable fashion using simulation tools (OPNET 16).

4.2 Research Methodology:

The research methodology presented in this thesis is based on three steps that considered:

- Identify the key factors influencing Throughput performance in LTE networks.
- Determine the performance LTE Throughput and DL and UL Throughput Calculation for LTE FDD.
- Determine the suitable VoIP model to support real time application.
- In this Quantitative approach, following four steps are considered:
- The LTE network models are designed in the OPNET simulator based on different network entities.
- Evaluate the performance of different simulation scenarios in terms of VoIP when LTE is deployed.

- Collect quantitative data regarding throughput for analysis of the network performance.
- The simulation results are collected using OPNET in terms of different statistical graphs and tables.

4.3 LTE Throughput:
From the 3GPP specification:

- 1 Radio Frame (10ms) = 10 Sub-frame.
- 1 Sub-frame = 2 Time slots.
- 1 Time-slot = 0.5 ms (i.e. 1 Sub-frame = 1 ms).
- 1 Time-slot = 7 Modulation Symbols (when normal CP length is used).
- 1 Modulation Symbols = 6 bits; if 64 QAM is used as modulation scheme. Radio resource is managed in LTE as resource grid Radio resource is managed in LTE as resource grid.
- 1 Resource Block (RB) = 12 Sub-carriers. Assume 20 MHz channel bandwidth (100 RBs), normal CP. Therefore number of bits in a sub-frame = 100RBs x 12 sub-carriers x 2 slots x 7 modulation symbols x 6 bit = 100800 bits. Hence, data rate = 100800 bits / 1 ms = 100.8 Mbps.
- If 4x4 MIMO is used, then the peak data rate would be 4 x 100.8 Mbps = 403 Mbps.
- If 3/4 coding is used to protect the data, we still get 0.75 x 403 Mbps = 302 Mbps as data rate.

4.4 DL and UL Throughput Calculation for LTE FDD:
The FDD system has a paired spectrum, same bandwidth for Downlink as well as for Uplink. 20 MHz FDD system have 20MHz for Downlink and 20MHz for Uplink. For throughput calculation, suppose:

- Bandwidth 20MHz.
- Multiplexing scheme - FDD.
- UE category- Cat 3.
- Modulation supported as per Cat 3 TBS index 26 for DL. (75376 for 100RBs) and 21 for UL (51024 for 100 RBs). So the throughput can be calculated by a simple formula: Throughput = Number of Chains x TB size.
- So DL throughput = 2 x 75376 =150.752 Mbps.4.
- UL throughput = 1 x51024 =51.024 Mbps. As we have 2 receive chains and one transmits chain [11].

V. Simulation and Discussion:

Table (3) 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 3 LTE Parameter

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

5.1 Simulation Scenarios for Throughput Performance:
The following network is deployed with the help of OPNET Modeler 16.0. LTE supports scalable bandwidth i.e. 1.4, 3, 5, 10, 15 and 20 MHz. To evaluate the throughput, 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.

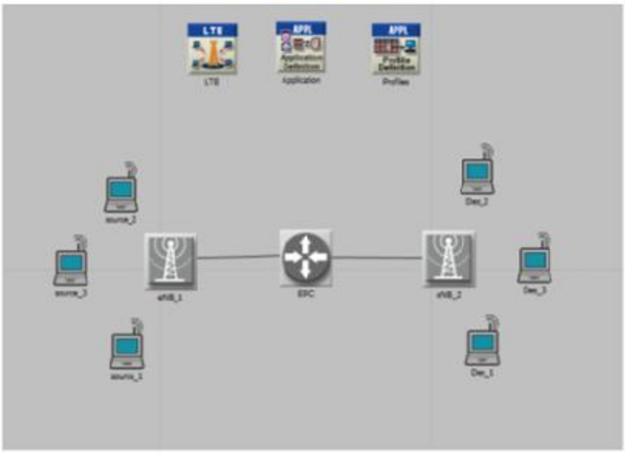


Figure 3 Network Scenario for LTE 1.3MHz

5.2 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.

5.3 Throughput performance in scenarios:

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. Figure (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.

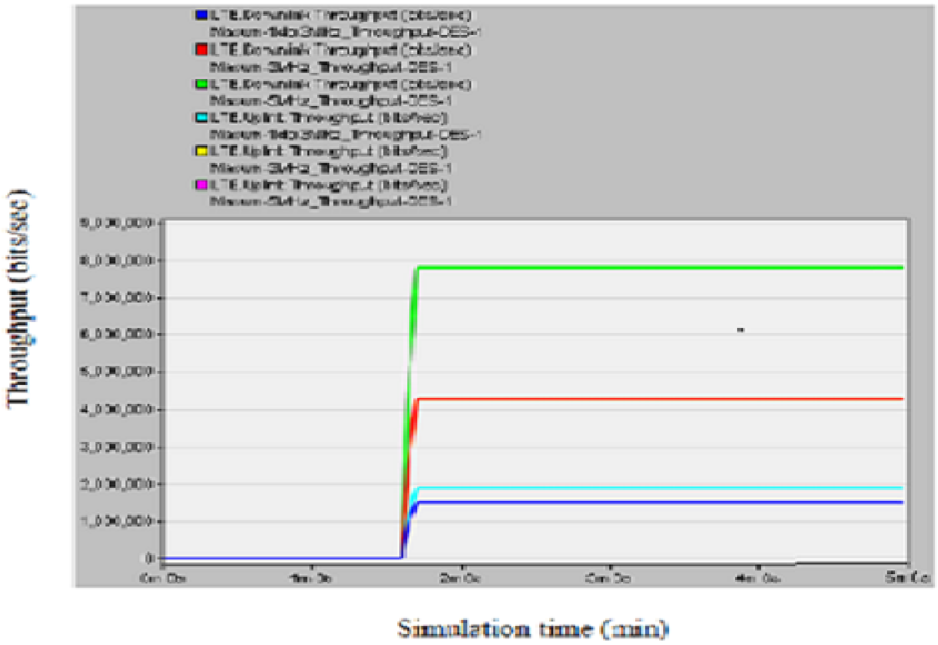
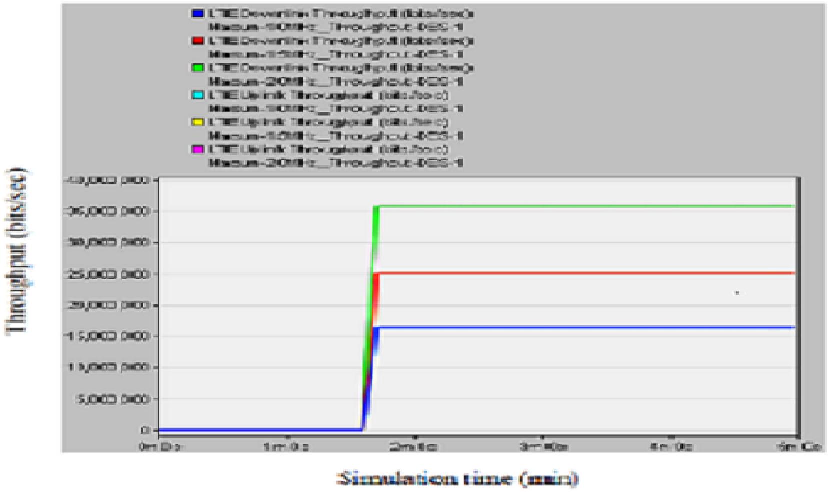


Figure 4 Downlink Uplink Throughput of 1.3,3 and 5 MHz Scenarios

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.

Figure (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.
Out of the six different scenarios, 20 MHz bandwidth achieves the maximum throughput.



Figuer 5 Downlink and Uplink Throughput of 10, 15 and 20 MHz Scenarios

5.4 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 (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 precede as source node while the nodes of eNodeB-2 are served as destination node. As mentioned earlier, 20 MHz bandwidth is used in the simulation scenario where maximum throughput is about 35.88 Mbps.

Half of traffic is generated in the baseline scenario by creating the VoIP calls in order to obtain packet through put, traffic sent and received values. In this scenario, first call is generated at 100th second of the simulation because warm up 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.

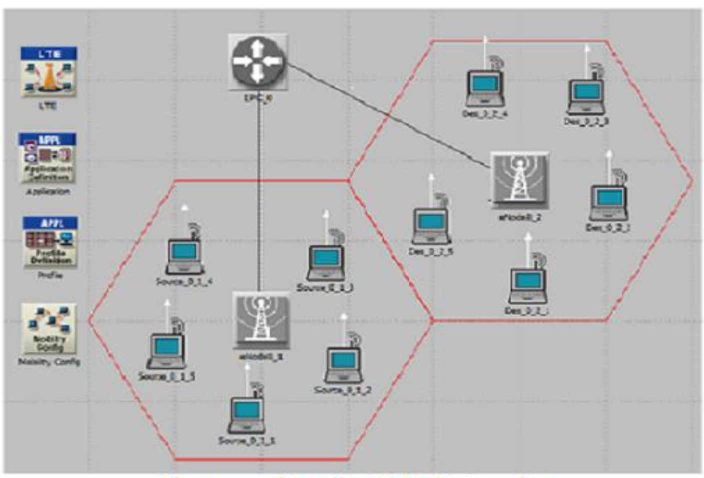


Figure 6 Baseline VoIP Networks

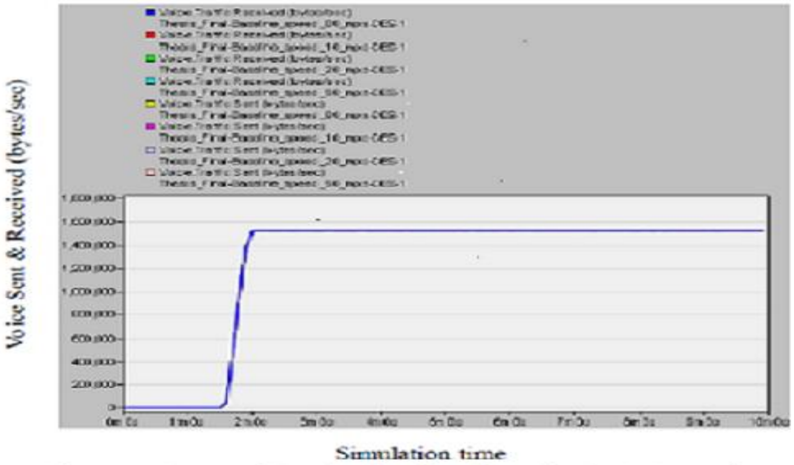


Figure 7 Sent and Received traffic of Baseline VoIP Network

Table 4 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

Table 5 Summary statistics of VoIP traffic Received of Baseline VoIP Network

Min. [Bytes]	Avg. [Bytes]	Max. [Bytes]	Std Dev [Bytes]	Avg. Packet Loss(%)
0	1463236	1520653	262508	0.003
0	1463239	1520573	262499	0.003
0	1463237	1520707	262512	0.003
0	1463236	1520587	262421	0.003

VI. Conclusion

In this thesis work, an effective study, analysis and evaluation of the through put performance evaluation for VoIP in the LTE network has been done. The evaluation is made by simulating in OPNET Modeler 16.0 based on some performance metrics.

Tow network scenarios have been simulated: Throughput Performance, Baseline VoIP network scenario. 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.

The demand for the higher quality and through put 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.

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