Performance Evaluation for Voice over LTE by using G.711 as a Codec

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***Abstract* -LTE is the advanced technology for mobile network that operates completely in packet domain with less component compared to previous technology like HSPA or HSPA+ and it's provide a very high data throughput in both direction upload or download. This development from network side was flowed by development from user side, actually the mobile phone has been changed to the Smart-Phone with a high number of application that use network technology like Video application, VoIP application, Game. This new technology forces the researcher and the developer to optimize the network settings to have a good performance.**

This paper perform a study and analysis of the Quality of Experience (QoE) at the end user for Voice over the LTE network (VoLTE), after the implementation of VoLTE by using G.711 as a voice codec and will use IP as a core network, will analyze the network performance in order to sort out if this codec is the most recommended codec for VoLTE environment.as per IUT recommendation.

Keywords-component: LTE, QoE, IP, VoIP, VoLTE, G.711.

I. INTRODUCTION

Since the apparition of 3rd generation mobile network the usage of the mobile technology PS part is not limited anymore to voice for data usage such as Mail and ordinary internet, but it's become more complex and developed with the usage of the video over IP and Voice Over IP named by VoHSDPA for 3G+ technology and VoLTE for LTE technology.

The recent mobile technology is LTE that provide a high data speed in uplink and downlink with high level and strict quality of service provided because of usage of intelligent entities in the network starting from the eNodeB to the core entities that use router from 4th and 5th generation. For that our research team aim to complet the previous work done by sorting out if the G711 codec recommended by IUT specification for VoLTE usage

In the previous paper of the research team, their finding was that G.729 codec is the best compatible codec for VoIP utilization, the performance of VoIP using G.729 is the better compared to the other VoIP codec like G.723 and G.711 [7].

The actual work and interest of the research team, presented in this paper will study in detail the performance of VoLTE when using G711 as a codec after implementation on the most important software for LTE simulation named Opnet 17.6

After introducing the paper, the rest of the paper is organized as follows, Section II presents theory background about VoIP technology, Section III presents principal of LTE technology and the different entities. Different simulation scenarios are presented in Section IV. Results and discussion are presented in section V, and finally a general conclusion of this work is presented in Section VI.

II. VOIP TECHNOLOGY

A. *VoIP Transport System*

Different network functions are required for delivering VoIP traffic over data networks. These functions include call signaling and control and media transport. Different signaling protocols are used, with the common ones being H.323 and Session Initiation Protocol (SIP) [9], [2]. These protocols are used to setup, control and terminate a VoIP call session. Once a session is established, voice stream is transported using two transport protocols, namely Real-Time Protocol (RTP) and User Datagram Protocol (UDP) over IP. Whereas UDP provides a connectionless service suitable for real-time transmission, RTP provides a means for carrying real-time details such as packet sequence and time-stamping.

B. VoIP Codecs

RTP and UDP are used to carry voice signals that are digitally encoded. This means that each voice signal is converted from its analog form into a digital form at the transmitter device. The analog signal is firstly sampled based on a sampling rate of 8 KHz, and then a quantization process takes place in which each sample is represented by 8 or 16 bits. Next, the output is encoded according to many factors: the compression rate and the framing time or the frames length. Finally, one or more of these frames are encapsulated into an RTP/UDP/IP packet for transmission over the network. All these processes are accomplished by one of various audio codecs, each of which uses a different algorithm. Table I shows some features of the most common codecs: G.711, G.723.1 and G.729, and how they vary in many aspects such as bit rate, encoding algorithm and coding delay [6][2].

C. QoS Requirements of VoIP

Ensuring high voice call quality over the "best effort" IP network is the key challenge in delivering VoIP traffic. QoS for VoIP is defined using different parameters, with the common ones being end-to-end delay, jitter and packet loss [9]. The most metrics is:

End to End Delay: End-to-end delay consists of end-system and network delay. The end-system delay occurs due encoding and decoding delay and de-jitter buffering delay. The network delay is mainly caused by the propagation and queuing delay at network devices. A one-way end-to-end delay of 150ms is the recommended delay value defined by the G114 recommendation of the International Telecommunication Union (ITU-T). It also considers a delay of 400ms is the maximum acceptable value. However, the recommendation also shows a measurement of user satisfaction against the acceptable end-to-end delay. It shows that after a delay of about 260 ms, there will be unsatisfied users because the delay starts to be noticeable. In this paper, an end-to-end delay of 260ms is considered to be the maximum acceptable delay [7].

Jitter: the end-to-end delay variation between two consecutive packets is called jitter. A jitter of less than 50ms is considered to be acceptable for high quality VoIP calls. Packet loss occurs in the network due to many factors such as congestion network failure. It is also caused by end systems when the jitter buffer is overloaded. a packet loss rate upper than 10% is tolerable for high bit rate codecs such as G.711 whereas 5% is considered to be the acceptable limit for low bit rate codecs such as G.729A [7].

Throughput: Throughput refers to how much data can be transferred from source to destination in a given amount of time. It depends upon the bandwidth also. Throughput is a measure of data rate (bits per second) generated by the application.

D. SIP Architecture

The Session Initiation Protocol (SIP), defined by Internet Engineering Task Force (IETF), is an application- layer control (signaling) protocol for establishing, modifying and terminating sessions with one or more participants. These sessions may be Internet telephone calls, multimedia distribution or multicast conferences. It has been standardized within IETF for the invitation to multicast conferences and Internet telephone calls. In the context of SIP, a user is usually identified by an email- like address such as cleint@sip.ma, where "client" is a user name or phone number, and "sip.ma" is a domain name or numerical address of the user [9].

The main entities in SIP are **User Agent (UA), Proxy Server, Redirect Server** and **Registrar**:

User Agent (UA) is the endpoint entity. User Agents initiate and terminate sessions by exchanging requests and responses. RFC 2543 defines the User Agent as an application, which contains both a user agent client and user agent server as follows:
User Agent Client (UAC): a client application that initiates SIP requests.
User Agent Server (UAS): a server application that contacts.
Registrar is a server that accepts REGISTER requests for the purpose of updating a location database with the contact information of the user specified in the request.
Proxy Server is an intermediary entity that acts as both a server and a client for the purpose of making requests on behalf of other clients.
Redirect Server is a server that accepts a SIP request, maps the SIP address of the called party into zero (if there is no known address) or more new addresses and returns them to the client.

III. LONG TERME EVOLUTION NETWORK (LTE)

A. LTE description

The main function of IP Network is to send the user data from the source to destination. Data is constructed as a series of packets. All the packets are routed through a chain of routers and multiple networks to reach the destination. In the Internet, router takes independent decision on each incoming packet. When a packet reaches a router, it forwards the packet to the next hop depending on the destination address present in the packet header. The process of forwarding the packets by the routers is done until the packet reaches the destination.

3GPP Long-term Evolution (LTE) [1] is the latest standard in the GSM/UMTS line specified in 3GPP Release 8. It replaces the WCDMA transmission scheme of UMTS so that OFDMA (Orthogonal Frequency-Division Multiple Access) is used for downlink while SC-FDMA (Single-carrier FDMA) is used for uplink traffic.

Orthogonal frequency-division multiplexing (OFDM) is an FDM type of scheme that is used as a digital multi-carrier modulation method where a number of closely spaced orthogonal sub-carriers are used to carry data. The data is divided into several parallel data streams or channels, one for each sub-carrier. A flexible resource allocation is achieved through dynamic assignment of sub-carriers to a specific node. Each sub-carrier is modulated with a conventional modulation scheme at a low symbol rate. Furthermore, MIMO (multiple-input, multiple-output) antenna technology is used in LTE. Minimum transmission time interval (TTI) is 1 ms and 64QAM was added as a modulation scheme[1].

The Dedicated Traffic Channel (DTCH) in LTE is mapped to DLSCH and ULSCH (Downlink Shared Channel and Uplink Shared Channel) respectively. It uses HARQ and adapts dynamically to the link quality. Spectrum flexibility was an important design goal for LTE and it was built to scale using bandwidths ranging from 1.4 MHz to 20 MHz in both paired and unpaired configurations. A wide range of frequency bands are expected to be used for LTE including the 700 MHz band allowing for indoor usage and wide coverage. LTE provides data rates up to 100 Mbits/s in the downlink direction, uplink data rates up to 50 Mbps in the uplink direction and latencies in the radio access network at 10 milliseconds[1].

B. LTE architecture

The system is non-backward compatible with GSM or UMTS and hence requires a new infrastructure. The upgraded version LTE Advanced is designed to meet the requirements from the fourth generation (4G) radio access network of 1 Gbits/s in data rate for stationary applications and 100 Mbits/s for mobile applications. The first commercial LTE network was opened in Stockholm and Oslo in December 2009. A wide range of frequencies are expected to be used. The structure of LTE networks is changed radically from the GSM and UMTS network structures see figure 1. eNB(Evolved NodeB) is the only node type in EUTRAN (Evolved UTRAN) responsible for all radio interface-related functions. Main node types in the EPC (Evolved Packet Core) are the MME (Mobility Management Entity) responsible for mobility, UE (User Equipment) identity, and security management functions, the S-GW (Serving Gateway) terminating the interface towards E-UTRAN, and the PGW (PDN Gateway) terminates the interface towards the packet data network (PDN) [12].

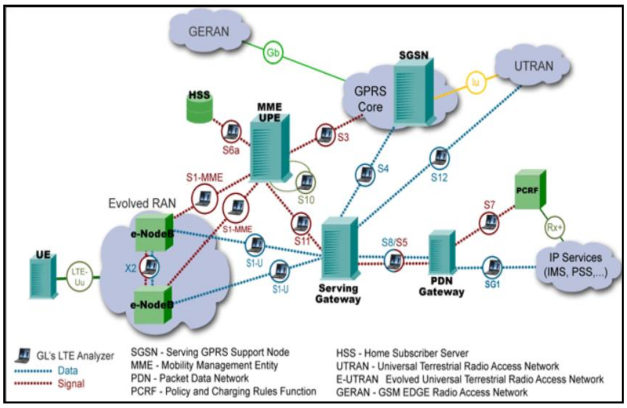
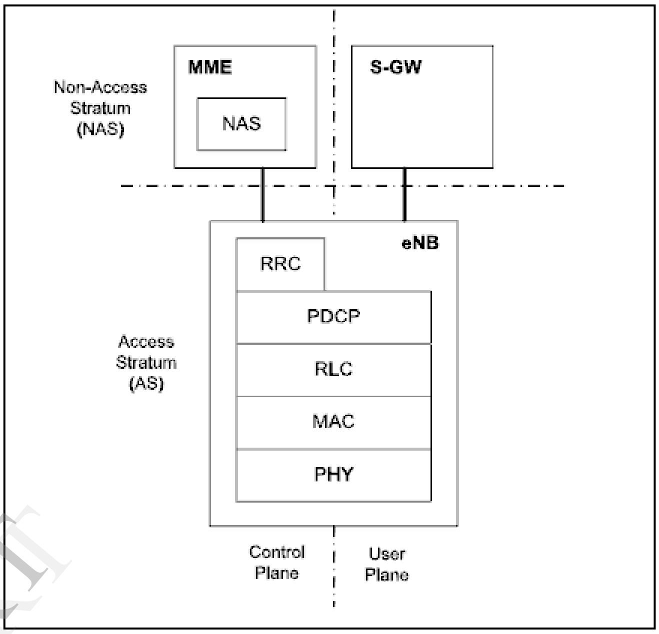


Figure 1: LTE Network architecture

1) LTE E-UTRAN overview

eNBs provide the E-UTRAN with the necessary user and control plane termination protocols. Fig. 2 gives a graphical overview of both protocol stacks. In the user plane, the protocols that are included are the Packet Data Convergence Protocol (PDCP), the Radio Link Control (RLC), Medium Access Control (MAC), and Physical Layer (PHY) protocols. The control plane stack additionally includes the



Radio Resource Control (RRC) protocols[12].

Figure 2: LTE E-UTRAN Protocol stack

The main principal functionalities of each layer are summarized in the following

- NAS (Non-Access Stratum)
  - Connection/session management
  - Authentication.
  - Registration.
  - Bearer context activation/deactivation.
  - Location registration management.
- RRC (Radio Resource Control)
  - Broadcast system information
  - Management of RRC connection.
  - Security functions including key management.
  - Mobility functions.
  - QoS management functions.

  - UE measurement reporting and control of the reporting.
- RLC (Radio Link Control)
  - Error correction through Automatic Repeat request (ARQ).
  - Segmentation according to the size of the transport block and re-segmentation in case a retransmission is needed.
  - Concatenation of SDUs for the same radio bearer.
  - Protocol error detection and recovery.
  - In-sequence delivery.
- MAC (Medium Access Control)
  - Multiplexing/demultiplexing
  - Scheduling information reporting.
  - Error correction through Hybrid ARQ (HARQ).
  - Local Channel Prioritization.
  - Padding.

2) Evolved Packet Core overview

The EPC is all-IP-based core network that can be accessed through 3GPP radio access (UMTS, HSPA, HSPA+,LTE) and non-3GPP radio access (e.g. WiMAX, WLAN), allowing handover procedures within and between both access types. The access flexibility to the EPC is attractive for operators since it enables them to have a single core through which different services are supported. The main components of the EPC and their functionalities are as follows.

- Mobility Management Entity (MME)

This is a key control plane element. Among other functions, it is in charge of managing security functions (authentication, authorization, NAS signaling), handling idle state mobility, roaming, and handovers. Also selecting the Serving Gateway (S-GW) and Packet Data Network Gateway (PDN-GW) nodes is part of its tasks. The S1-MME interface connects the EPC with the eNBs.

- Serving Gateway (S-GW)

The EPC terminates at this node, and it is connected to the E-UTRAN via the S1-U interface. Each UE is associated to a unique S-GW, which will be hosting several functions. It is the mobility anchor point for both local inter-eNB handover and inter-3GPP mobility, and it performs inter-operator charging as well as packet routing and forwarding.

- Packet Data Network Gateway (PDN-GW)

This node provides the UE with access to a Packet Data Network (PDN) by assigning an IP address from the PDN to the UE, among other functions. Additionally, the evolved Packet Data Gateway (ePDG) provides security connection between UEs connected from an untrusted non-3GPP access network with the EPC by using IPSec tunnels.

IV. SIMULATION SCENARIOS

OPNET 17.6 Simulator is used to simulate VoLTE architecture. OPNET (Optimized Network Engineering Tools) [7] is a discrete event simulation tool, which provides a comprehensive development environment supporting the modeling and simulation of communication networks and which contains data collection and data analysis utilities. OPNET allows a various numbers of closely spaced events in a sizeable network to be represented accurately. It uses a modeling approach where networks are built of nodes interconnected by links. Each node's behavior is characterized by the constituent components and these components are modeled as a state-transition diagram [1].

In this paper presents we present an architecture of VoLTE that use 6 eNodeB with 9 LTE UE that support VoLTE for E-UTRAN part. A core network the type Nexus 7000 attached to an IP backbone. Furthermore, two items were added and configured to allow VoLTE clients and servers to run VoLTE, these are the Application Configuration and Profile Configuration objects. The Application Configuration object provides predefined applications such as VoIP with configurable parameters. On the other hand, the Profile Configuration object defines the behavior of the application regarding the start and the end time and repeatability.

For LTE parameters such as used quality of service, used Bandwith and used frequency is defined in LTE configuration item

The figure 3 shows the general architecture for our simulation.

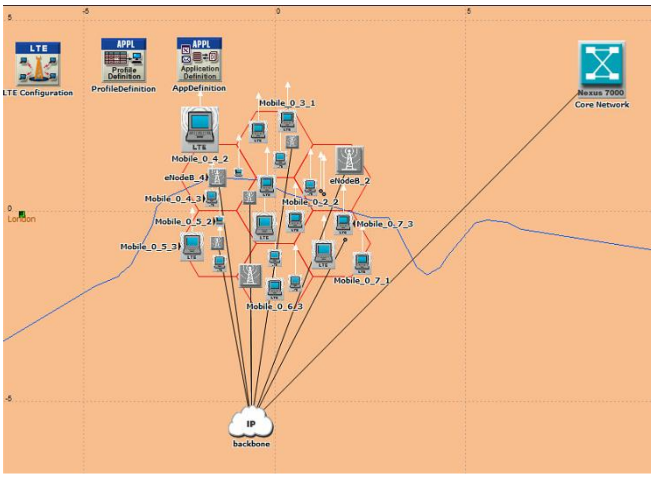


Figure 3:VoLTE opnet scenario

A. *Simulation model*

The topology of simulation is illustrated on figure 3.

Below are the common components between the two architectures:

- 6 eNodeB the type “LTE eNodeB atm, Ethernet” network each one have 3 user equipment that support VoLTE
- An IP-Phone also using SIP as the signaling protocol for both sides.
- All the eNodeB have been connected by using 1000 Base-T links

The eNodeB model is diescubed in the figure 4:

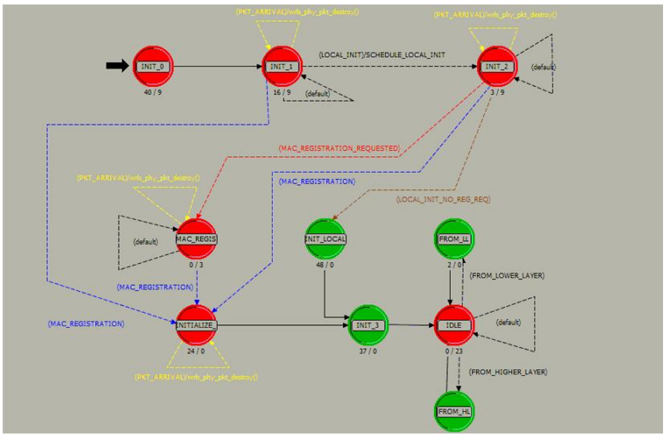


Figure 4: eNodeB model details

The application definition and profiles definition was added to the simulation in order to define used applications, in our case three applications with its profiles VoIP, FTP and Video. For VoIP G.711 is chosen as a codec as chown in the figure 4 below:

Attribute	Value
Silence Length (seconds)	default
Talk Spurt Length (seconds)	default
Symbolic Destination Name	Voice Destination
Encoder Scheme	G.711
Voice Frames per Packet	1
Type of Service	Interactive Voice (6)
RSVP Parameters	None
Traffic Mix (%)	All Discrete
Signaling	None
Compression Delay (seconds)	0.02
Decompression Delay (seconds)	0.02
Conversation Environment	(...)

Figure 5: Voice codec details

For LTE radio parameters is described below and shown in the figure 6:

- DL frequency is 2300 Mhz and UL frequency is 1920Mhz
- Bandwith is 20 Mhz
- Cyclix prefix type is 7 symbols per slot

⊟ UL SC-FDMA Channel Configura...	(...)
Base Frequency (GHz)	1920 MHz
Bandwidth (MHz)	20 MHz
Cyclic Prefix Type	Normal (7 Symbols per Slot)
⊟ DL OFDMA Channel Configuration	(...)
Base Frequency (GHz)	2110 MHz
Bandwidth (MHz)	20 MHz
Cyclic Prefix Type	Normal (7 Symbols per Slot)

Figure 6: LTE radio parameters

V. RESULTS AND DISCUSSION

The obtained measurement is analyzed and discussed to evaluate the performance of VoLTE to sort out if G711 is the best codec as per IUT recommendation. The most affected parameters of QoS in VoLTE are: jitter, MOS, end-to-end delay and traffic (send and receive). The period of simulation is 100 minuts in order to display the results in a clear way. The Figure 6 shows the detail of execution time.

1) Traffic received and sent:

Figure 7 and 8 describes the traffic sent and received in the network in byte/second. The traffic send is equal the received traffic and both of them is start at 2nd minute. The maximum researched value is 230 bytes/second

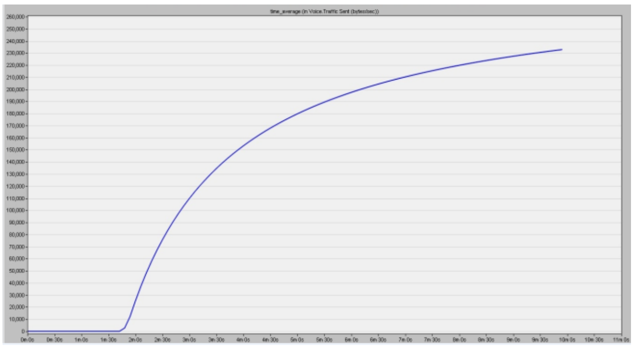


Figure 7: Traffic sent

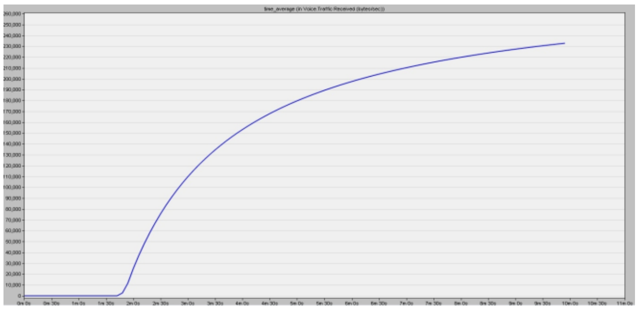


Figure 8: Traffic received

2) End to End Delay

Figure 9 shown the End to End delay. We can note that the value of E2E delay become stable after the second minute time that the traffic become higher. The stable value is 0.12 second and that is very acceptable. 0.12 second is the value of E2E delay obtained in the case of VoIP by using SIP architecture for G.711

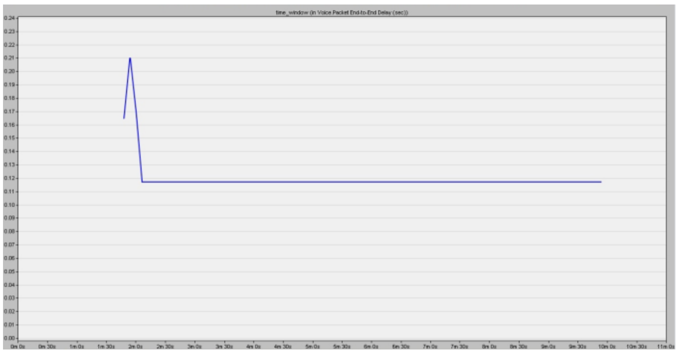


Figure 9: End to End delay

3) Mean opinion Score:

MOS gives a numerical indication of the perceived quality of the media received after being transmitted and eventually compressed using codecs. In our case as shown in Figure 10, The MOS value is between 4 and 4.2 and this is the best value for VoLTE

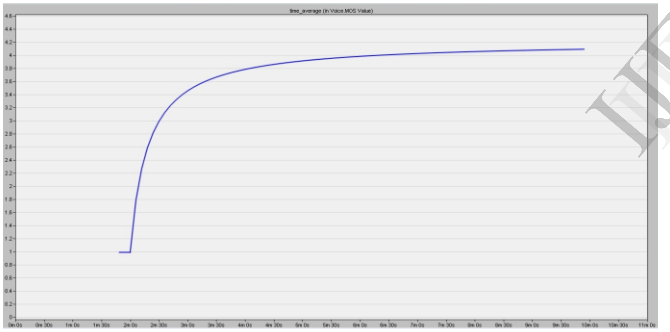


Figure 10: Mean Score Opinion (MOS)

4) Jitter:

Jitter is the time variation between packets arriving caused by network congestion, timing drift, or route changes. In the simulation the value of jitter is very low around 0. Figure 11 shows the variation of jitter

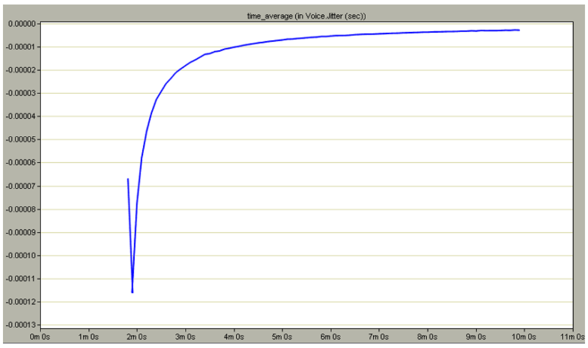


Figure 11: Jitter (second)

CONCLUSION

This paper has analaysing the VoLTE performance when using G.711 as a codec The result shows that the VoLTE performance is with the same level or high compared to VoIP using LAN netwerk because the performance still in the same value and we can add the mobility aspect for VoLTE and the high speed.

This work has shown that the performance of VoLTE is very good when using G.711 as a codec. After sorting out that G.711 respect IUT recommendation for VoLTE, the research team aims to extend this study to analyze and compare the network performance VoLTE (Voice over Long Term Evolution) using G.729 as a codec and also for different possible configuration for LTE.

REFERENCES

[1] 3GPP TS 36.401, Evolved Universal Terrestrial Radio Access (EUTRA) and Evolved Universal Terrestrial Radio Access Network (EUTRAN) Overall description, Stage 2. June 2010.

[2] Emad Aboelela ,NETWORK SIMULATIONS EXPERIMENTS MANUAL,Computer Network,3rd Edition.

[3] S. Brunner and A. Ali, “Voice over IP 101: understanding VoIP networks”, Juniper Networks, white paper, Aug 2004.

[4] John F. Buford, “IP Communication Services”, Special Issue, Journal of Communications, Volume 7, Number 2, February 2012.

[5] R. Dimova, G. Georgiev and Z. Stanchev, “Performance Analysis of QoS Parameters for Voice over IP Applications in a LAN Segment”, International Scientific Conference Computer Science, 2008.

[6] Rossitza Goleva and Maria Goleva, “VOIP traffic shaping analysis in Metropolitan Area Networks”. International Journal Information Technologies and Knowledge, Vol 2, 2008.

[7] Ajith Kumar and Sheela Ganesh Thorenoor, “Analysis of IP Network for different Quality of Service”, 2009 International Symposium on Computing, Communication, and Control (ISCCC 2009) Proc. of CSIT vol.1 (2011) © (2011) IACSIT Press, Singapore.

[8] Younes Labyad, Mohammed MOUGHIT and Abdelkrim HAQIQ, “Performance Analysis and Comparative Study of Voice over IP using Hybrid Codec”, Complex Systems (ICCS), 2012 International Conference,Agadir

[9] S. P. Mehta, “Comparative Study of Techniques to minimize packet loss in VoIP,” 2005. [Online]. Available: http://www.rh.edu/~rhb/cs_seminar_2005/SessionB3/mehta.pdf.

[10] J. Rosenberg et al., “SIP: Session Initiation Protocol”, Internet Engineering Task Force (IETF), RFC 3261, June 2002.

[11] Willis, Peter, “Core IP and MPLS Networks,” IEE Annual Course on Telecoms Networks - The Next Generations, London, United Kingdom, pp. 15-33, July 2005.

[12] Ian F. Akyildiz ∗, David M. Gutierrez-Estevez, Elias Chavarria Reyes, “The evolution to 4G cellular systems: LTE-Advanced”, ELSIVER, Physical Communication 3 (2010)