# QoS Voice and Video over IP at LTE Network (Review Paper)

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## ABSTRACT

The objectives of this paper is to analyze voice and video over IP over LTE network from previous studies overview to be better than QoS to voice and video over IP over LTE network by using description and mathematics model only. After achieving the methodology use description and a previous study found that the LTE characteristics give high QoS in all parameters of QoS.

*Keywords— Voice over Internet Protocol; Quality of Service; Orthogonal Frequency Division Multiplexing Access; Single carrier FDMA*

## I. INTRODUCTION

The study was initiated in November 2004 by 3GPP Evolution, Terrestrial Radio Access Network (UTRAN), which was the foundation for UMTS/WCDMA systems., which wanted to create a new system to achieve higher data transfer rate, low-latency, packet optimized radio access technology, increase density and try to overcome the interference problems that existed in previous generations. And then began its final planning in 2008 and began to work at the beginning of 2012 in some countries of the world.[1]

Orthogonal Frequency Division Multiplexing Access (OFDMA).  It is one of the main components and it is used as a signal carrier and associated access schemes and uses a standard IEEE (802.16). OFDMA is used in a number of other systems from WLAN, WiMAX to broadcast technologies including DVB and DAB.OFDMA has the flexibility to diffuse across different frequency bands and average interactions with adjacent cells. One frequency network can be covered and provide excellent coverage and can give frequency diversity by deploying carriers throughout the user spectrum and allows per-channels or per-sub channels power [3 ,2] .

**Single Carrier FDMA (SC-FDMA):** Is a single carrier multiples access technique which has similar structure and performance to OFDMA.  It utilizes single carrier modulation and orthogonal frequency multiplexing DFT-Spreading in the transmitter and frequency domain equalization in the receiver.  In view of its advantages, the use of ODFMA and the associated access technologies, OFDMA and SC-FDMA are natural choices for the new LTE cellular standard.

In LTE we use OFDMA for downlink transmission and SC-FDMA for uplink transmission because OFDMA is high peak-to-average power ratio (PAPR), which raises the cost and lower the power efficiency of a transmitter's power amplifier. With a lower PAPR, the power amplifiers at mobile terminals employing SC-FDMA can be simpler and more powerful and efficient than they would be with OFDMA transmission. On the other hand, with its high signalling rate, the frequency domain equalizer of an SC-FDMA link is far more complicated than an OFDMA equalizer. With SC-FDMA transmission confined to the LTE uplink, the complicated equalizers are required only at base stations and not at mobile terminals [1] .

The objective of this paper is to analyze and explain voice and video over IP over LTE network from previous studies overview to be better than QoS to voice and video over IP over LTE.

The challenge for LTE was to achieve high data rate, reduce delay and increase network users. And we succeeded in that. LTE achieve peak data rate minimum 100mbps in downlink and minimum 50mbps in uplink, up to 200 active users in cell, less than 5ms user-plane latency, in mobility support up to 500 km/h, enhanced multimedia broadcast and multicast service (E-MBMS), Spectrum flexibility 1.25MHZ —20MHZ and enhanced support for end-to-end QoS. Multi-Input-Multi-Output (MIMO) multi-carrier channels are dependent resource

scheduling. Frequency domain equalization and fractional frequency reuse.

Table 1: LTE specific.

| PRAMETER | LTE |
|---|---|
| Channel bandwidth | 1.4,3,5,10,15,and 20 |
| Downlink | OFDMA |
| Uplink | SC-FDMA |
| Duplexing | FDD and TDD |
| Subcarrier mapping | Localized |
| Subcarrier hopping | Yes |
| Data modulation | QPSK,16-QAM.46-QAM |
| Subcarrier spacing | 15KHZ |
| Channel coding | Coding and turbo coding |

## II. PREVIOUS STUDY

Soniaforconi, Alessandro Vizzarri, they conducted a study entitled Review of Studies on End-to-End QoS in LTE Networks at the University of Rome Tor Vergas The aim of the study was to examine the end-to-end quality of service and to take advantage of wireless network features, drawing on previous studies. And the most important features of LTE and its architecture. The first part of the paper contained the quality of service and its ability to differentiate between user services. They also searched for the wireless standard that preceded LTE WIMAX. The second part of the paper focused on quality of service in E-UTRAN (Evolved Universal Terrestrial Radio Access Network) for LTE network [10] .

Sohaibbhat, Amayi kakakati, shalinisah. The aim of the study was to clarify the sound performance of LTE using different voice codec's. And the ability of LTE to reach the highest level and high productivity in both downlink and uplink to ensure the quality of service experienced by the user. OPNET was used to simulate a system that uses multiple audio encoding and its effect on LTE. The results were shown that using strong sound coding can improve the MOS values, good readings in the End-to-End delay [11].

Alessandro Vizzari. conducted a study entitled Quality of Experience Assessment for VoLTE Services at the University of Rome Tor Vergata. He has published a service quality paper for VoIP over LTE that LTE is the first 3GPP wireless network to be fully based on the Internet protocol and the high accessibility it has. And how to manage quality. The OPNET simulator was used to simulate voice transmission over the Internet based on the voice quality of service parameters for MOS, voice traffic send and received, packet loss, based on the uplink and downlink in LTE. [12].

Salauddin Ansari, Rachna Gupta.They conducted a study entitled Evaluate Performance of Voice over LTE Networks using Voice Codec's Department of CSE & NGF College of Engineering & Technology Palwal, Haryana, India. They published a paper on voice over IP over LTE networks and searches about the need for service quality subscribers and the reputation that VOIP has enjoyed. Several companies have made use of the infrastructure that can provide VOIP support and ways to get good quality of services at a lower cost. And the benefits of LTE in the continuity of the call and the ability to support the LTE network for previous generations networks and broadband capabilities [13].

Ahmed J Jameel, Maryam Mshafiei, they conducted a QoS Performance Evaluation of Voice over LTE Network in Ahlia University, Manama, Bahrain. In this paper, they explained the previous generations and the great transformation of each generation and the next generation and the transformation were not easy as it took a lot of work, effort and time. The LTE must contain the best design for a circuit of the mobility is a packet switched. There is a significant increase in data traffic and the wide capacity of the LTE network and the replacement of voice transmission through old and modern technologies [14].

From the previous studies and what the researchers have shown from the studies and results, we find that LTE is a real addition to the idea of transferring voice and video through mobile networks and provide what people need from the quality of service.

## III. LTE ARCHITECTURE

The old system cannot be replaced with the new system because it will cause large losses for the companies that use the systems or who wish to move to the new system so, the new system is planted to comply with the old systems and replace it step by step. The let architecture has two important part: Evolved UMTS terrestrial radio access network (E-UTRAN). Evolved packet core (EPC) from a protocol stack perspective. Figure 1 show LTE architecture
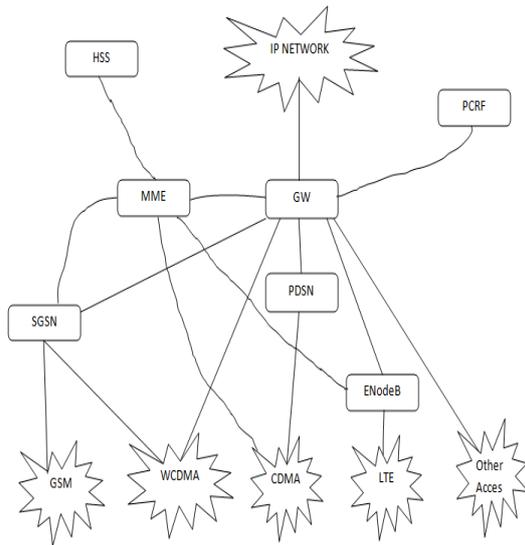
Figure 1: LTE Architecture

Evolved Node B (eNodeB): The function of eNodeB provides radio resource management function and all radio interfaces – related function. Mobility management entity (MME): Manages mobility, user identity and security parameters. Serving gateway: Node that terminates the interfaces between another network, responsible for routing and for wording and provides mobility.

Packet data network gateway (PDN-GW): Its responsibility is providing internet connectivity, QoS and mobility between LTE and non LTE.

### IV. VOICE AND VIDEO OVER IP

Voice and video over IP are a digital service of new multimedia services that rely on Internet protocol that has been certified by manufacturers for use in wireless networks. It has become one of the most popular and popular services, replacing communications because of the low cost, easy and speed of communication and access. It is no longer limited to calls only, but went beyond that in the radio and television services, conferences and meetings and live broadcast services and also are used in security cameras. Since the old roads are in the same direction and the new method depends on the current in both directions, which indicate the low cost and ensure access to the quality of service required. Protocols for voice and video over IP which are H.323 and SIP:

H.323: It is a standard for a family of protocols used in calls and multimedia communications. Of the H.225 family used for call control and the H.245 protocol used in call handling. A simplified example of using the H.323 standard for a connection is that starting with an H.225 exchange to another point and use this exchange with the signalling protocol Q.931 and then start the call management phase that you control as described earlier using the H.245 protocol [4].

SESSION INITIATION PROTOCOL (SIP) It is a multimedia signalling protocol that used multimedia session establishment, modification and termination. SIP incorporates elements of two important Internet protocols:

HTTP (client/server design, URL, response codes) and SMTP (text-encoding scheme, header style), SIP can be used with other IETF protocols for a complete multimedia architecture. Such as RTP /RTCP protocols that transmit real-time applications and provide quality of service. Internet protocols, control information and signals to be exchanged between network entities. These are complex facts because the Internet is shared across different types of devices and different types of networks, Hence the need for a Session Initiation Protocol [6 ,5] .

QOS PARAMETERS OF VOICE AND VIDEO. The networks that use the data have shown significant disadvantages of public switched telephone (PSTN) and a variety of quality parameters and the quality of the Internet communication service and can be mentioned in jitter, delay, packet loss, MOS.[7] .

MOS: The mean opinion score is a subjective measure used to measure sound quality and it is between 1 to 5. (5 = excellent, 4 = good, 3 =fair, 2 = poor and 1 = bad).
Jitter: It is the difference in the arrival time of successive packets. Jitter is calculated over an interval of time [8].
Delay: It is the delay in packet access and measured by the delay in time calculation between the sender and receiver.[9]

## V. CONCLUSIONS

From the previous studies, we found that the throughput values in the uplink between 140mpbs-160mpbs, the end-to-end delay between 0.11-0.17 and the MOS values between 3.9-4.35 and so LTE achieved high service quality in all QoS parameters for voice and video over IP.

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