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Evaluate Performance of Voice over LTE Networks using Voice Codec's

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***Abstract*— Long Term Evolution (LTE) is a 3GPP standard for wireless transmission systems. It is also the first 3GPP wireless standard which is fully IP-based. Because of its probability to arrive very high throughput (for example 100 Mbps in downlink), an effective end-to end QoS treatment is required for guaranteeing a good QoS detected by end subscriber (QoE). Voice service provided by LTE systems is Voice over IP (VoIP) facility, with no QoS-aware technique. LTE offers a native and unique QoS-aware technique for end-to-end facility providing based on QCI and EPS bearer. However effective end-to-end QoS management of VOLTE should deal several aspects, this paper examines the voice codec effects on end-to-end VoLTE performance. Several voice codec's are taken in various scenarios modeled utilizing OPNET simulator software tool. Final comparison between them is offered.**

Keywords: LTE; VoLTE, VoIP; LTE network performance, end-to-end QoS; LTE KPIs; OPNET;

I. INTRODUCTION

Subscribers get pleasure of VOIP services in developed countries because of the existence of technologies i.e. 3G, 4G and LTE. VOIP is the main issue in developing countries because of the absence of these technologies as several main telecommunication companies are stressing on combining their system with VOIP technologies for providing the enjoyment of these less expensive services to the users. VOIP services have become very famous among medium and small organizations mainly for business process outsourcing companies and call centres, which uses data and voice services to a great extent. Several companies are attempting to make use of infrastructure which can provide the support to VOIP services in a manner such that they can have good quality of services with lesser recurring costs. Call centres needs Quality of Voice (QOV) and Quality of Service (QOS), as call centres provide services i.e. sales, customer support and data collection but the quality can endure because of varying computational needs, unsuitable codec scheme, traffic load and resource availability, , the wrong selection of service suppliers and the unsuitable infrastructure. A call center utilizes VOIP for having best economic standing but many call center wants to save recurring costs and initial investment too much and end up making very unsuitable decisions with respect to selection of infrastructure and integration of VOIP service and PSTN.

VOIP cannot be as good as PSTN, but a call center should be smart enough to make the right selections to have a better financial standing. Codec performance is relied on the conditions as every codec has particular needs. If the suitable environment is not supplied the codec will have lower performance and the result shows a great amount of packet loss and jitter. The call managing service should maintain the change codec depend on traffic and channel situations, based on call destination information, with respect to a policy for selection of codec. VOIP service supplier and developers of call managing software require information on what is a suitable policy for selection of codec. Several small industries fail in the very first year of deployment due to many reasons. some reasons are:

- Absence of financial and deployment plan.
- Deficiency of information related to the entities and governing bodies in the area.
- Unsuitable network architecture.

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- Inappropriate selection of service provider according to the requirements of the business.
- Selection of unsuitable Codec and tools.
- Unsuitable handling of IP pooling and Call Manager.

II. VOICE OVER LTE

Voice functionality is supplied by the IP Multimedia Subsystem (IMS) in this solution. IMS is a central network architecture that is incorporated on top of the LTE network as depicted in Figure 1.1. The IMS network is primarily utilized to supply all the basic voice facilities that are supplied by the subsisting CS networks. It also facilitates improved multimedia services i.e. real time gaming, video conference etc.

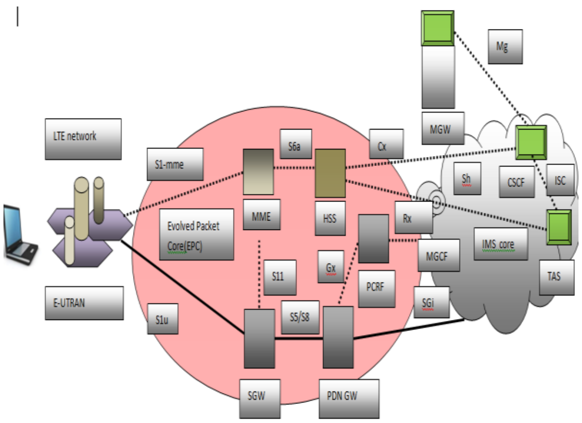

Figure 1: VoLTE

The major benefit of utilizing an IMS based solution is that it uses the LTE architecture completely instead of using the existing CS networks for providing voice service. The IMS network can be integrated with the legacy 2G/3G networks then it can provide support to voice call continuity even when the user moves out of LTE range. Thus, the user can uses the same facilities in roaming also. This solution is being designated as the long term solution as it can provide improved features to the LTE network and also supports combination with the available 2G/3G networks.

Because of these significant characteristics LTE is able to offer good performances, particularly for those bandwidth consuming facilities i.e. data downloading or video streaming services. On the other side voice service (VoIP over LTE) also remains one of the most significant facilities provided by LTE network.

LTE is first 3GPP fully IP-based wireless standard. It means each end-to-end link between LTE network and end subscriber IP protocol as communication transport protocol. For this cause Circuit Switching (CS) data routes are not available in LTE system but only Packet Switching (PS) data routes.

For guaranteeing least Quality of Service (QoS) needs for its service provided to the end subscriber and then to enhance his perception, LTE standard offers a native and unique QoS-aware technique for end-to-end facility delivering.

A. Reference architecture

LTE reference architecture is divided into two important parts:

• Evolved UMTS Terrestrial Radio Access Network (E-UTRAN)

• Evolved Packet Core (EPC)

From a protocol stack perspective, LTE depends on data plane (DP) and user plane (UP) same as others 3GPP wireless standard. Fig. 1 shows LTE reference architecture. [2]

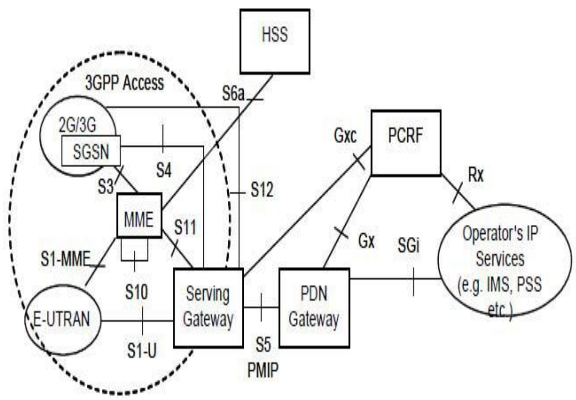

Figure 2: LTE network architecture

III. LITERATURE REVIEW

Improving, optimizing and examining voice traffic over data networks have been an important issue to developers and researchers, various schemes have been suggested depending on analyses from simulated traffic and real word.

S.Alshomrani et al (2012) had investigated yhe performance of VOIP traffic over WIMAX networks. The effect of various voice codec schemes and statistical distribution of VOIP over WIMAX has been discussed in this paper. The study provides an insight into VOIP performance in WIMAX networks using parameters such as delay, jitter, packet loss and MOS.

U R ALO et al.(2013) has used simulation techniques to analyse the performance of VOIP over wireless LAN for an increased number of VOIP calls. This paper shows results of simulation experiments using appropriate graphs and tables and measures factors such as wireless LAN clients, encoder scheme and use of high number of clients which shows a great effect on performance of VOIP over wireless LAN.

Nafisa Bashrik et al.(2014) In this paper the author has focussed on UMTS network based on wideband CDMA technology – a third generation telecommunication system that contains improved performance and quality of service that contribute to the development of human communication. The author has compared both UMTS and WIMAX using OPNET simulator tool. The author has analysed various important critical parameters such as MOS,end to end delay,jitter and packet delay variation and has concluded by saying that WIMAX is a better technology to support VOIP applications compared to UMTS.

Ben-Jye Chang et al (2008) had suggested an algorithm (a cross-layer-based adaptive vertical handoff algorithm) with predictive Radio Signal Strength (RSS) to decrease the unneeded handoff though importantly improving usage and minimizing connection dropping in distributed network. Vertical handoff was a significant process for obtaining continuous smooth transmissions in this distributed network. Most of previous works had utilized the RSS-based mechanism to find out handoff thresholds, which results a important ping-pong effect that increases unneeded handoff. Though combining the RSS-based technique with a hysteresis method decreases the unneeded handoff, it was endured from low usage and high dropping.

Enrique Stevens-Navarro et al (2008) had suggested a decision algorithm for the vertical handoff selection phase in distributed wireless networks. This decision algorithm was depended on the MDP formulation with the aim of increasing the required total reward of a connection. For simulating the QoS of the mobile connection, a link reward function was utilized.

In (Flizikowski, Majewski, & Przybyszewski, 2010), the authors have examined to a greater extent the data, video and audio support services in WiMAX Networks. Their research work was stressed on analysing the QoS (Quality of Service) deployment over WiMAX Networks and compared the performance obtained by utilizing WiMAX service classes i.e. Extended real time Polling Service (ertPS), Unsolicited Grant Service (UGS). The study which is done out by these researchers have affirmed that WiMAX Networks provide support to the real-time application more in comparison of other wireless access techniques i.e. 3G and WLAN.

In 2008, (Ansari & Haghani), suggested a traffic-aware scheduling algorithm for the deployment of VoIP applications over WiMAX Networks. The authors severely analysed the performance of the suggested algorithm in comparison of several notable established methods. They further elaborated how the efficiency of VoIP over WiMAX networks can be enhanced by the utilization of their suggested scheduling techniques. But their suggested algorithm was not examined in terms of available performance metrics to determine and demonstrate its robustness in QoS supports.

Rong et al. (2007) gave an combined adaptive power allocation (APA) - call admission control (CAC) downlink resource management framework for OFDM-TDD based network by considering the subscriber and service supplier.

Niyato and Hossain (2007) suggested a determining model in an integrated WIMAX / Wi-Fi network for adaptive bandwidth sharing. Game theory has been utilized to examine and determine the bandwidth sharing between Wi-Fi access point routers and a WiMAX base station.

Hung-Yu et al. (2005) talked about the interference challenges and suggested a better method for usage of WiMAX mesh by designing of scheduling algorithm scheme and multi-hop routing. This method takes into account both interference conditions and traffic load demand. Simulation results depicts that the suggested methods efficiently enhanced the performance of network throughput in IEEE 802.16 mesh networks and obtained high spectral usage.

A routing algorithm and a scheduling mechanism is proposed by Liqun *et al. in* 2005 to improve the spatial reutilization in wireless mesh networks and to obtain better spectral efficiency and network throughput. This model needs the recipient to be interference free and take the interference range same as the communication range.

The authors in (Shrivastava & Vannithamby, 2009) proposes that though WiMAX Networks are effective in supporting data traffic, but the capacity of VoIP when utilized over IEEE 802.16e WiMAX system is not so much effective In their work, the authors utilized recurring scheduling as a technique in IEEE 802.16e WiMAX system for reducing occurrence of MAP overhead. The only deficiency in their suggested persistent/group scheduling technique is that it produces “resource hole” in the frame at the data distribution area which causes to ineffective resource distribution. Most of the VoIP QoS investigations have been carried out on Wireless LAN, Ethernet LAN in comparison of WiMAX access networks. Most of the time when it has been done with the WiMAX networks, the investigators have failed to see at some famous complicated codec schemes/algorithms with decreased value of voice frame size per packet that can be employed on video /voice calls/conference to improve the quality of VoIP performance when utilized over WiMAX Networks. These complicated codec techniques supplies wonderful compression efficiency that preserves network bandwidth necessarily in wireless technologies i.e. WiMAX networks. Furthermore, most of the investigators that did work on research of VoIP system performance when utilized over WiMAX networks employed other network modellers i.e. E-modeling, NS-2, NetSim, etc for modelling, designing and simulation of the network.

This research work will employ OPNET Network modeller which is quite universal among IT students and communication professional to see at the some codec algorithms as well as performance standards approved and defined by ITU-T for video and voice conferencing utilizing VoIP over IEEE 802.16e standard WiMAX networks in a campus network.

IV. VOLTE QOS ASSESSMENT

An effective QoS assessment of VoLTE facility is a necessary item for LTE networks operators for various causes. [6]
In wireless systems QoS detected by end subscribers, also known as Quality of Experience (QoE), is necessary. However, in situation of a VoIP service i.e. VOLTE QoE is strictly associated to the quality of speech, QoS-oriented mechanisms are to be carried out for reducing latencies or transmission delays.
Another view is that LTE network subscriber end-to-end IP link to/from UE: an end-to-end mechanism to delay reduction is suggested. [7] [8]
At last, VoLTE is service voice for LTE networks. It is featured by QoS needs quite different from other facilities integrated with, like FTP downloading or HTTP web browsing.
An effective QoS assessment for VoLTE is made in three important areas to be examined and maintained:

- Voice codec
- LTE QoS characteristics: QCI & Bearer
- IP network routing improvement (DSCP mapping)
- Network impairments (network faults or congestions)

Coding and decoding mechanisms are very significant for preparing a good digital signal to be received or transmitted. In situation of LTE these mechanisms should be combined with its QoS management characteristics conducted at network level (QCI and EPS bearer), DSCP mapping criteria and Type of Service (ToS), network congestion avoidance. [9]
This paper is only concentrated on the first area: voice codec. However for 3GPP in situation of VoLTE it is essential to utilize minimum voice codec of Adaptive Multi-Rate Narrow Band (AMR-NB) family, this work builds analysis of VOLTE Key Performance Indicators (KPIs) in several scenarios with various voice codec.
MOS parameter is the most significant KPI utilized to measure VoIP service QoS (VoLTE involved). [9]

TABLE I VOICE CODEC CONSIDERED FOR SIMULATION

Simulation scenarios			
Voice Codec	*Sampling frequency [KHz]*	*Transmission rate [kbit/s]*	*Target MOS*
G.711	8	64	> 4.1
GSM EFR	8	12.2	4.3
AMR 12.2 k	8	12.2	4.3
IS 641	8	7.4	4
G.729A	8	8	3.92

For measuring the performances of end-to-end VoLTE in this work any network transport elements are maintained (for instance, SIP signaling, DSCP and other QoS IP-based characteristics, etc) but only voice codecs. [11] [12]
However, an effective end-to-end mechanism to quality of service requires to examine both quality of content provided (voice in situation of VOLTE) and network performances, adopting main network KPIs are inquired together with MOS:

- End-to-end packet delay
- Voice traffic received (packets/seconds)
- Voice traffic sent (packets/seconds)
- LTE downlink delay (seconds)
- Voice packet delay variation
- LTE uplink delay (seconds)

V. SIMULATIONS

Simulation of VoLTE services is performed utilizing OPNET Simulator 17.5 PL6 software tool, depending on Discrete Event Simulation (DES) method. However, aim of this work is to examine only effect of several voice coded on end-to-end performance of VoLTE, Type of Service (ToS) assumed is only Best Effort (BE).

A. Scenarios

Five different VoLTE scenarios are assumed. Each utilizes a different voice codec: GSM EFR, G.711, IS 641, AMR 12.2K, G.729A. Table II shows relationship between scenario and voice codec utilized during simulation. [13]

TABLE II. SIMULATION SCENARIOS

Simulation scenarios			
Id	*Voice Service*	*Voice Codec*	*Type of Service (ToS)*
1	PCM Quality Speech	G.711	Best Effort (BE)
2	GSM Quality Speech	GSM EFR	Best Effort (BE)
3	GSM Quality Speech	AMR 12.2k	Best Effort (BE)
4	GSM Quality Speech	IS 641	Best Effort (BE)
5	IP Telephony	G.729A	Best Effort (BE)

B. OPNET Settings

1) Network Topology

Fig 3 represents configuration of simulated LTE network in a general campus region 50 x 50 Km.

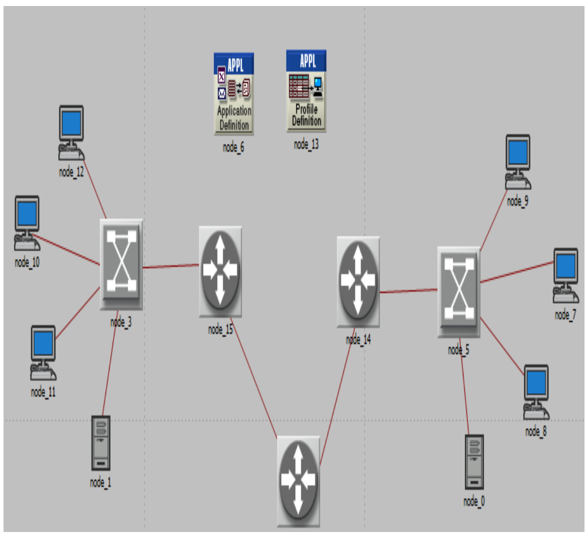

Figure 3: simulated LTE network topology

2) LTE settings

Whole LTE network is simulated by following parameters shown in Table III.

3) Application configuration

In OPNET simulator, various applications are previously defined and proper. In this paper voice application is chosen. A novel application is generated and named *Voice*. Voice application is established with a start offset of 40 seconds till the simulation period end. The same application is utilized for all scenarios but every scenario utilizes a different voice codec.

TABLE III LTE NETWORK SETTINGS

LTE Network Settings		
Network Node	*Parameter Description*	*Parameter Value(s)*
User Equipment	Antenna gain	- 1 dBi
	Modulation and coding scheme	9
	Multipath Channel mode (Downlink)	LTE OFDMA ITU Pedestrian B
	Multipath Channel mode (Downlink)	LTE SC-OFDM ITU Pedestrian B
	Pathloss	Free space
	Receiver sensitivity	-200 dBm
eNodeB	Sectors	3
	LTE bandwidths	10 MHz
	Duplex mode	FDD
	eNodeB antenna gain	15 dBi
	Receive antennas	2
	Transmit antennas	2
	Operating power	20
	Receiver sensitivity	-200 dBm
EPC	DRX for idle mode	256
	N3 bufer size	8192 bytes
Link (PPP DS3)	Traffic load	default

4) Profile configuration

A unique profile is generated. It is known as *Voice Profile*. Main profile settings are shown in the following table. In table IV Voice Profile settings are defined.

TABLE IV VOICE PROFILE SETTINGS

Voice Profile settings	
Attribute	*Value*
Profile name	Voice Profile
Application name	Voice
Start time Offset (seconds)	Constant (40)
Duration (seconds)	End of profile
Inter-repetition	Exponential (300)
Number of repetitions	Unlimited
Repetition Pattern	Serial
Operation Mode	Simultaneous
Start Time (seconds)	Constant (40)
Duration (seconds)	End of simulation
Inter-repetition	Constant (300)
Number of repetitions	Constant (60)
Repetition Pattern	Serial

Voice Profile utilizes a unique application, *voice* application. Voice Profile utilizes also a start offset of 40 seconds. So there are two different start offset of 40 seconds: the first one is associated to Voice application configuration, the second one is associated

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Volume 5, Issue 5, May 2016

to Voice profile configuration. It means that packets are going to be forwarded after 80 seconds from starting of simulation.

In situation of VoLTE simulation, in OPNET simulator a gold bearer is selected. It offers a 96 Kbps link bit rate both in uplink and downlink.

5) Statistics

DES statistics chosen for VoLTE simulation are:

• *Global statistics:*

• *Node statistics:*

In VoLTE simulation the following statistics are chosen:

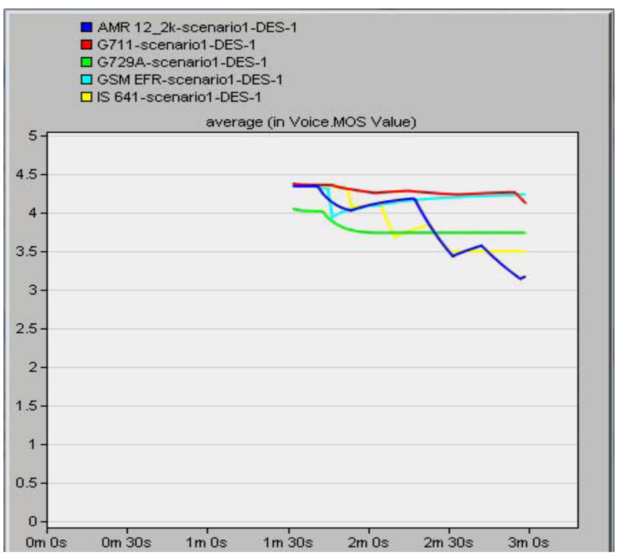
• *Global statistics* (simulation results are offered at whole network level)*:* IP, LTE, Voice

• ***Node statistics*** (simulation results are offered at network single node)*: LTE*, IP, UDP, LTE PHY, Voice

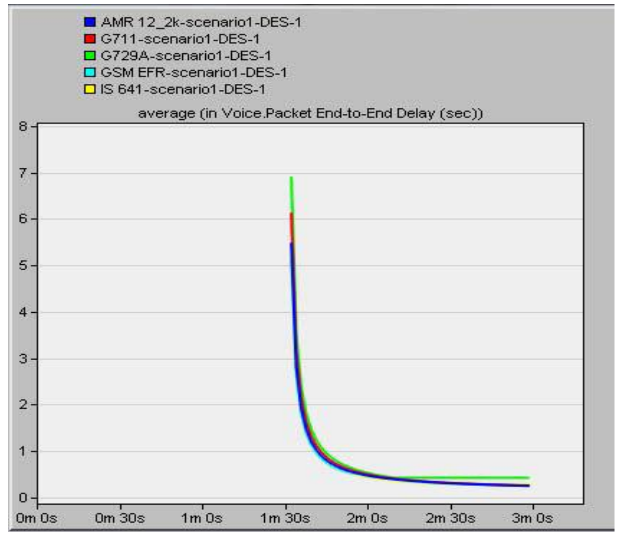
C. Simulation results

In this paragraph main results of simulation depending on KPIs mentioned in paragraph V are talked about.

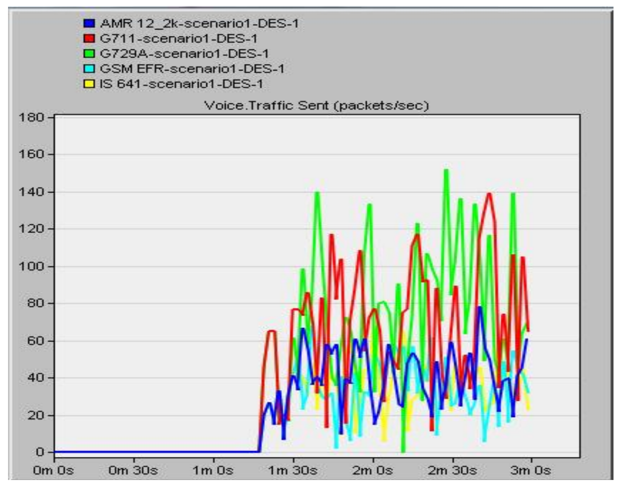
Fig 5 represents for every scenario their graphic representation with respect to MOS (a), end-to-end packet delay (b), voice traffic sent (c), voice traffic received (d), voice packet delay variation (e), LTE downlink delay (f), LTE uplink delay (g) .



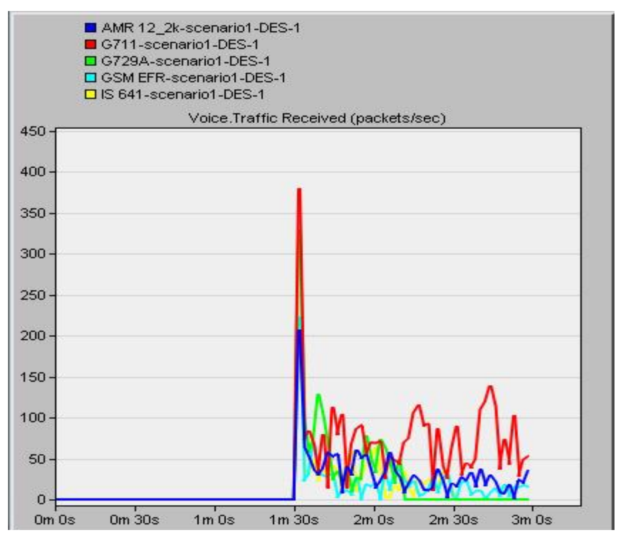
(a)



(b)



(c)



(d)

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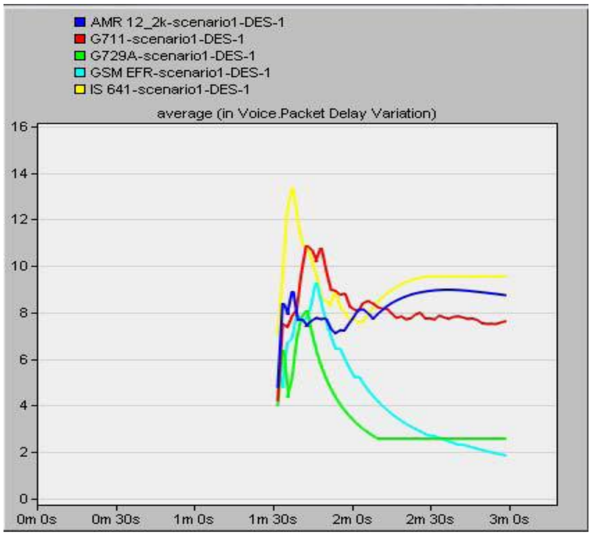

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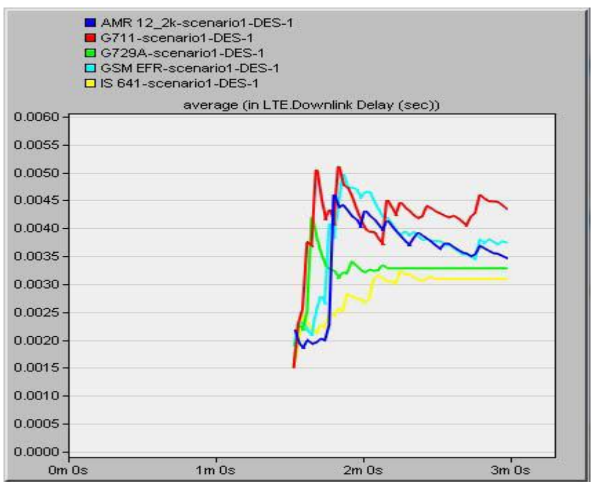

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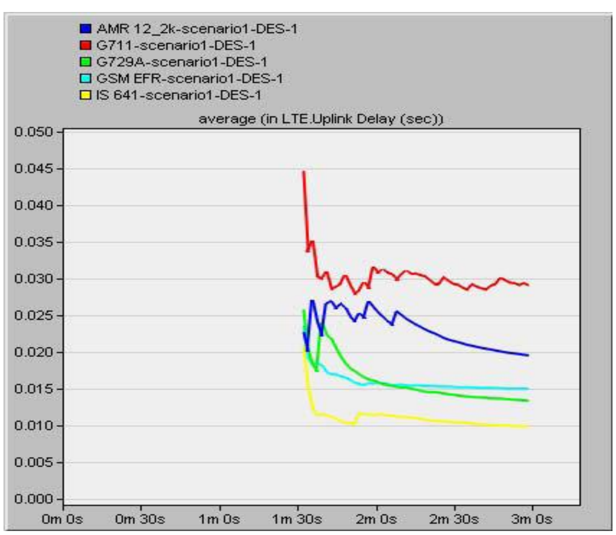

(g)

Figure 5: Graphic representation of simulation results.

G.711 and GSM EFR codec can ensure better performance with respect to MOS, while AMR 12.2k only with respect to end-to-end packet delay.
IS 641codec offers a good performance with respect to LTE delay both in uplink and in downlink.
G.729A and GSM G711 offer a good performance with respect to forwarded/obtained voice traffic, while GSM EFR and G.729A with respect to packet delay variation. G.729A offers minimum value of end-to-end packet delay. Related to MOS performances, table IX compare evaluated MOS values with relative target values for every scenario.

VI. CONCLUSIONS

Aim of this work is to measure end-to-end QoS of VoLTE concentrating on effects of voice codec. Various voice codec's are taken and employed in five different scenarios. Any network transmission element is taken. An effective analysis of end-to-end QoS KPIs is shown. It depends on end-to-end delay, MOS, voice traffic received and sent, voice packet delay variation, LTE uplink /downlink delay. From a voice codec perspective, analysis of MOS values evaluated in each scenario shows better performance reached by GSM EFR and G.711 codec's. Other voice codec's (IS 641, AMR 12.2k, G.729A) show a very discontinuous nature because they are more influenced by network transmission elements i.e. transmission delays. GSM EFR is related to AMR-NB codec family and it is essential to utilize in LTE.
Future works are going to inquire uses of LTE QoS characteristics and effective network management mechanisms for enhancing VoLTE end-to-end QoS perception by end subscriber.

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