An End-to-End QoS Performance Evaluation of VoLTE in 4G E-UTRAN-based Wireless Networks

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***Abstract*— Long Term Evolution (LTE) is the Fourth-Generation (4G) mobile broadband technology. Its standardization has been finalized by Third-Generation Partnership Project (3GPP) in Release 8 technical specifications (R8). As users' demand for higher data rate continues to rise, LTE and its ability to cost effectively provide fast, highly responsive mobile data services, a scalable bandwidth and a reduced latency will become ever more important. However, the Evolved Packet Core (EPC) of the Evolved Universal Terrestrial Radio Access Network (E-UTRAN) based wireless networks (LTE and LTE-Advanced) is all-IP Packet-Switched (PS) core network and lacks native support for Circuit-Switched (CS) services. This introduces the problem of how to provide voice services in these networks. The controversy around many of the proposed solutions to provide a PS voice and the effects of this step on the deployment of LTE networks is presented. This paper also deals with the Quality of Service (QoS) in a Voice over LTE (VoLTE) service. It provides a comprehensive evaluation and validation of VoLTE QoS based on the International Telecommunication Union standard Recommendations (ITU-R) and 3GPP standard technical specifications. The initial results obtained give clear evidence that the VoLTE service fulfills the ITU-R and 3GPP standard requirements in terms of end-to-end delay, jitter and packet loss rate. Furthermore, the results related to implementing different LTE bandwidths clearly reflect how these bandwidths affect the overall network performance and end-user experience.**

Keywords- VoLTE; E-UTRAN; LTE; QoS; IMS.

I. INTRODUCTION

The Third-Generation Partnership Project (3GPP) has developed a new technology called Long Term Evolution (LTE) in Release 8 (R8) Technical specification [1]. 3GPP LTE aims to improve the Third-Generation (3G) Universal Mobile Telecommunication System (UMTS) technology to meet the International Mobile Telecommunications Advanced (IMT-A) requirements determined by the International Telecommunication Union (ITU) [2]. Some of the agreed features of LTE are a significant increase in data rates with up to 300 Mbps downlink (DL) and 75 Mbps uplink (UL); a scalable bandwidth of 1.4, 3, 5, 10, 15 and 20 MHz and a reduced latency [3]. However, the changes in this design were significant, with a flat all-IP Evolved Packet Core network (EPC) only supporting Packet-Switched (PS) services [1]. The EPC lacks native Circuit-Switched (CS) services support, including voice, which is considered as the main revenue for Mobile Service Providers (MSPs). This is different from most of the legacy UTRAN/GERAN wireless networks such as UMTS, which support both CS and PS services [2]. A user always expects voice as a basic service provided by the network operator and this raises the question of how to provide a voice call service to LTE users. The Evolved Universal Terrestrial Radio Access Network (E-UTRAN) is the radio wireless access for LTE and LTE-A and its architecture is simpler and flatter than Radio Access Network (RAN) in the 3G mobile networks as shown in Fig. 1. 3GPP presented the technical specifications for the E-UTRAN based networks (LTE and LTE-Advanced) in Release 8, 9 and 10. According to the 3GPP specifications, there is no guarantee that LTE has the ability to fulfil the ITU-R and 3GPP technical requirements related to QoS, especially with one way VoLTE end-to-end delay of less than 150 ms and a minimum of 98% packets successful delivery rate [4]. In this work, firstly we introduce a brief comparison between the proposed solutions for deploying voice service over LTE wireless networks. We also evaluated the VoLTE QoS performance in terms of end-to-end delay, jitter and packet loss rate. For this purpose, we designed a realistic baseline simulation for LTE wireless networks based on 3GPP R8 technical specifications, including IMS. We then simulated different LTE bandwidths and depending on the results we investigated the effects of deploying these bandwidths on the service quality and end-user experience.

The rest of this paper is organized as follows. Section II briefly introduces a description of the VoLTE proposed technologies. Section III explains the QoS architecture in LTE wireless networks. Section IV describes the designed simulation environment. Section V discusses and analyses the simulation results and finally, we conclude this work in Section VI.

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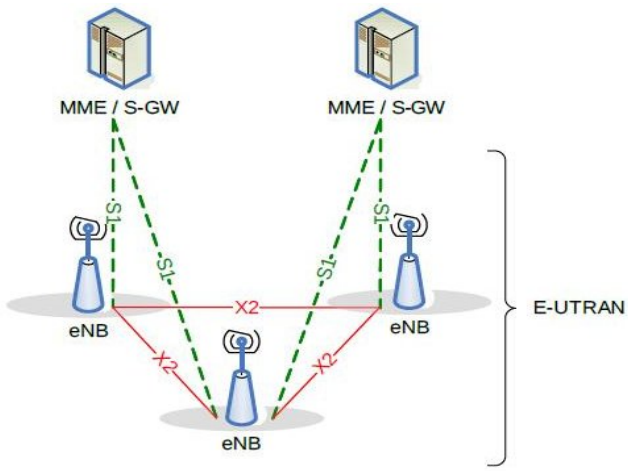

Figure 1. E-UTRAN System Architecture [5]

II. Deployment Of VoLTE Over 3GPP 4G E-UTRAN-Based Networks

This paper discusses mainly Voice over LTE (VoLTE) technology based on IP Multimedia Subsystem/ Multi Media Telephony (IMS/MMTel), which is standardized by 3GPP to provide voice service to LTE wireless networks. Other proposed technologies such as Circuit Switched Fall Back (CSFB), Voice over LTE via Generic Access (VoLGA) and Over The Top (OTT) were investigated and described briefly in this paper.

A. *VoLTE via IMS /MMTeL*

Voice is a fundamental service to consider in any Next Generation Mobile Networks (NGMNs). In fact, IMS with MMTel are the key to make this possible and provide a required High Definition (HD) telephony system to LTE [2] [6]. In VoLTE, a software upgrade is required to the LTE network and its PS core network (EPC). VoLTE uses a QoS Class Indicator value equal to one (QCI=1) and the Conversational QoS class for either originating or terminating a voice call. For more detailed information about the procedure of UE to originate a voice call in a roaming scenario refer to [7]. According to the 3GPP technical specification in [6] IMS is an access independent based on the Session Initiation Protocol (SIP), defined by the Internet Engineering Task Force (IETF) to support voice and other multimedia services. The reference architecture of IMS is illustrated in Fig. 2. IMS provides a complete solution to handle voice over all-IP and PS wireless networks. GSM Association (GSMA) announced in 2010 it will consider IMS as a major solution in the one voice profile recommendations [8].

The first step of User Equipment (UE) to start a voice call is an IMS registration. Next the UE obtains the required bearers to complete the call followed by IP address allocation to be known by other users. Multi Media Telephony (MMTel) has originated in 3GPP Release 7. It is a service set in the IMS standard architecture that defines both Network to Network Interface (NNI) and User to Network Interface (UNI) [10]. It offers real time multimedia services based on IMS and allows users to use voice and other services. One of the major roles of MMTel is to maintain service quality of a minimum performance voice and video which support 3GPP codecs.

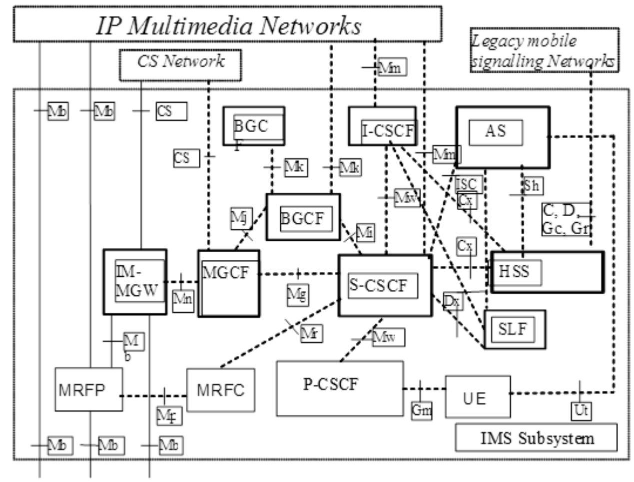

Figure 2. The IMS Reference Architecture [9]

B. *Circuit Switched Fall Back (CSFB)*

CSFB is a 3GPP standard bridging technology between the LTE PS and legacy CS wireless networks to obtain CS services [11]. The NGMN alliance has recommended CSFB to enable non-IMS roaming subscribers to use both PS and CS voice services in legacy CS networks. The precondition in CSFB is the LTE coverage must overlap with UTRAN/GERAN. CSFB was specified in the 3GPP technical specification in [11]. CSFB is an interim solution which is suitable to use when the visiting LTE networks do not have IMS or IMS still not fully deployed.

C. *Voice over LTE via Generic Access (VoLGA)*

VoLGA is a different mechanism which was defined by the VoLGA forum in 2009 based on the 3GPP Generic Access Network (GAN) specified in [12] and [13]. VoLGA connects the LTE PS network with MSC/VLR in UTRAN/GERAN using a special gateway called VoLGA Access Network Controller (VANC). However, VoLGA has not been accepted by the 3GPP standardization body as standard technology to provide voice to LTE users, which is the biggest disadvantage of this technology.

D. *Over The Top (OTT)*

Over The Top (OTT) means providing voice services through third party providers such as Skype or Google Talk. This service is provided either free of charge or is relatively inexpensive. Mobile operators might use OTT when they do not want to invest too much on deployment a very expensive IMS. However, there is no guaranteed QoS and no service continuity using this method, especially when UE moves outside an LTE coverage area. Call drop and call failure is always possible using this method **[**13**]**.

III. QOS IN E-UTRAN-BASED WIRELESS NETWORKS

Quality of Service (QoS) is a concept of providing a particular quality for a specific type of service. QoS is one of the main and the greatest challenges for the IP-based services that lack of a dedicated connection channel. As part of the rapid growth of multimedia applications over cellular networks, the QoS needs to be maintained a guaranteed service through wireless networks [14]. It is essential for LTE to provide an efficient QoS solution that the user experience of each service running over the shared radio link is satisfied. Thus, the Evolved Packet System (EPS) system selects different QoS data flows for each service.

A) QoS Architecture in LTE Networks

3GPP introduced the QoS architecture of the LTE/EPS in R8 technical specifications. As can be seen from Fig. 3, this end-to-end class-based QoS architecture has been introduced to support a mix of Real Time (RT) and non-Real Time (non-RT) services.

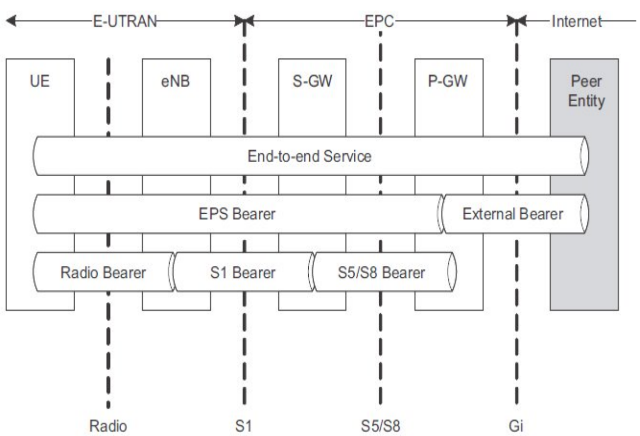

Figure 3. LTE/EPS Bearer Service Architecture [17]

The QoS in EPS is based on the data flows concept and bearers. Such flows of data are established between the UE and the Packet Data Network Gateway (PDN-GW) and mapped to bearers, with three individual bearers (Radio, S1 and S5/S8). The combinations of them provide the end-to-end QoS support to the LTE system. With the help of bearers, a scalar value, referred to as a QoS Class Identifier (QCI) with the help of bearers specifies the class to which the bearer belongs [15]. Table I illustrates the standardized QoS classes.

B) QoS for Voice over LTE (VoLTE)

For running VoLTE service over LTE networks, two default and one dedicated bearers are mostly required [18]. The first bearer is the default bearer, which is used for signaling messages. IMS uses this default bearer with QCI=5 for any SIP signaling related to it. The Packet Delay Budget (PDB) in this bearer is 100 ms between the PDN-GW and the UE with up to 10^{-6} Packet Loss Rate (PLR). This default bearer has the highest priority amongst all other QCI classes. The second bearer is also a default bearer, which is used for all other TCP-based traffic (e.g., Email). Up to 300 ms PDB is allowed in this bearer with a maximum of 10^{-6} PLR. It has the lowest priority among all other QCI classes. The last bearer is dedicated bearer. This dedicated bearer is used for conversational voice (VoLTE). Up to 100 ms PDB is allowed with maximum10^{-2} PLR. This bearer has the second highest priority among all other QCI classes and unlike the other default bearers, it is always GBR. It is associated with the first default bearer with Linked EPS Bearer ID (L-EBI) and has also Traffic Flow Template (TFT) which determines the rules of sending and receiving IP packets.

TABLE I. STANDARDIZED QCI CHARACTERISTICS [16]

QCI	Resource type	Packet error/loss rate	Delay Budget	QCI priority	Example services
1	GBR	10^{-2}	100 ms	2	Conversational voice
2		10^{-3}	150 ms	4	Real-time video
3			50 ms	3	Real-time gaming
4		10^{-6}	300 ms	5	Buffered video
5	Non-GBR		100 ms	1	IMS signaling
6			300 ms	6	Buffered video, email
7		10^{-3}	100 ms	7	Voice, RT video
8		10^{-6}	300 ms	8	TCP-based services
9				9	

IV. SIMULATION ENVIRONMENTS

The OPNET modeler from Riverbed Technologies Ltd. has been used to design the baseline for the LTE wireless network. The practical side of this study started by designing a baseline for the LTE wireless network. This baseline includes a complete implementation for VoLTE, the topic in our study. The LTE baseline network consists of seven LTE base stations (eNBs), three mobile stations (UEs) in each eNB (total of 21 UEs in the whole network), one IP Multimedia System (IMS) in addition to the LTE core network (EPC). The designed network contains Application Definition, Profile Definition, Mobility Management and OPNET LTE configuration entities. In addition, a number of wired and wireless links to connect between different nodes and the LTE EPC were used. Mobility is implemented with node velocity equal to 5 meters per second (m/s). Intra-frequency is the only handover likely to happen between cells of the same frequency. The only path loss model used in each UE is set to free space with no any obstruction for the propogated signal. The UE and the eNB transmission power are set to cell size based for both of them. This parameter is used to represent the transmission power (in wattage) to use for each UE/eNB. 20 MHz FDD bandwidth has been chosen to simulate the physical profile in the LTE designed network. In order to give a very realistic scenario to the designed network, we assumed that all the 7 eNBs are located in London, UK, which will be our simulation work space with 1 km eNB radius each. Additionally, we adopted a hexagon overlay for the implemented eNBs and put the UEs randomly between these eNBs for the same reason. The overall network model of the baseline LTE network is shown in Fig. 4. The LTE mobile nodes (UEs) are programmed and configured to run VoLTE services. The lte_wkstn_adv node model is used to represent a workstation with source and destination application running over TCP/IP and UDP/IP. Table II shows the important configuration parameters of the UEs in the baseline LTE network.

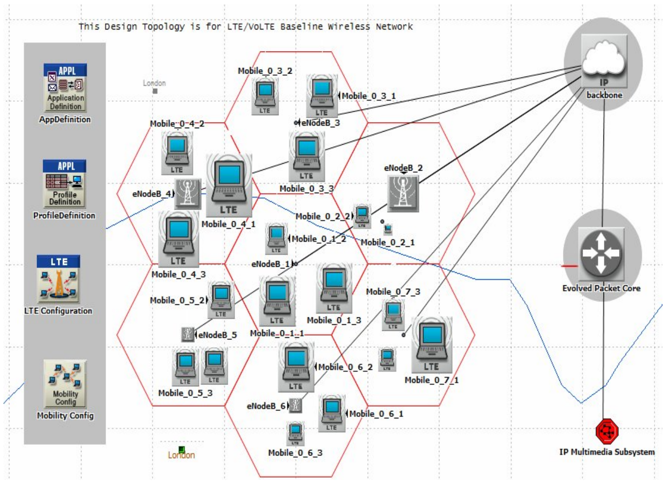

Figure 4. The Baseline LTE Designed Network (Network Model)

TABLE II. USER EQUIPMENT (UE) CONFIGURATION PARAMETERS

PARAMETER	VALUE
ANTENNA GAIN dBi	-1 dBi
MODULATION and CODING SCHEME INDEX	9
MULTIPLE CHANNEL MODEL (DL)	LTE OFDMA ITU Pedestrian B
MULTIPLE CHANNEL MODEL (UL)	LTE SC-FDMA ITU Pedestrian B
PATH LOSS MODEL	FREE SPACE
DL MIMO TRANSMISSION	Same NB Setting
NUMBER of RECEIVE ANTENNAS	2
NUMBER of TRANSMIT	1
HANDOVER TYPE	INTRA-FREQUENCY
VELOCITY	5 M/S
MEASUREMENT WINDOW SIZE	100 ms
CELL RESELECTION MEASUREMENT THRESHOLD	-112 dBm

The LTE base stations (eNBs) are programmed and configured to provide radio coverage to the UEs in the LTE network. The lte_enodeb_3sector_4slip_adv node model is used to represent the LTE eNBs. This model of eNBs includes 3 sectors in each eNB and can maintain up to 4 serial line interfaces at a selectable data. Any one of the 7 eNBs nodes can communicate with one or more UEs, in addition to the EPC. The eNBs, UEs and EPC in the design network have been programmed in a way such that each one of them has a unique ID and name. Table III shows the important configuration parameters of the eNBs in the designed network. The IMS model is used to deliver High Definition (HD) voice and a set of Rich Communications Services (RCSs); it also gives a more realistic scenario to deliver VoLTE service. The IMS model consists of Proxy Call Session Control Function (P-CSCF), Serving-CSCF (S-CSCF) and Interrogating-CSCF (I-CSCF). These components are used for the signaling procedures of the VoLTE calls between different users in the network. The IMS signaling flow in the LTE network requires the highest priority as it is the first procedure which is invoked towards the establishment of the VoLTE call. Hence, all the IMS signaling packets are marked with priority equal to 1 in both radio and core networks in the QCI [17]. Note that 1 is IMS priority while 5 is the value of IMS QCI.

TABLE III. EVOLVED NODEB (eNB) CONFIGURATION PARAMETERS

PARAMETER	VALUE
ANTENNA GAIN dBi	15 dBi
DUPLEXING SCHEME	FDD
PATH LOSS MODEL	Free Space
BANDWIDTH	20 MHz
NUMBER OF RECEIVE/ TRANSMIT ANTENNAS	2
HANDOVER TYPE	INTRA-FREQUENCY
DL MIMO TRANSMISSION TECHNIQUE	Spatial Multiplexing Codewords 2 Layers
MEASUREMENT THRESHOLD	-44 dBm
eNB SELECTION THRESHOLD	-110 dBm

The EPC is one entity which includes all the main required core network parts; the Mobility Management Entity (MME), the Serving Gateway (S-GW), and the Packet Data Network Gateway (PDN-GW). The lte_access_gw_atm8_ethernet8_slip8_adv model is used to represent the LTE core network. The voice model used to generate VoLTE in the designed LTE network is a G.711 Pulse Code Modulation (PCM) voice codec. A summary of the G.711 parameters which have configured in the baseline simulation is illustrated in Table IV.

TABLE IV. VOICE CONFIGURATION PARAMETERS

PARAMETER	VALUE
VOICE FRAMES PER PACKET	1
TYPES OF SERVICE	Interactive Voice
SILENCE LENGTH (SECONDS)	0.65 Second
TALK SPURT LENGTH	0.352 Second
COMPRESSION DELAY	0.02 Second
DECOMPRESSION DELAY	0.02

The VoLTE application in OPNET simulation enables two UEs to establish a virtual channel and they can communicate using digitally encoded voice signals. Fig. 5 describes the data traffic flow in OPNET LTE simulation

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between EPC and the LTE eNB through GPRS Tunneling Protocol (GTP). GTP tunneling is located at both nodes (EPC, eNB) and it is dynamically established between them to carry the EPS required bearers shown before in Fig. 3. The User Datagram Protocol *(*UDP) is the default transport protocol used for this application. The voice data arrive in spurts called talk spurts that are followed by silent periods. A talk spurt is an uninterrupted burst or a period of time in which the listener does not detect a pause. During a silent period, packets are transmitted quite rarely. Internally, the voice packets are sent over Real-Time Protocol (RTP) streams. Traffic is generated in the network model only when the application is active, therefore the traffic duration equals the application duration. The voice application in our simulation starts at 100 seconds from the simulation start time. The period of time before the traffic is generated is called warm-up time, which is very important for any simulation scenario [4]. The reason why this time is important is because any simulation running is started with empty systems. During this time all buffers are configured before starting the traffic generation.The node and the PHY process models of the LTE eNB and LTE UE of our simulation are illustrated in Fig. 6 and Fig. 7, respectively.

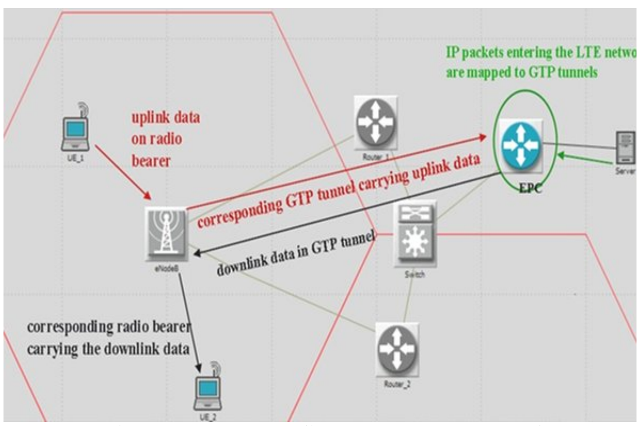

Figure 5. GTP Tunneling Between LTE EPC and LTE eNB and Data Traffic Flow

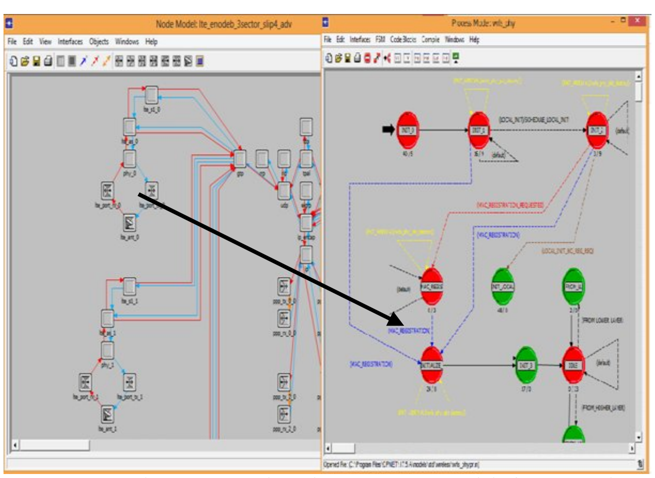

Figure 6. Node and PHY Process Models for LTE Enb

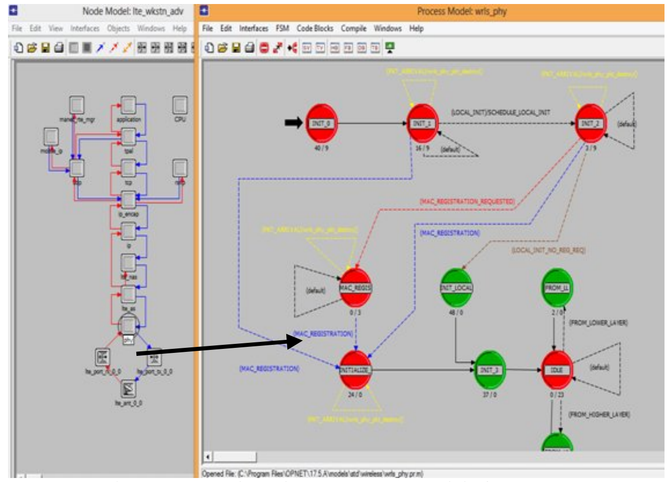

Figure 7. Node and PHY Process Models for LTE UE

V. SIMULATION RESULTS

The performance evaluation has been conducted in terms of different QoS factors such as end-to-end delay, packet loss rate and jitter in two main scenarios.

A. VoLTE QoS Baseline Scenario

The VoLTE end-to-end delay (mouth-to-ear delay) is one of the most important factors to consider when we measure the VoLTE QoS. It should be strictly maintained under reasonable limits and must be carefully monitored. End-to-End delay is measured from the ingress of the UE at the sender side to the egress of the UE at the receiver side. The equation used to calculate this QoS factor based on our simulation design is:

VoLTE end_to_end_delay= Network delay+Encoding delay+Decoding delay+Compression delay+Decompression delay+Dejitter_buffer delay (1)

Fig. 8 shows end-to-end delay for VoLTE service in the baseline LTE network during 600 seconds simulation time. The X-axis represents the simulation time in seconds, while the Y-axis represents the end-to-end delay in seconds. The VoLTE traffic starts at 100 seconds as we mentioned earlier (see Section IV for more details). At 100 seconds, the end-to-end delay was 165 ms. There was a slight increase in this value to become 184 ms and then a gradual decrease until reaching its stable level equal to 119 ms. It then continues with that value until the end of the simulation. The average end-to-end delay for the VoLTE is found 126 ms. This value fulfills the ITU-R and the 3GPP standard requirements with up to 150 ms for one way VoLTE end-to-end delay to experience high quality [19] [16]. In fact, an end-to-end delay up to 250 ms is still quite satisfactory for the majority of users if we considered about 100 ms required delay for packet processing and propagation delay in the core network [20] [21].

The VoLTE packet loss rate is another important QoS factor to examine. Packet loss rate generally refers to the percentage of packets that are lost during the transition from the sender to the receiver in the network. Ideally, in VoLTE steady networks, there should be no packet loss [22]. However, VoLTE users are still satisfied if this percentage is a maximum 2% based on the 3GPP requirements [23] [16]. This means, at least 98% of the total transmitting packets have to arrive successfully to the final destination. The equation to calculate the packet loss rate is as follows:

$$\text{Packet Loss Rate} = \frac{\text{Packets sent - Packets received}}{\text{Packets sent}} * 100\ \% \quad (2)$$

Fig. 9 demonstrates a comparison between VoLTE traffic sent and traffic received. The X-axis represents the simulation time in seconds, while the Y-axis represents the VoLTE traffic sent/received in packets per seconds. It is clear from Fig. 9 that the VoLTE traffic sent/received increased sharply after 100 seconds. At 104 seconds, the rate of the packets sent was 69.64 packets/second, while on the other hand, the number of the packets received was 67.41 packets/second at the same time. The traffic reaches its peak values at 138 seconds and then it stays on this steadily level with 1800 packets/second until the end of the simulation. Overall, the amount of traffic generated and received was almost identical. This is due to the stable LTE network that does not involve any congestion by other applications. The total number of packets sent/received in the baseline scenario were 290300/290286 packets respectively. The packet loss rate for VoLTE was found to be 0.0048%, which is an excellent rate. This result meets the ITU-R and 3GPP standard requirements which were clarified before in Section I.

The difference in response time between different packets received in the destination side is called jitter. For any stable system with steady packet stream, the value of this QoS factor should be always 0 as there is no variation in the delay of the received packets. However, if the jitter is so large then it can cause an out of order situation to the receiving packets. This can lead to confusion if the working application is a voice service, which results in poor service quality. The ITU-R has recommended 25 ms jitter as an acceptable value for the delay variation [21]. From Fig. 10, it can be seen that the jitter value was - 0.00000136 at time 120 seconds and then after 6 seconds (time=126 seconds) becomes 0 and stayed at this value until the end of the simulation. Negative jitter indicates that the time difference between the packets at the destination node was less than that at the source node. This result indicates clearly that the overall jitter value is ideal and reflects that the designed system is very stable.

B. LTE Bandwidth Implementation Scenario

One of the interesting features of LTE is its ability to support scalable bandwidths from 1.4 MHz up to 20 MHz. It is necessary to examine the effects of using different LTE bandwidths on the service quality and end-user experience. In the second scenario, we implemented other bandwidths (1.4 MHz and 5 MHz). Fig. 11 compares between end-to-end delay in the three different LTE bandwidths (1.4, 5 and 20 MHz). The higher the bandwidth (20 MHz) is the higher the data rate supported, as a result, the lowest end-to-end delay. 1.4 MHz is the highest end-to-end delay in the same figure. However, all the results for all the implemented bandwidths are still within the acceptable threshold of 150 ms. Fig. 12 shows a comparison between the same groups of LTE bandwidths, but in this case in terms of the VoLTE jitter. For jitter calculation, we used the following formula:

$$\text{Jitter} = (T4 – T3) – (T2 – T1) \quad (3)$$

Where:

T1 and T2: The time of leaving two consecutive packets from the source node.

T3 and T4: The time of arrival same packets to the destination node.

Contrary to expectations, Fig. 12 shows that LTE jitter with 20 MHz has better jitter values than LTE jitter with 5 MHz, which in turn also, has a better jitter values than 1.4 MHz as it includes many negative values. These results reflect that the higher the data rate gives better jitter performance. However, in fact, theoretically, there is no direct relation between LTE bandwidth and its jitter. The higher bandwidth can affect positively on the end-to-end delay, but not necessarily lead to reduced jitter value.

The downlink (DL) delay in LTE networks is the time started from when the traffic arrives at the LTE layer of the eNBs until it is delivered to the higher layer of the corresponding UEs. On the other hand, the uplink (UL) delay in LTE networks is the time started from when the traffic arrives at the LTE layer of the UEs until it is delivered to the higher layer of the corresponding eNBs. It is straightforward to show that the Packet Delay in the DL direction is less than the UL direction, although the resulting delay in both sides follows the same standard requirements of 50 ms one way delay [20]. This is due to the higher power in the DL side from the eNBs which is around 43 dBm, compared to the UE’s power which is about 23 dBm in the UL side [17]. The UL/DL delay in the designed LTE network for different LTE bandwidths is illustrated in Fig. 13 and Fig. 14 respectively. As can be seen from these figures, the relation between the UL/DL delay and the LTE bandwidth is directly proportional. The results show that there is up to 0.03 seconds DL delay and up to 0.017 seconds UL delay. These results meet the requirements mentioned before, for the one way voice radio UL/DL delay.

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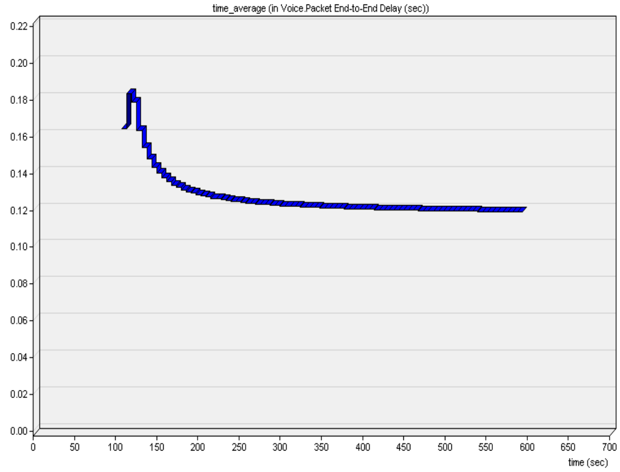

Figure 8. VoLTE End-to-End Delay

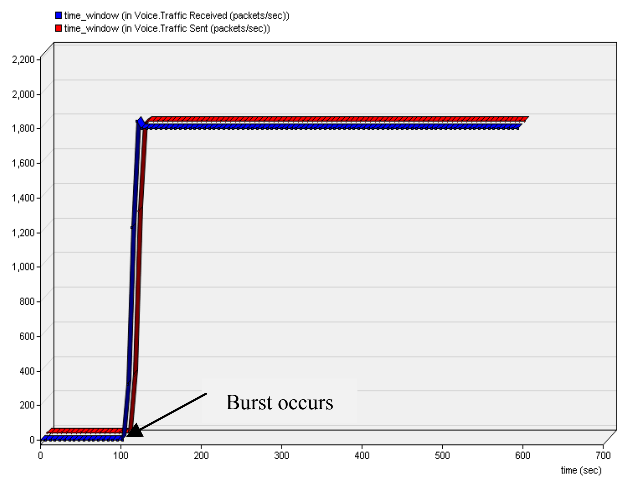

Figure 9. VoLTE Traffic Sent/Received

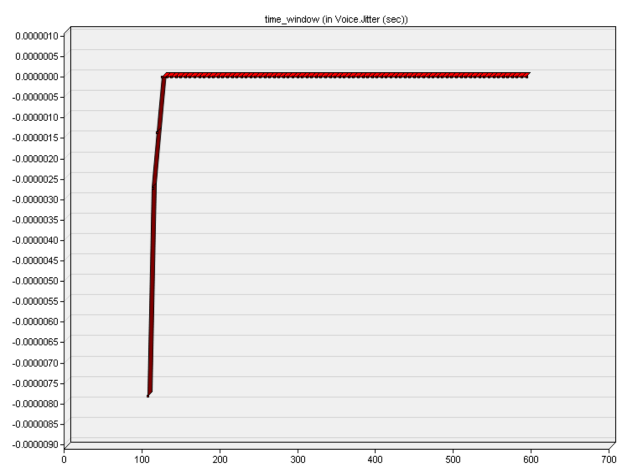

Figure 10. VoLTE Jitter

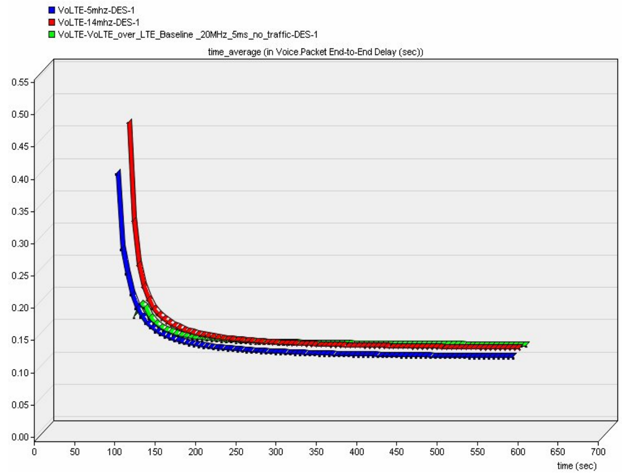

Figure 11. End-to-End Delay for Different LTE Bandwidths

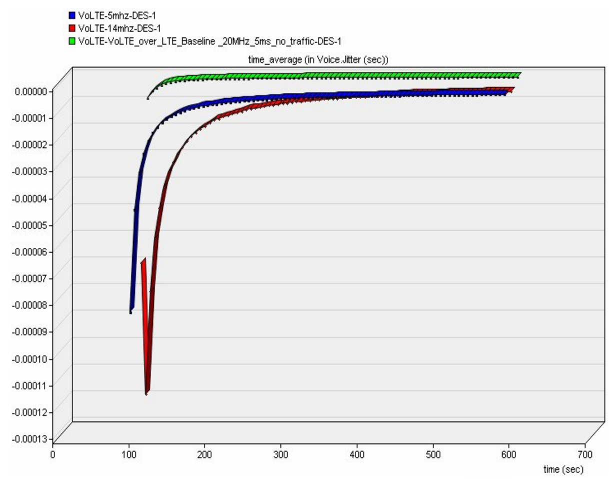

Figure 12. Jitter for Different LTE Bandwidths

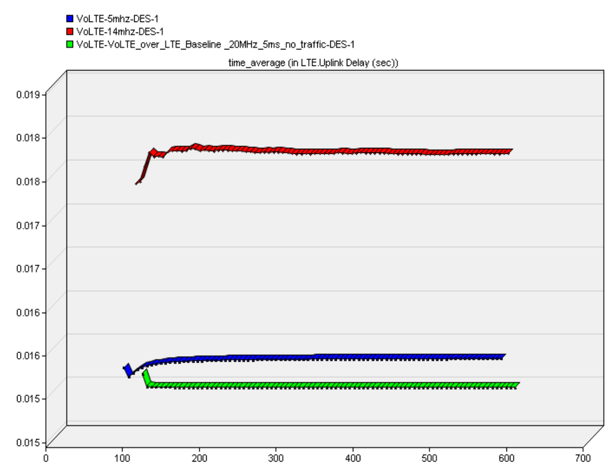

Figure 13. LTE Uplink Delay for Different Bandwidths

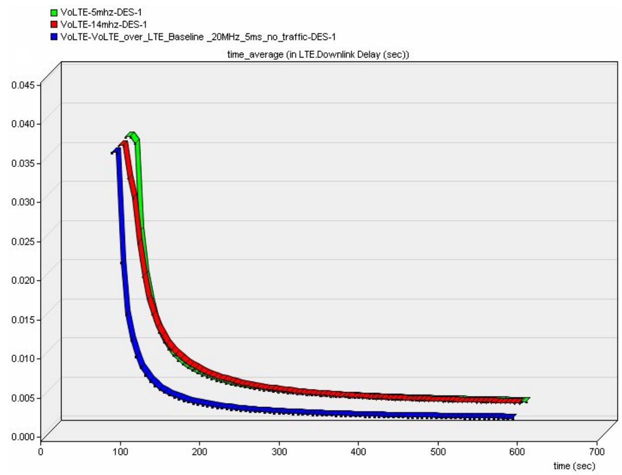

Figure 14. LTE Downlink Delay for Different Bandwidths

VI. CONCLUSION

In this paper, a realistic Voice over LTE (VoLTE), including IMS over the baseline LTE wireless network was simulated and its performance in terms of Quality of Service (QoS) was evaluated and validated using OPNET modeler wireless suite 17.5. VoLTE is a standard technology that is required to support packet voice calls over a purely Packet-Switched (PS) LTE wireless networks. It provides better QoS, which results in better end-user experience over CSFB, VoLGA and OTT. In conclusion, this work has demonstrated that the simulation results have matched the ITU-R and 3GPP standard requirements related to the VoLTE over 4G LTE. The simulation results are significant in three different QoS respects; end-to-end delay, jitter and packet loss rate. It has been found that the overall VoLTE end-to-end delay was about 0.12 ms, and its packet loss rate was about 0.005%, while jitter was almost 0. Furthermore, another different simulation scenario was designed to investigate the effects of different LTE bandwidths on the VoLTE service quality. Three different LTE bandwidths (1.4, 5 and 20 MHz) were implemented and their effects on the VoLTE end-to-end QoS and LTE DL and UL delays were also studied. The results show that LTE can achieve better performance with a 20 MHz bandwidth.

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REFERENCES

[1] ETSI TS 136.300, “Evolved Universal Terrestrial Radio Access (E-UTRA) and Evolved Universal Terrestrial Radio Access (E-UTRAN); Overall description; Stage 2,” v8.4.0, April 2008.

[2] M. Sauter, Beyond 3G-Bringing networks, terminals and the web together: LTE, WiMAX, IMS, 4G Devices and the Mobile Web 2.0: Wiley. com, 2011.

[3] D. Astély, E. Dahlman, A. Furuskar, Y. Jading, M. Lindstrom, and S. Parkvall, “LTE: the evolution of mobile broadband,” Communications Magazine, IEEE, vol. 47, pp. 44-51, 2009.

[4] ITU-T Recommendation G.114, “One way transmission time”, February 2003.

[5] C. C. Lin, K. Sandrasegaran, H. Ramli, and M. Xue, "Requirement of handover modeling in the downlink 3GPP long term evolution system," in Advanced Information Networking and Applications Workshops (WAINA), 2010 IEEE 24th International Conference on, 2010, pp. 305-310.

[6] ETSI TS 124.237, “LTE; IP Multimedia (IM) Core Network, IMS service continuity; Stage 3,” v10.3.0, June 2011.

[7] M. R. Tabany and C. G. Guy, " Performance Analysis and Deployment of VoLTE Mechanisms over 3GPP LTE-based Networks," International Journal of Computer Science and Telecommunications, Vol. 4, Issue. 10, Oct. 2013, pp.1-8.

[8] R. Ferrus, O. Sallent, and R. Agusti, "Interworking in heterogeneous wireless networks: comprehensive framework and future trends," Wireless Communications, IEEE, vol. 17, pp. 22-31, 2010.

[9] 3GPP TS 23.228, “Technical specification Group Services and System Aspects; IP Multimedia Subsystem (IMS); Stage 2, (Release 9) v9.3.0, March 2010.

[10] ETSI TS 122.173, “UMTS; LTE; IP Multimedia Core Network Subsystem (IMS) Multimedia Telephony Service and supplementary services; Stage 1,” v10.3.0, April 2011.

[11] ETSI TS 123.272, “Technical Specification Group Services and System Aspects; Circuit Switched (CS) fallback in EPS; Stage 2,” (Release 10), v9.13.0, January 2013.

[12] VoLGA TS stage 3, “Voice over LTE via Generic Access; Stage 3 Specification; Phase1,” v1.5.0, June 2010.

[13] 3GPP TR 43.318, “Technical Specification Group GSM/EDGE Radio Access Network; Generic Access Network (GAN); Stage 2 (Release 8),” v8.4.0, February 2009.

[14] L. Li and S. Shen, "End-to-End QoS performance management across LTE networks," in Network Operations and Management Symposium (APNOMS), 2011 13th Asia-Pacific, 2011, pp. 1-4.

[15] H. Ekstrom, "QoS control in the 3GPP evolved packet system," Communications Magazine, IEEE, vol. 47, pp. 76-83, 2009.

[16] 3GPP TS 23.203, “Technical Specification Group Services and System Aspects; Policy and charging control architecture (Release 10),” v10.6.0, March 2012.

[17] S. Sesia, I. Toufik, and M. Baker, LTE: the UMTS long term evolution: Wiley Online Library, 2009.

[18] GSMA PRD IR.92: “IMS Profile for Voice and SMS, ” March 2013.

[19] Y. Fan, M. Kuusela, P. Lunden, and M. Valkama, "Downlink VoIP support for evolved UTRA," in Wireless Communications and Networking Conference, 2008. WCNC 2008. IEEE, 2008, pp. 1933-1938.

[20] J. Puttonen, T. Henttonen, N. Kolehmainen, K. Aschan, M. Moisio, and P. Kela, "Voice-over-IP performance in UTRA Long Term Evolution downlink," in Vehicular Technology Conference, 2008. VTC Spring 2008. IEEE, 2008, pp. 2502-2506.

[21] S. A. Ahson and M. Ilyas, VoIP handbook: applications, technologies, reliability, and security: CRC Press, 2008.

[22] Cisco,“Quality of Service for VoIP,” Cisco systems, 2011.

[23] T. Henttonen, et al., "Performance of voip with mobility in utra long term evolution," in Vehicular Technology Conference, 2008. VTC Spring 2008. IEEE, 2008, pp. 2492-2496.