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A Review on Voice over Internet Protocol (VOIP) over LTE Networks

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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).The Quality of Service (QoS) is a necessary issue for real time application i.e. video and voice telephony. For satisfying the users requirement, it is essential to improve the Quality of Service. A throughput performance improvement for Voice over IP (VoIP) applications in the static LTE network by utilizing a disseminated Client-Server model is formulated to make the services better that are supplied to the end user. The proposed Client-Server model is modelled in the OPNET simulator 16.0 with several Subscribers Stations (SSs), Base Stations (BSs), and some Server Base stations is chosen by using the Nearest Neighbourhood Algorithm utilizing Orthogonal Frequency Division Multiplexing (OFDM) methods. The suggested method is equated with the subsisting Centralized model which utilizes FDM (Frequency Division Multiplexing) techniques. The results got for the VoIP throughput of the suggested Client-Server model depicts that network performance is improved.

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 [2]. 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 [11]. Call centres needs Quality of unsuitable infrastructure. A call center utilizes VOIP for having best economic standing but many call centre 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 [12]. VOIP cannot be as good as PSTN, but a call centre 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.

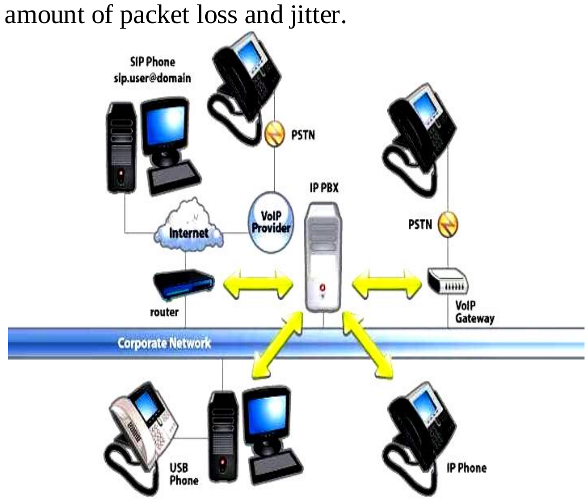

Figure 1: Structure of Voice over Internet Protocol (VOIP)

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 [3]. 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.
- 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 VIA IP MULTIMEDIA SUBSYSTEM (VOLTE)

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 2. 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.

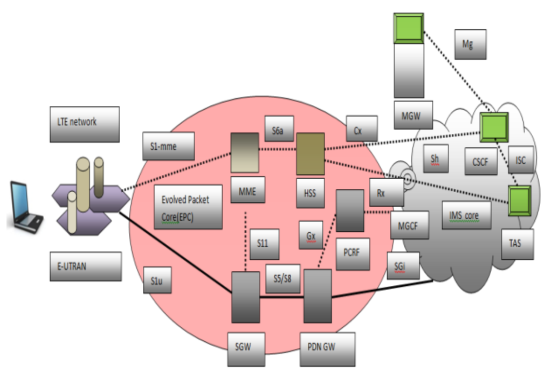

Figure 2: VoLTE

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.

III. QOS ARCHITECTURE IN LTE

In LTE, the QoS is provided by means of a bearer which uniquely identifies the packet flow between the user and the PDN-GW and is responsible for the priority that is given to a packet flow across the LTE network. Bearers are established after the successful authentication and registration of the user in the LTE network. The LTE bearer architecture is shown in the Figure 3. Each bearer is associated with a Traffic Flow Template (TFT) which is used to differentiate the types of packets that flow through it. The TFT does this classification based on one of the following parameters:

- Port numbers
- ToS/DSCP Values
- Source/Destination address
- Protocol (TCP/UDP)

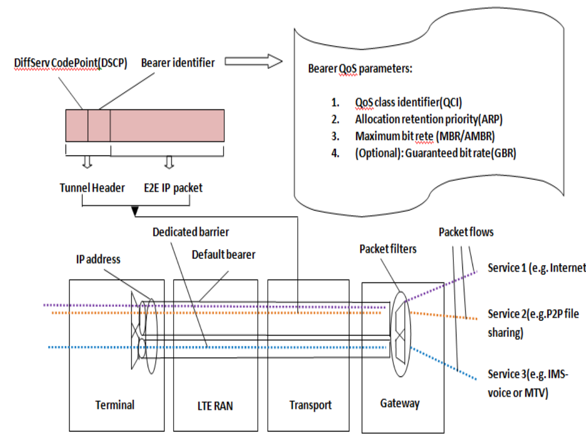

Figure 3: QoS Architecture in LTE

IV. 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.

V. 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.
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]
For measuring the performances of end-to-end VoLTE in this work any network transport elements are maintained (for instance, SIP signalling, DSCP and other QoS IP-based characteristics, etc) but only voice codec's. [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)

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

VI. PROPOSED DISTRIBUTED MODEL

As talked about by Lawal et al. [11], the client server base stations are chosen at the central server MAC layer utilizing Nearest Neighbourhood Algorithm. The proposed distributed model was modelled by using OPNET simulator 16.0 with various SSs, BSs and many Server BSs utilizing OFDM schemes. In this client server model, Server Base stations are chosen utilizing Nearest Neighbourhood Algorithm for providing information about network from a central server to the closest Client Base stations utilizing OFDM schemes. This decision of selection is done on the central server MAC layer. The transformation of information between the Server BSs and central server utilizes a Client-Server algorithm. Communication between Client BSs and Server BSs utilizes Client-Server communication algorithm for providing information of network between Server BS and Client BS. The LTE network configuration dimensions of the Centralized client server model in which the MAC service class definitions, SC PHY profile and FDM PHY profile in the static LTE network were organized . Also the LTE configuration dimensions for the disseminated model in which the MAC service class definitions, SC PHY profile and OFDM PHY profile in the static

LTE network were organized as illustrated in the Figure 4.

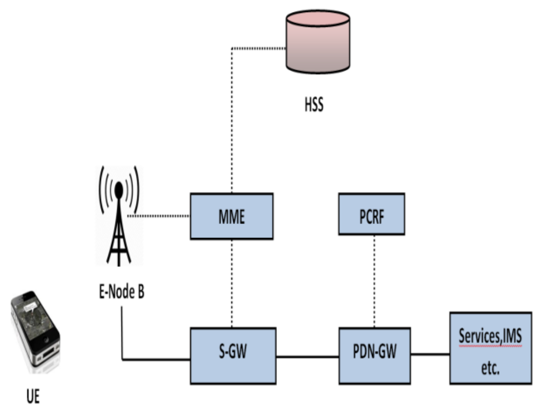

Figure 4: LTE Architecture

CONCLUSION

Quality of Services is the major challenge in VoIP systems. A VoIP application needs a better throughput, a higher fairness index and less packet drop throughout the network. The VoIP throughput was planned to provide reliable and continuous end-to-end data transfer over the networks. In this paper, a model for throughput performance enhancement for VoIP applications in the static LTE network was demonstrated utilizing disseminated Client-Server for providing the end user with enhanced network.

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