

# QoS in VoIP over 3G network and Pricing Strategy

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**Abstract—Research has shown that the voice over IP (VoIP) technology has been experiencing a major quality of service (QoS) problems, such as, jitter, delay (latency) and packet loss. These factors degrade the voice quality, which causes dissatisfaction on the VoIP users. The quality of VoIP communications also depends on the resource availability. This paper concerns VoIP performance evaluation over 3G network.**

**Techniques that are going to be employed in this research to manage QoS are: Call Admission Control (CAC) to prevent congestion in voice or packet traffic, the E-Model for quality of voice call prediction and pricing strategy as a tool to improve QoS in VoIP and render satisfaction to end users. The OPNET modeler will be used for simulation purpose.**

**Index Terms— QoS, UMTS, VoIP.**

## I. INTRODUCTION

In recent years, voice over internet network services has become a widely used service. Internet network service was firstly designed for busy data transportation and not optimized for real-time traffic, but recently Internet has shown an interest towards VoIP services. The VoIP technology has been used in some events to replace traditional long-distance telephone technology. VoIP application which operates over the Third Generation (3G) network developed by Third Generation Partnership Project (3GPP), is supported by a broadband wireless access supplied by a High Speed Download Packet Access (HSDPA) [1]. The 3G networks are based on the Universal Mobile Telecommunications System (UMTS) platform.

The VoIP technology brings added value services that are not available in the Public Switched Telephone Network (PSTN) service and providing a low cost to the end-user. VoIP infrastructure and services should be commercially viable. Thus, QoS offered by VoIP providers has to be close to the one provided by the PSTN. VoIP in 3G networks has most advantages over the PSTN. However, VoIP has challenges, which include packet loss, jitter and delay which affects voice quality. The VoIP challenges, including the availability of the network are the major factors in which the quality of VoIP communications depends on [2]. The focus is on analyzing the VoIP call quality over (UMTS) network using OPNET simulator.

This paper is structured into 4 sections: Section II presents the previous related work, Section III outlines the research goals and objectives, Section IV presents the methodology of this study and finally, section V presents the conclusion.

## II. RELATED WORK

There has been quite a lot of research done in the past, involving the quality of VoIP in 3G system. Several techniques that are believed to bring improvements in VoIP quality have been proposed and techniques of how to control call and data congestion due to QoS factors.

To initiate a VoIP call, at least, signaling protocols, that include, Session Initiation Protocol (SIP), H.323, H.248 (MEGACO) and MGCP [3] are required. SIP is defined in [4, 5].

In VoIP technology, a codec is essential for encoding and decoding speech. There are many types of codecs that can be used for this function. [6] proposed packet loss reduction to VoIP by means of AMR codec speech, whereby AMR codec maintains the toll quality of speech signals. According to [7], an AMR codec is a compulsory codec for conversational speech services within 3G systems. AMR codec consists of eight bit-rates which range from 4.5 kbps to 12.2 kbps and it is able to switch its bit-rate every 20 ms of speech frame depending on channel and network conditions [8].

[7] proposed an end-to-end quality of service analysis in VoIP over 3G networks, whereby they checked if jitter in 3G networks have a negative effect on the end-user voice quality. According to [1], an E-model technique evaluates the quality of VoIP in wireless networks. This E-model technique accepts a wide range of telephone damages into consideration, like damages due to low-bit rate coding, one-way delay, echo and noise [9].

In [10], guaranteeing end-to-end quality to a VoIP call over UMTS network is proposed, whereby the VoIP application parameters (voice codec, packet size and de-jittering delay), and UMTS air interface parameters (coding rate and interleaving span) are used. To ensure the quality of the VoIP call, the delay and the loss of voice packet over the air interface need a thorough control mechanism. A methodology to set VoIP application parameters and UMTS

air interface parameters, so that the quality of VoIP calls within a UMTS party is ensured, was developed in [11].

Failure to future network upgrades can be due to non-profit to internet providers. Therefore, this may cause poor internet services that lead dissatisfaction to end-users. Pricing strategy is another tool used in real-time application to ensure QoS.

The dynamic pricing that maximizes revenue while satisfying blocking rate target has been developed in [12], whereby a calculus of variations and Lagrangian mechanism to solve the carried load problem is utilized. This carried load may be the result of packet delay on the network. In [13], a packet-marking based pricing scheme is proposed. This scheme is for networks with multiple service providers.

### III. RESEARCH GOAL AND OBJECTIVE

The purpose of this study is to improve the quality of service in VoIP over 3G network and propose pricing strategy that will control and manage the usage of the 3G network resources. The objectives of the proposed study involve the following:

- To improve QoS factors in VoIP over 3G networks.
- To study and implement the techniques, such as, (1) E-model that will control and manage the impairments brought by the packet delay, packet loss to ensure quality voice and (2) Call Admission Control (CAC) to prevent voice traffic congestion in the network.
- To come up with a pricing strategy that will control call admission and improve QoS.

### IV. RESEARCH METHODOLOGY

The design of this research is attempted as follows:

- Literature review in 3G network, VoIP QoS and Pricing Strategy.
- Modelling of VoIP over 3G/UMTS network and Pricing Strategy.
- Simulation of the proposed models using OPNET modeller and MATLAB for pricing strategy.

#### A. Modeling

The VoIP over UMTS network model has been developed, where a VoIP server is connected in the UMTS model. The QoS factors will be controlled and managed to ensure good quality VoIP call. The investigation of the existing pricing strategy models is being conducted.

#### B. Simulation

The developed VoIP over UMTS network model is going to be simulated and evaluated on OPNET modeler and

the pricing strategy model will be simulated using MATLAB software.

### V. CONCLUSION

Based on a review on existing work, this paper focuses on implementing the techniques that will ensure QoS in real-time application. Especially, when transporting voice over IP network. These techniques have been studied and are in the process of implementation.

### REFERENCES

- [1] A. Arjona, *et al.*, "Towards high quality VoIP in 3G networks an empirical approach," *I.J. Communications, Network and System Sciences*, vol. 4, pp. 285-385, 2008.
- [2] E. Myakotnykh, "Adaptive speech quality in Voice-over-IP communications," Degree of Philosophy Ph.D. Dissertation, Information Science and Telecommunications, University of Pittsburg, 2008.
- [3] Tasuka, "VoIP with SIP."
- [4] J. Cumming, "SIP market overview," 2003.
- [5] G. americas, "Why SIP-I: a switching core protocol recommendation for GSM/UMTS operators," 2007.
- [6] J. W. Seo, *et al.*, "A study on the application of an AMR speech codec to VoIP," 2001.
- [7] R. Cuny and A. Lakaniemi, "VoIP in 3G networks: An end-to-end Quality of Service analysis."
- [8] J. Matta, *et al.*, "A source and channel rate adaption algorithm for AMR in VoIP using the Emodel," pp. 92-99.
- [9] J. Cao, "An E-model implementation for QoS across a hybrid UMTS network," Masters, RMIT University, 2009.
- [10] F. Poppe, *et al.*, "Guaranteeing quality of service to packetised voice over the UMTS air interface," presented at the IEEE, 2000.
- [11] F. Poppe, *et al.*, "Choosing the UMTS air interface parameters, the voice packet size and the dejittering delay for a voice-over-IP call between a UMTS and PSTN party," presented at the IEEE INFOCOM, 2001.
- [12] R. C. Hampshire, *et al.*, "Dynamic pricing to control loss systems with quality of service targets," 2008.
- [13] L. He and J. Walrand, "Pricing and revenue sharing strategies for Internet service providers."

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