Performance Evaluation of VoIP Services over UMTS-Network with Differentiated UMTS Bearer Services

Dhawa Sang Dong, Anand Gachhadar
Department of Electrical and Electronics Engineering
School of Engineering, Kathmandu University, Dhulikhel, Nepal
Email: dhawa1984sdong@gmail.com, anand.gachhadar@ku.edu.np

Abstract

Quality of service (QoS) for voice over IP network is not consistent because of absence of dedicated links as in circuit switched (CS) network. Because of wider mobility to users in UMTS network and voice over IP network being low cost service, performance evaluation for VoIP service over UMTS network is realized more important. Prioritized forwarding strategy to different traffics can be controlled allocating Forward Access Channel (FACH) Scheduling weights in Radio Network Controller (RNC) and Resource Reservation Protocol (RSVP) for the resource allocation like buffer space, bandwidth etc. In this paper, End-to-End QoS of UMTS is mapped to external IP network with Differentiated Service Code Point (DSCP) field in IP packet header specifying different per hop behavior (PHB) and significant improvement in QoS of VoIP service over UMTS network has been observed. Hybrid model of QoS enhancement with DiffServ Code Point integrated to FACH scheduling weight inside UMTS network and RSVP scenario has been studied in this work and the model has been purposed to reduce End-to-End delay maintaining better Mean Opinion Score (MOS).

Keywords: QoS, DSCP, FACH Scheduling, RAB service, CNB service

1. Introduction

As the circuit switched (CS) network is specially designed for voice signal transmission, the network has good qualitative speech in terms of MOS value and good spectral efficiency in terms of voice band [2]. Gateway GPRS service node (GGSN) enables access to external IP network without multimedia that is data services only while IP Multimedia Subsystem (IMS) defined 3GPP releases supports multimedia services too [13]. QoS is an End-to-End measure which is to be met through all the modules in the network where different applications have its own different QoS requirements [12]. When the voice packet are transmitted over packet switched (PS) network, the quality of services of the network must have closer performance in terms of QoS parameters that CS network can provide [2] and the main purpose of QoS architecture [10] in UMTS network is to optimize the delivery services to preferential service quality to the application with optimal data rate, reducing data loss and controlling latency. QoS architecture defined by 3GPP consists of UMTS Bearer services consisting Core Network Bearer (CNB) services and Radio Access Bearer (RAB) services [9] with RAB service ensuring transmission of control signal and user traffics between CN and MT and CNB efficiently controls the UMTS Backbone Network utility to ensure the UMTS QoS. With the increased data speed and high mobility services in PS networks, data over internet is becoming better choice for users in terms of cost and better option for service provider in terms of resource management and so the voice application is "de facto standard" in mobile terminals.

Depending up on the quality of signaling at the time of call setup and call termination, delivery quality and call quality, voice communication over any network is categorized as good quality or not. Performance of a VoIP service can be analyzed with the parameters End-to-End delay (including queuing delay), packet loss, jitter, throughputs and traffics intensity over the network. E-model as specified by the standard ITU-T G.107 gives the impairment factor (R) ranging from 0 to 100 with linear relation with perceived quality of voice [2]. This R factor can be transformed to MOS scale value with 3.5 or above to be acceptable quality of voice [2].

In 2G and 2.5 G wireless network, voice signal is transmitted over CS network while data signals are switched over GPRS network or infrastructure [4]. Different signaling protocols such as session initiation protocol (SIP) or H.323 are used to establish session between end users with appropriate retransmission timer [4]. SIP can use either TCP or UDP transport protocols while H.323 v1/v2 can use TCP protocols for transmission of packets [4]. Optimum selection of signaling protocols and transport protocol determines the call set up performance. In [4] different combination of protocol stacks, SIP, UDP and RLP, SIP, TCP and RLP, H.323 and TCP, are studied and SIP/UDP/RLP with adaptive retransmission timer was found to be optimum.

Since voice signal is analog, codec GSM-FR or G.729A is used to encode the voice stream at transmitting end, i.e. digitization of voice stream to transmit over IP network [1]. Encoded signal or bit streams are to be decoded to analog signal at the receiving end which is performed by codec [1].
Transmission rate can be controlled with Decision Feedback Scheme with RTCP (Real Time Transmission Control Protocol) where RTCP gives information about network conditions and quality level of reception, consequently congestion is controlled [5]. Rate control unit adopt the transmission rate according to round trip time and End-to-End packet loss [5]. Two bytes of field from error detection and correction control field in UDP header of VoIP are removal which is supported by 3GPP [6] reducing payload in UMTS network and power consumption.

A hybrid model to improve QoS is proposed in this study for VoIP service over UMTS network. In the model proposed, G.729A codec with 1ms frame size is used with DiffServ Code Points Expedited Forwarding (EF) to conversational type traffic, Assured Forwarding (AF) to streaming and interactive type traffics and Best Effort (BE) to background traffic and with RSVP and FACH scheduling where FACH is used to transport control signal and some user data in transport channel to secondary common control physical channel in physical layer whereas DCH is used for user data with dedicated channel in transport layer and mapped to dedicated physical data channel in physical layer [11].

Rest of the paper is structured as follow. Section II deals with literature review, section III with network simulation model in OPNET, section IV with evaluation of results from simulation and analysis and section V concludes the whole work and studies done in this paper.

Literature review

In [4], session set up time attention has been paid, is affected by FER which is measure of quality of wireless link. H. Fathi and et al. in [4], has evaluated setup time for VoIP session within the range of 0-10 percent of FER at radio link showing significant reduction with Radio Link Protocol (RLP) and average of 46% decrement was found with adaptive transmission timer. Signaling protocol H.323 and SIP have comparable performance for FER less than 2% while greater than 2% SIP has better performance compared to H.323.

At primitive stage of VoIP technology, Cuny and Lakaniemi in 2003 [7] evaluated end-to-end delay of 221.96 ms ranging FER 0 to 5% at radio link is developed to end-to-end delay of 150 ms with frame size of 20 ms in 2008 [1]. E.R. Vale, et al got 6 bytes of reduction to high probability signal level in VoIP is obtained with roughly 15% of power saving [6] if the network code of error correction and detection is efficiently developed.

Dimension of de-jitter buffer size and Block Error Rate(BLER) has significant impact on QoS of the PS network and BLER of 2% has transmission delay of 132.7 ms which is less than ITU-T G.114 specification [2]. MOS of value 3.91 was evaluated as acceptable value greater than threshold of value 3.5 at BLER 2% and buffer size of 180 ms and maximum playout interruption at zero buffer size and 19.2 at buffer size of 180 ms at BLER 2% [2]. Sivabalakrishnan and Manjula concerned about the improvement of QoS with adaptive data transmission rate with the network congestion level implementing Decision Feedback Scheme (DFS) [4], using RTCP which uses senders' report and receivers' report to estimate the data rate consequently packet loss and delay can be reduced focusing on optimum utilization of bandwidth available.

20 ms frame size was found to be better option for the codec G.729A and GSM-FR while for G.711 high packet losses was found in terms of end to end delay which has direct impact on MOS for voice quality [1]. Frame numbers per VoIP packet has effects on QoS as increased number results increasing delay a frame of 20 ms per VoIP packet was found to be best for GSM-FR and G.729A Codec [1]. Mallah and Islam [10] studied the impacts of data rate on QoS for different data traffics classes streaming, background, interactive and conversational class type and studied how Forward Access Channel scheduling affects the traffics over UMTS network. In [10], Mallah and Islam analyzed the mean end-to-end delay for data rate of 60 kbps and for different combination of scheduling weight to different traffic classes and found scheduling has significant impact on end-to-end delay.

Shreevastav and et al. proposed two layer scheduler (TLS) to improve QoS and cell throughput [8] and in the proposed algorithm, first layer performs resource reservation with the weight assigned based on delay sensitivity. Quality of radio link periodically measured by UE is reported as channel quality indicator (CQI) to corresponding NodeB and is used to ensure the packet delivery in the algorithm [8] where users with higher buffer occupancy and enough CQI value is dequeues first among the other users by scheduler. The simulation in NS-2 environment for proposed algorithm shows better throughput among other algorithm with improved jitter and delay in [8] indicated with peak signal to noise ratio (PSNR) results. A similar analysis was performed by Saad and et al in [12] that isstudied end to end QoS for DiffServ IP Network model and compared the result to their hybrid model

DiffServ/RSVP concluding significant improvement in End-to-End delay, packet loss, delay variation and throughput.
Jadhav and et al in [3] investigated the end-to-end delay, MOS, jitter and delay variation with the discovery of WiMAX with outstanding performance over UMTS with numbers of margins and discussed about the self configuration of switching from UMTS to WiMAX architectures and vice versa to ensure the QoS required depending up on the network loads.

SIP signaling scheme with adaptive retransmission timer has minimum end to end delay with RLP for FER of 2% or greater. From [2],it can be concluded that zero buffer size is recommended but maximum playout interruption is to be avoided and PS network is out of imagination without buffer at nodes so de-jitter buffer of size 180 ms was evaluated at BLER of 2% with acceptable transmission delay with minimum playout interruption of 19.2 at 3.91 MoS greater than threshold of 3.5. RTCP can use DFS to control the data transmission rate reducing packet loss and end to end delay and TLS can be deployed for better throughput and improved jitter and delay. Form [10] higher FACH scheduling weight significantly reduces the End-to-End delay whereas for same scheduling weight, higher delay sensitive traffics are guided to dedicated channels (DCH) and less sensitive traffics by either shared channels or by dedicated channels. From [8] and [12], RSVP and DSCP can prioritize the traffic with higher delay sensitive so that delay sensitive class traffic is dequeued

Table 1
FACH Scheduling Weight

Parameter	Scheduling weight
Signaling	10.0
Conversational	7.0
Streaming	3.0
Interactive	2.0
Background	1.0

with higher forwarding behavior in all hops in the intermediate nodes in IP network.

I II. Network Simulation

Simulation of network was done with optimal CODEC [1]

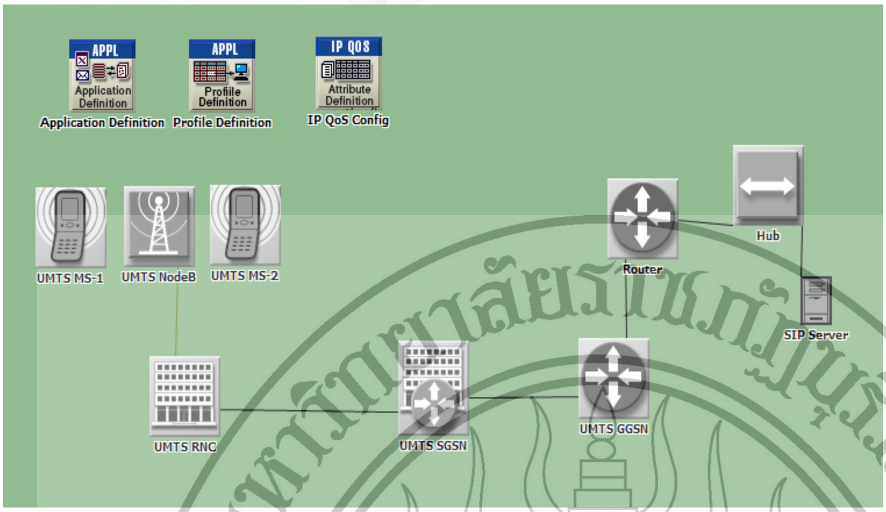

Figure1. Simulation Model

along with appropriate frame size 20 ms and transport policies RSVP, FACH scheduling and DSCP mapping to UMTS QoS are studied in two scenarios using OPNET Modeler. In first scenario VoIP traffics over UMTS network with codec G.729A, RSVP enabled and FACH scheduling weight assigned as shown in table 1 for different traffics is simulated and the results are compared with that of second scenario where DSCP has been allocated to different types of traffics as shown in table 2 along with that of scenario first.

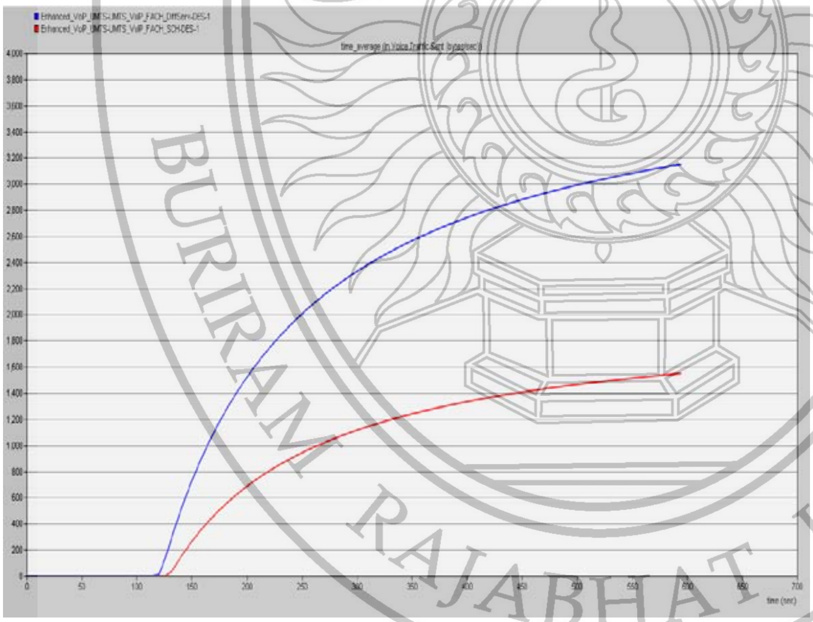

Figure 2. Traffic Sent (bytes/sec)

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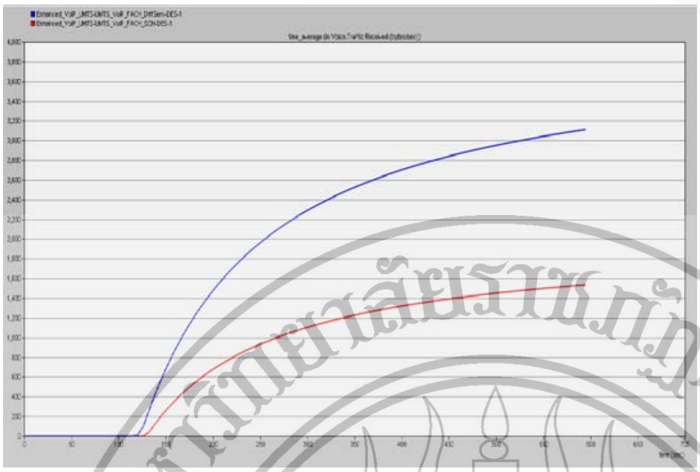

Figure 3. Traffic Received (bytes/sec)

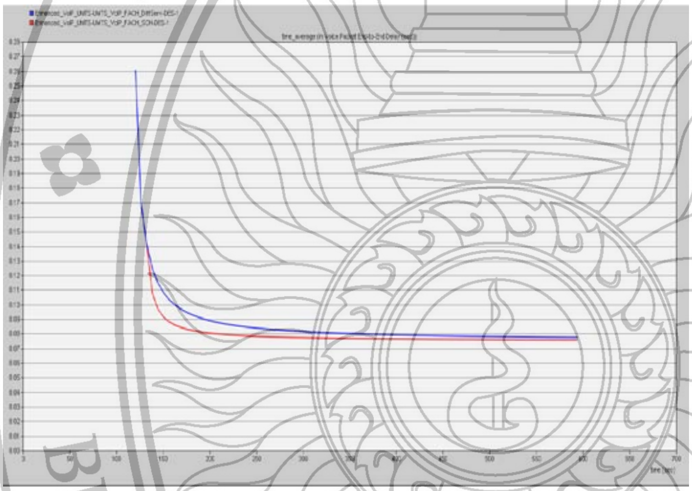

Figure 4. End-to-End delay (sec)

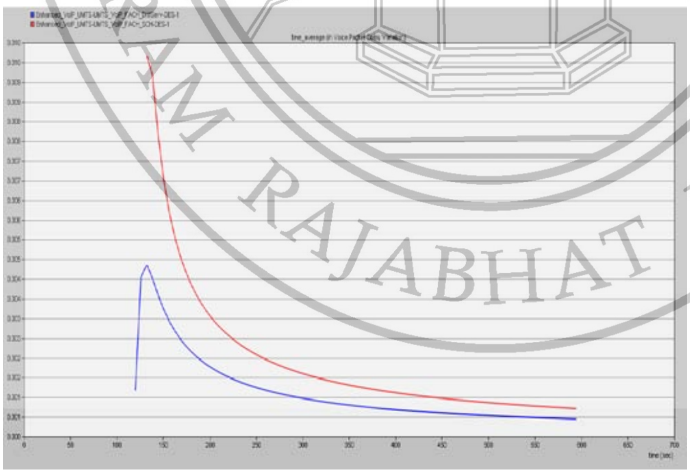

Figure 5. Delay variation

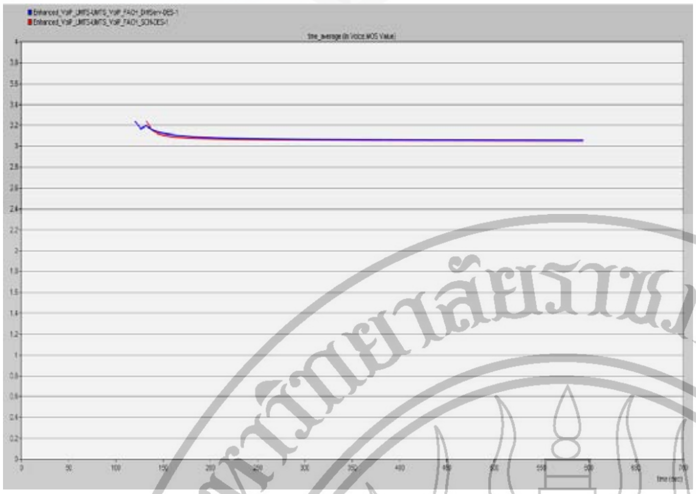

Figure 6. MOS Value

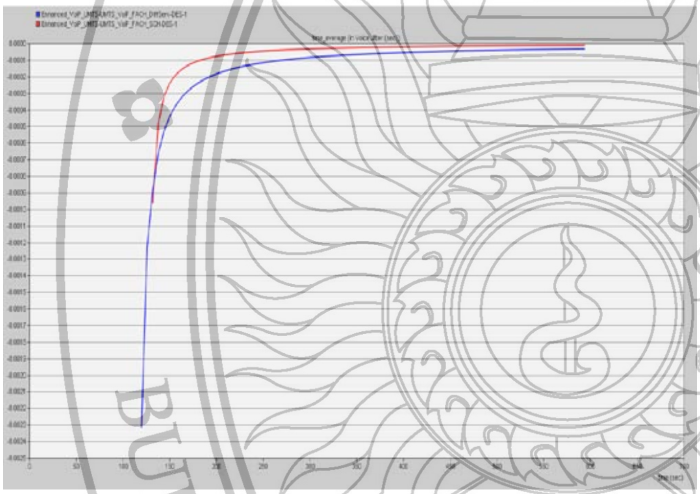

Figure 7. Jitter (sec)

IV. Result and analaysis

Fig 2, 3, 4, 5, 6, 7 are some snapshots during simulation of networks in two different scenarios and their comparative studies. For overall observation, DSCP integrated network model has significant QoS enhancement for the VoIP service over the UMTS network.

From fig. 2, 3 and 4, clearly End-to-End delay is comparable even for almost two times higher traffic intensity in case of DSCP based QoS and similar End-to-End delay converge to similar range is due to the UMTS bearer service in UMTS backbone network. Observed End-to-End delay is in the range of 80 ms as in fig.4 which is as ITU-T recommendation to be less than 150 ms and from fig. 5, delay variation has significant enhancement in DSCP integrated scenario.

From fig. 2, 3 and 6, DSCP based UMTS bearer service has similar MOS value where both the network setup has

V. Conclusion

Better CODEC as in [1] G.729A with frame size of 1 ms will have good MOS and acceptable end to end delay. Delay can be minimized with FACH scheduling weight in UMTS Backbone network where conversational traffic is given higher scheduling weight than other streaming, interactive and background traffic over the UMTS network. Similar prioritized forwarding behavior in the external IP network can be negotiated by UMTS backbone network with DiffServ Code Point marked in packet header. Thus a QoS for VoIP can be modeled within UMTS Backbone network defining different UMTS bearer service to different class of traffics and in external IP network forwarding scheme with differentiated class of traffic with different DSCP allocated to different data traffic over IP network with significant improvement in End-to-End delay and better MOS though higher traffic load over the network.

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