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UMTS VoIP Codec QoS Evaluation

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Abstract: *Voice over Internet Protocol (VoIP) has been an interactive subject chosen by most studies. The increase of using Real Time Applications service such as (VoIP) is resulting in the high growth of Telecom and broadband field to meet the request of providing high quality of VoIP at any place, the prime Goal of this paper is to analyze and evaluate of an appropriate voice (CODEC) schemes depending on the Quality of service (QoS) of VoIP in Universal Mobile Telecommunication system (UMTS), network was implemented in OPNET Modeler 14.5. The quality is evaluated based on some QoS parameters such as Jitter , MOS, end-to-end delay and packet loss to investigate the performance of different codecs QoS scheme in UMTS VoIP network. The VoIP codecs used in the measurements of QoS are: G.711, G.723.1, GSM-FR and G.729A. Simulations showed that G.711 and GSM-FR are the best schemes provide best quality of voice. G.723.1 can be selected and use in UMTS depending on conditions. The results analyzed and the performance evaluated will give network Planners an opportunity to select the codec for VoIP performance enhancement which lead to the satisfactions of customers .*

I. Introduction

Day by day Internet technology has changed the Way and behaviours of people communicate With the rapid growth of wireless Packet-switched networks , sending data through the Internet Rather than the Public Switched Telephone Network (PSTN) has become a better option in terms of cost for users and Service providers, leading to huge growth of voice applications Over IP networks. With the new emerging set of mobile phones VoIP has become a factual standard for Voice applications in the Internet. Mobile phone users can make a voice/video call through the Internet anywhere anytime with better communication quality and less cost than PSTN. With the telecom industry moving towards the next generation Wireless networks which are going to provide high quality Service and higher down-link/up-link speed, VoIP continues to improve its QoS, mainly for long distance calls. This Improvement is going to impact businesses like Multinational companies, as well as the normal users to a great Extent than ever imagined . Deployment of Universal Mobile Telecommunications System (UMTS) as a part of 3G network. As a complete Network system, it provides wider coverage and high mobility to fulfil the user demands in any places including office, Home, urban and rural areas. UMTS supports packet-based Applications including real-time multimedia applications such As VoIP with a peak down-link data rate in this paper, we take VoIP as an application scenario to analyze and evaluate UMTS VoIP QoS using different Codecs, in Order to investigate how well those codecs cope with Real-time multimedia applications. This analysis will help identify the strengths and weaknesses of the codec in terms of QoS and can guide the applications to choose the best codec we have designed and implemented UMTS simulation module in OPNET and carried out extensive simulations to analyze the Mean Opinion Score (MOS), packet end-to-end delay, jitter And packet delay variation for different type of VoIP traffic with different Codec. Our simulation results show that UMTS has better QoS to support VoIP performance enhancement. The rest of the paper is organized as follows Section II briefly gives background VoIP. UMTS. Section III deals with the simulation setup used in OPNET for UMTS .Section IV evaluates and analyzes the Simulation results of the VoIP application running on UMTS. Section V discusses the related work. Finally In Section VI we conclude this paper.

II. VoIP

Within the context of VoIP communications, audio codecs play most crucial part in processing voice elements from analogue to digital and then from digital to analogue again. Though the performance of audio codecs is not the only driving and determining factor for the performance of any VoIP scenario, it can be well admitted that the performance of the audio codecs and their related features dominantly govern the degree of performance of any VoIP application. Thus, the performance of audio codecs is one of the most important aspects to consider while dealing with VoIP communication and its deployment. This paper presents the evaluate of VoIP under different audio codecs.

III. UMTS

UMTS is proposed to converge packet-switched and circuit switched networks. Its IP Multimedia System (IMS) is used for multimedia communications. IMS was originally defined by the Third Generation Partnership Project (3GPP) for the next generation mobile networking applications and uses SIP as the Signalling protocol. With the availability of UMTS, four service types have been proposed and incorporated into the QoS model of UMTS-Conversational class-for voice/video telephony, with low end-to-end delay and low jitter, two-way-Streaming class - for streaming video, with low jitter , One-way - Interactive class - for web browsing, with low loss/error - rate, two-way.

VI. VoIP and Codecs

Developers of voice-over-IP (VoIP) systems face many obstacles as they try to develop architectures that merge traditional POTS-based networks with packet networks. One of the biggest challenges to the successful development of these systems is quality of service (QoS). Unlike traditional IP systems, end users will demand that new voice-enabled packet systems deliver service at all times. Therefore, designers must produce architectures that deliver voice and data services to users when they want it. In order to deliver the quality voice services that users demand, VoIP system designers must tackle the packet loss problems that are inherent in traditional packet based networks. To do this, engineers are employing new coding techniques that bring a packet-loss protection mechanism to the VoIP architecture. The operators are forced to improve the quality of communication. This can be achieved by increasing the bandwidth and making the IP backhaul that fulfils the demand of the users at lower cost providing better QoS.

4.1- VoIP Codecs

Every system implementing VoIP uses an audio codec to compress the audio signals at one end and decompress the same at the other end. Although most of them are standardized, VoIP vendors implement proprietary codecs too. The type of Codec used is an important factor that affect the VoIP call quality as higher the compression, lesser the size of data to be transmitted over the other side. The Codecs also introduce a digitizing delay as each algorithm requires a certain amount of data to be buffered before it is processed. Some examples of popular standardized Codecs are listed in the table below:

Codec	Coding Algo	Sampling rate
GSM- FR	PRE-LTP	13 kbps
G.711	PCM	64 kbps
G.723.1	ACELP	5.3 kbps
G.729A	CS-ACELP	8 kbps

V. Topology and configuration

The deployed topology for the simulation environment is shown in Fig. 1. The network topology shows the networking elements used along with their interconnections. The model as in figure 1 comprises user equipments, node B and Radio Network Controller(RNC) which is connected to the packet switched network via Serving GPRS Support Node (SGSN) and Gateway GPRS Support Node (GGSN) which in turn is connected to the IP Network four users were used. SIP was used as the signalling protocol which required the network architecture to have SIP proxy server. Four audio codecs named G.711, G.729A, G.723.1and GSM-FR were chosen for the simulation. The configuration for the codecs used in the deployed VoIP scenario

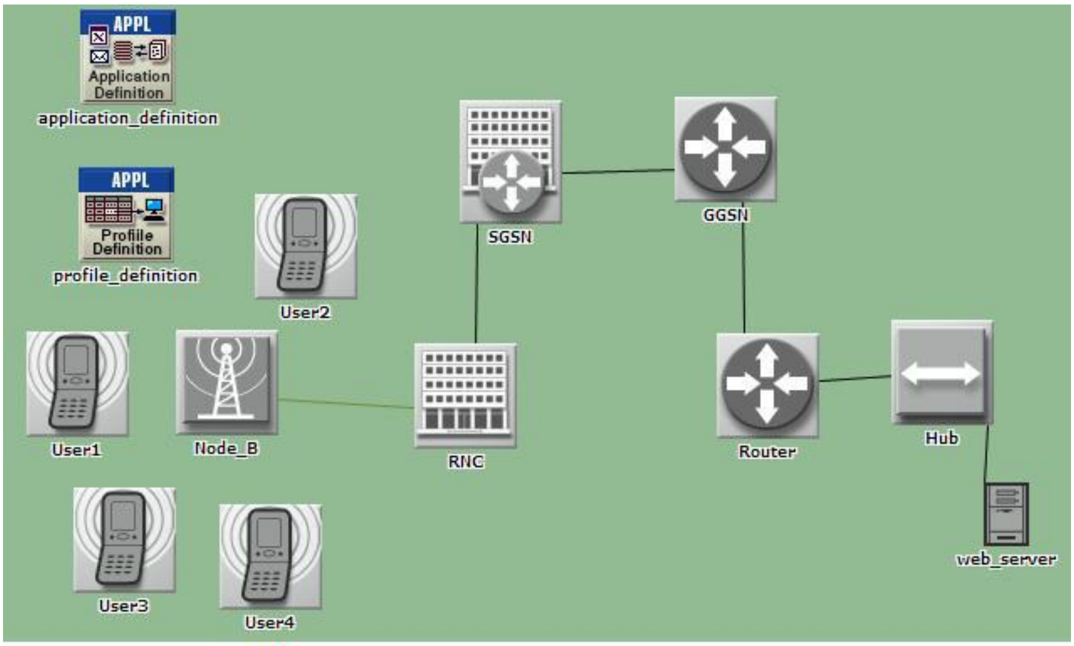

Figure 1: UMTS Model

VI. Performance Metrics

In this paper, I used the following four metrics to evaluate The performance of UMTS VoIP codecs in terms of end to end QoS.

6.1- Jitter

When the packets are sent from the Codec after compression, they are sent at a constant rate with equal spacing between them. But when they are received at the other end, the decompression algorithm also expects the packets to arrive with equal spacing between them and in the same order as they were sent. But since network imposes delays at packet level, the packets may arrive at different time intervals and they may not arrive in the same order, as they were sent.

6.2- MOS

The Mean Opinion Score (MOS), recommended by ITU-T in 1996, is the most widely Used subjective measure of voice quality. A MOS value is Normally obtained as an average opinion of quality based on Asking people to grade the quality of speech signals on the five point scale (Excellent =5; Good=4; Fair=3; Poor=2; Bad=1) under controlled conditions as set out in the ITU-T standard p.800.

6.3- Packet End-to-End Delay

end-to-end delay the time required for a packet to be traversed from source to destination in the network and is measured in seconds. Generally in VoIP network there are three types of delays occurring during the packet transverse. They are: sender delays when packets are transverse from source node network delay and receiver delay. D-Packet loss: Packet loss is another factor that can degrade the performance of VoIP. The packet loss can occur if packets are lost during the transmission or if the packets arrive too late to be useable by the receiving application.

6.4- Packet loss

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VII. Results and Analysis

In this paper same attributes and same simulation environment ,but with different codecs, The result that performance of each codec is evaluated in the network model depending on the QoS. The simulation was run for four different scenarios to collect QoS related statistics as discussed each scenario based statistics were for the audio codec used to configure the respective scenario. The collected statistics are presented in the following discussion in a compare contrast fashion to facilitate the understanding of the implications of a single factor or parameter on different codecs.

7.1- Jitter

Figure 2 illustrated the collected statistics as drawn show that G.711 was highest affected by jitter where the other three had least impact of jitter than that of G.711,mainly G723.1 was least affected .

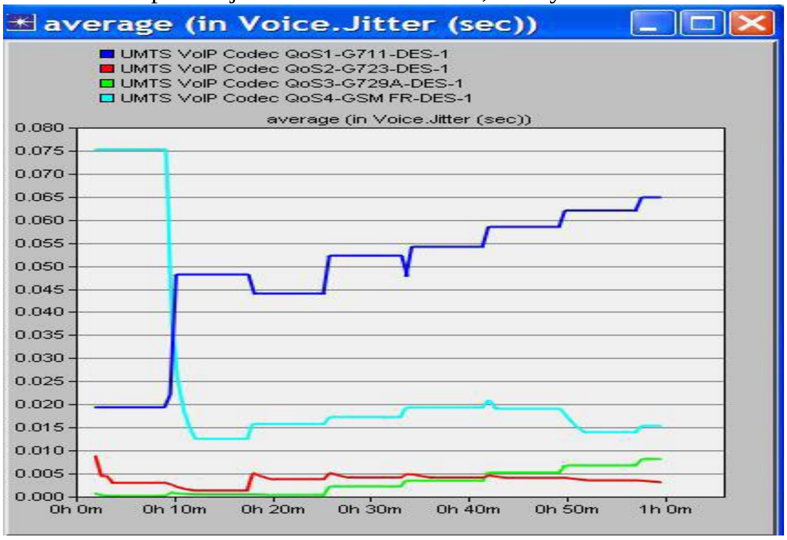

Figure 2: Jitter

7.2- MOS

The MOS value for the codecs collected from simulation is shown in Figure 3. The observation showed that GSM-FR had highest MOS value which was close to 2.8. G.729A had MOS value almost 2.4; G.711 had MOS value which was 2.11. And last lowest MOS value was G.723.1 which was 1.5.

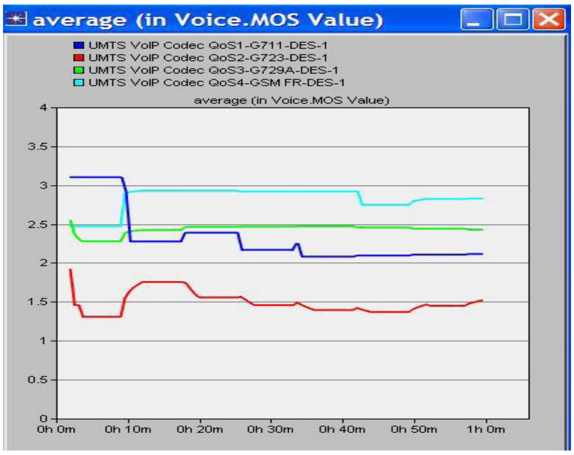

Figure 3: MOS

7.3- End-to End delay

Figure 4 portrays the End-to-End delay of the audio codecs. The result indicated a higher packet end-to-end delay for G.711 which was around 1.5 sec. The delay for G.729A was around 1.02 sec and the delay for G.723.1 was 1.05 sec .and GSM-FR was around 0.8 sec which was the best one.

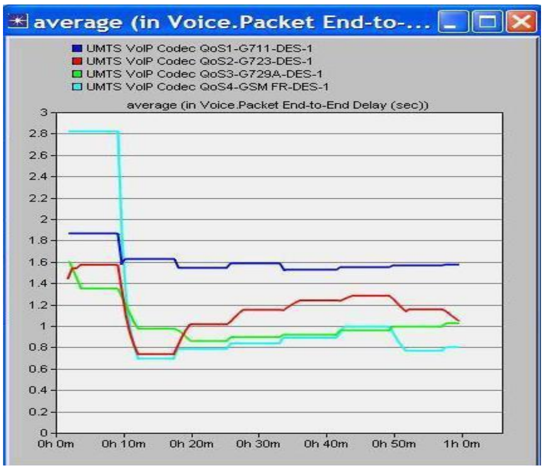

Figure 4: End to-End delay

7.4- Packets sent and received

Figure 5 shows packets send and received by the four voice codec schemes. The simulation results clearly indicate that, the packet loss of G.729A codec was lowest than the other codecs, while G723.1 was the highest one which is bad.

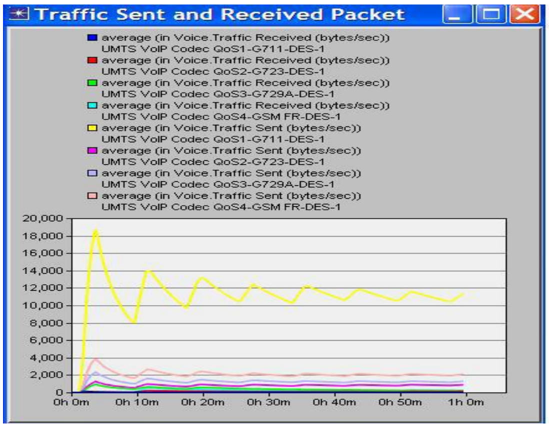

Figure 5: Packet sent and received

VIII. Discussion of Results

The consequences eventuated from the simulation results are summarized in below table which is a comparative QoS for the four audio codecs in UMTS VoIP. The imparity in Codecs QoS for the UMTS VoIP had varying level of efficiency. When we chose MOS value, G.723.1 had the lowest MOS value if compared to GSM-FR, G.729A and G.711. On the other hand, G.723.1 had better QoS and performance in jitter, while the other three codecs had comparatively higher jitter. While G.711 codec was packet delay variation lower, but had higher end-to-end delay which is a conclusive considering factor for deciding upon QoS. The results from this paper showed that the audio codecs were capable of performing fairly well in UMTS VoIP scenarios. If MOS and end-to-end delay are taken to be the most favour QoS factors to effect the decision on the most proper codec for VoIP in UMTS from the selected codecs, GSM-FR and G.729A could be chosen over G.711 and G.723.1 relying on the acquired results from simulation.

Type	G.711	G.723.1	G.729A	GSM-FR
Jitter(Sec)	0.065	0.003	0.008	0.015
MOS(Value)	2.1	1.5	2.4	2.8
End-to end delay(Sec)	1.6	1.1	1.0	0.8
delay Variation (Bytes/sec)	0.05	0.07	0.03	0.06

IX. Conclusion and Future Work

From the simulation result we can consider Performance of different VoIP codecs in UMTS is evaluated and analyzed using the OPNET Modeler. A variety of simulations are carried out to get the most effective and efficient results. On the basis of results attained, conclusion for the selection of VoIP codecs in UMTS is made. Depending on the results it is concluded that in UMTS network the best VoIP quality is given while using GSM-FR. The quality of G.723.1 codec is observed low as it is a low quality codec. Hence it can be used in all the networks depending on the environment and users density. The conclusions will be helpful and useful for the network planners and operators and also for the beginner researchers to further work on these issues. The VoIP Codecs QoS of the UMTS will be the main focus of the future work.

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