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A Performance Analysis of VoIP Traffic over Wireless LAN and Wan Using Different Codecs

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Abstract: ***A simulation model is presented to analyze and evaluate the performance of VoIP based integrated wireless LAN/WAN with taking into account various voice encoding schemes. Using OPNET Modeler software, The network model was simulated. Several parameters that show the QoS like MOS, jitter, end to end delay, traffic send and traffic received are calculated and analyzed in Wireless LAN/WAN scenarios. With Regard to this evaluation, Selection codecs G.729A observe the best choice for VoIP.***

Keywords: VoIP, Codecs, QoS

1. Introduction

Through the new years, there is an increasing trend in real-time voice communication using Internet protocol (IP). Voice over Internet Protocol (VoIP) is a technology that allows delivery of voice communications over the Internet or other packet switched networks rather than the classical Public Switched Telephone Network (PSTN). Considerable VoIP applications are obtainable on the internet: Yahoo messenger. Tango, Fiber and also Skype. All these applications give high quality and exact free calls. In VoIP, the analog voice signal from the transmitter is converted into the digital form before compressing and encoding it into a stream of IP packets for transmit to the receiver over IP network. At the receiving end, Digital to Analogue Converter (DAC) works on recreating the original analog voice signal after reassembling received IP packets in order and processing it.

There are numerous writers have worked on different Quality of Service (QoS) parameters using different service classes in different network types. A research was carried on different quality parameters influencing on the VOIP serving performance. The research indicates that these parameters of QoS are wanted to raise the performance of a VoIP. Investigators in [2], have compared the performance of the VoIP in both Ethernet LAN (802.3) and Wireless LAN (IEEE 802.11). They examine how VoIP performs in two different network setups and analyzes the results obtained using OPNET simulator. They also examine the optimization of IEEE 802.11e for QoS using the priorities to provide real-time service for VoIP. Various QoS parameters like throughput and average delay for VoIP using different protocols are analyzed in Ref. using OPNET simulator. Simulation results show that the Optimized Link State Routing protocol has better performance in terms of throughput and average delay. Similar analysis has been conducted into analyze the QoS of VoIP deployment over WiMAX network and compare

A comparison was carried out between different codecs (G.711, G.729A, and G.723.1) which are the most appropriate to improve QoS for VoIP.

2. VoIP Codecs

When used in VoIP, we usually send 3-6 G.729 frames in each packet. We do this because the overhead of packet headers (IP, UDP, and RTP together) is 40 bytes and we want to improve the ratio of "useful" information.

G.729 is a licensed codec. As far as end users are concerned, the easiest path to using it is to buy a hardware that implements it (be it a VoIP phone or gateway). In such case, the licensing fee has already been paid by the producer of the chip used in the device. A frequently used variant of G.729 is G.729a. It is wire-compatible with the original codec but has lower CPU requirements.

G.723.1

G.723.1 is a result of a competition that ITU announced with the aim to design a codec that would allow calls over 28.8 and 33 kbit/s modem links. There were two very good solutions and ITU decided to use them both. Because of that, we have two variants of G.723.1. They both operate on audio frames of 30 milliseconds (i.e. 240 samples), but the algorithms differ. The bitrate of the first variant is 6.4 kbit/s and the MOS is 3.9. The bitrate of the second variant is 5.3 kbit/s with MOS=3.7. The encoded frames for the two variants are 24 and 20 bytes long, respectively.

G.723.1 is a licensed codec, the last patent that covers it is expected to expire in 2014.

GSM 06.10

GSM 06.10 (also known as GSM Full Rate) is a codec designed by the European Telecommunications Standards Institute for use in the GSM mobile networks. This variant of the GSM codec can be freely used so you will often find it in open source VoIP applications. The codec operates on audio frames 20 milliseconds long (i.e. 160 samples) and it compresses each frame to 33 bytes, so the resulting bitrate is 13 kbit/s (to be precise, the encoded frame is exactly 32 and 1/2 byte, so 4 bits are unused in each frame). The codec's Mean Opinion Score is 3.7.

Speex

Speex is an open source patent-free codec designed by the Xiph.org Foundation. It is designed to work with sampling rates of 8 kHz, 16 kHz, and 32 kHz and can compress the audio signal to bitrates between 2 and 44 kbit/s. For use in VoIP telephony, the most usual choice is the 8 kHz (narrowband) variant.

iLBC

iLBC (internet Low Bit Rate Codec) is a free codec developed by Global IP Solutions (later acquired by Google). The codec is defined in RFC3951. With iLBC, you can choose to use either 20 ms or 30 ms frames and the resulting bitrate is 15.2 kbit/s and 13.33 kbit/s, respectively. Much like Speex and GSM 06.10, you will find iLBC in many open source VoIP applications.

Table 1: Features of the most common codecs: G.711, G.723.1, and G.729A

IUT-T Codec	Algorithm	Codec Delay (ms)	Bit Rate (Kbps)	Packets Per Second	IP Packet Size (bytes)	Comments
G.711	pulse-code modulation (PCM)	0 .375	64	100	120	Delivers precise speech transmission. Very low processor requirements
G.723.1	Multipulse maximum likelihood quantization/algebra c-code excited linear prediction	97.5	5.3	33	60	High compression with High-quality audio. Can use with dial-up. Lot of processor power
G.729A	Conjugate Structure – Algebraic Code Excited Linear Prediction (ACELP)	35	8	100	50	Excellent bandwidth utilization. Error tolerant

3. Network Model Description

In our simulation approach uses OPNET simulator for network modeling. OPNET simulator is an authoritative communication system simulator developed by OPNET Technologies.

This simulation model was run in three different scenarios to evaluate the difference in performance and to determine the best audio encoding schemes of utilizing VoIP over integrating Wireless LAN/WAN. All the scenarios follow the same structure and the same topology. The basic difference in all the scenarios is with the configuration of the corresponding application and the profiles. As one scenario is implemented with the codec G.711, G.723.1, and G.729.1. Various comparisons conducted to find the values of various parameters.

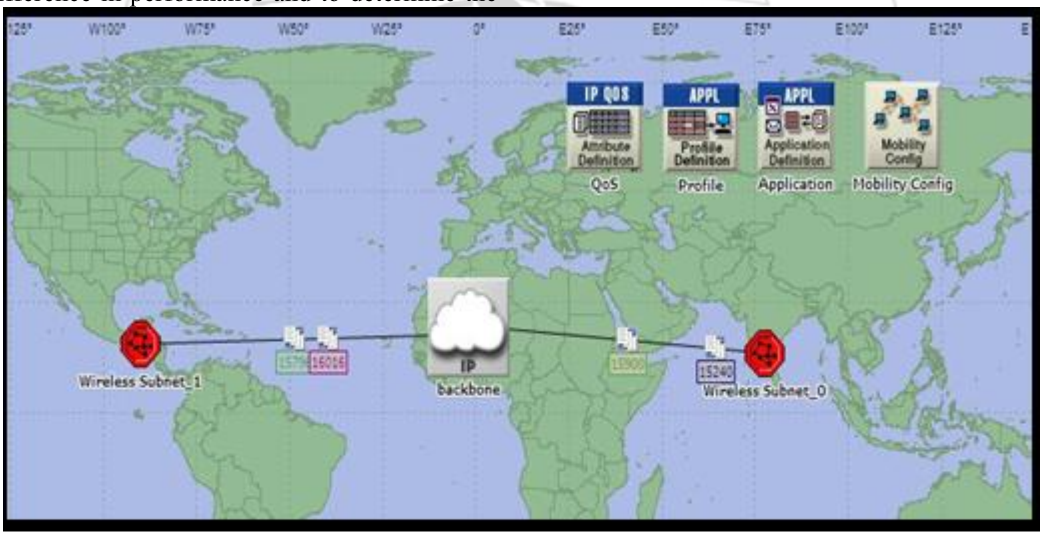

Figure 1: The Simulation Network Model

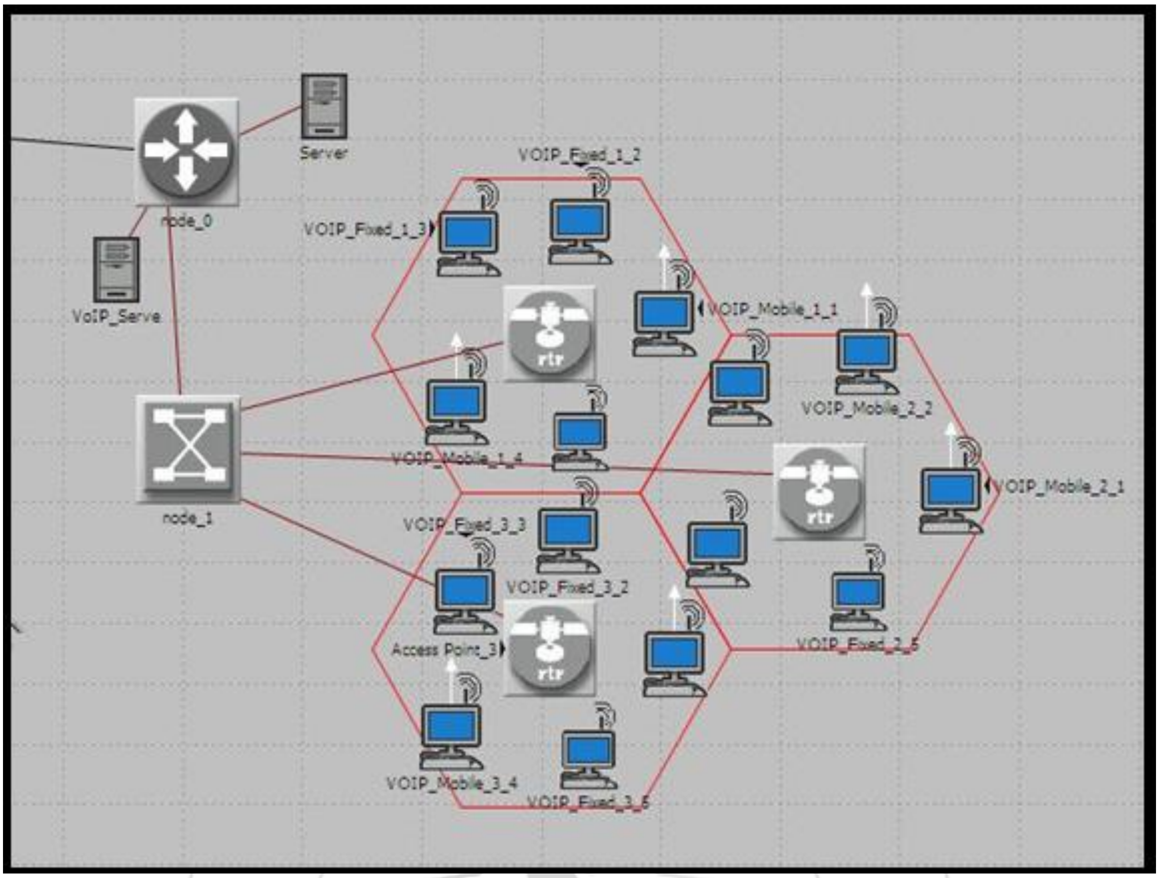

Figure 2: The wireless subnet_0

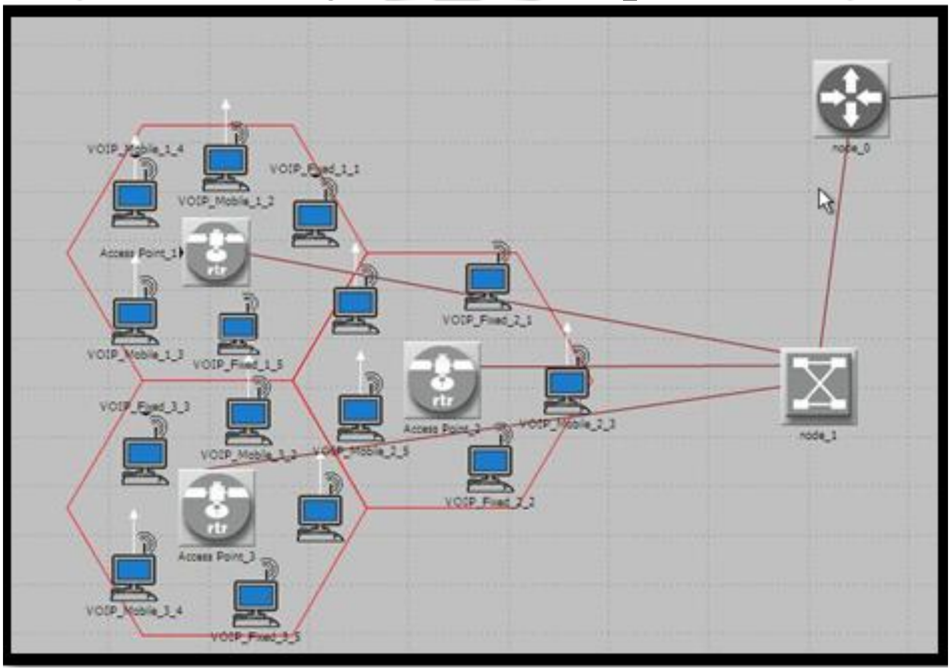

Figure 3: The wireless subnet_1

4. Basic Parameter

The Point to Point WAN was connected directly to the Ethernet Router in each subnet using a DS0 line. The model name of the link was 'point_to_point_link_adv'. The Wireless LAN environment used 1Gbps Ethernet cable for all the wired connections to the Router, Switch, and Servers. The OPNET model name for the Servers was 'ethernet_server'. Each Wireless LAN environment contains a Router and 2 fixed wireless workstations and 3 moving wireless workstations in each Router. All these wireless nodes use VoIP services. The Routers and workstations were using 802.11g running at a data rate of 54 Mbps. There was a total of 15 workstations in each subnet. The workstations were generating traffic across the WAN and the Wireless LAN environment to simulate a real office environment. The Routers OPNET model name was 'wlan_ethernet_slip4_adv. The fixed VoIP workstation model and the moving VoIP workstation model name were 'wlan_wksn_adv'

5. Results and Analysis

The following figures are obtained after collecting statistics by using OPNET Modeler simulation tool. Each figure shows a comparative picture of the three scenarios. All the three scenarios are using a different audio codec scheme such as G.911, G.923, and G.929. After successfully running the simulation, The result shows the impact of different codecs on different QoS parameters in a VoIP network. Following are the figures that show different QoS parameters like MOS, Voice packet end to end delay (sec), Voice jitter (sec), Voice traffic sends (packet/sec) and Voice traffic received (packet/sec).

The most widely used QoS metric in VoIP applications is MOS. The MOS value describes the voice perception quality. The average MOS value for the three codecs is represented in Figure 4. Codecs G. 711 and G. 729A have acceptable MOS values 3.685 and 3.067, respectively. On the other hand, the MOS value for G. 723.1 is 2.557 which indicates that the quality of service is poor if this codec used.

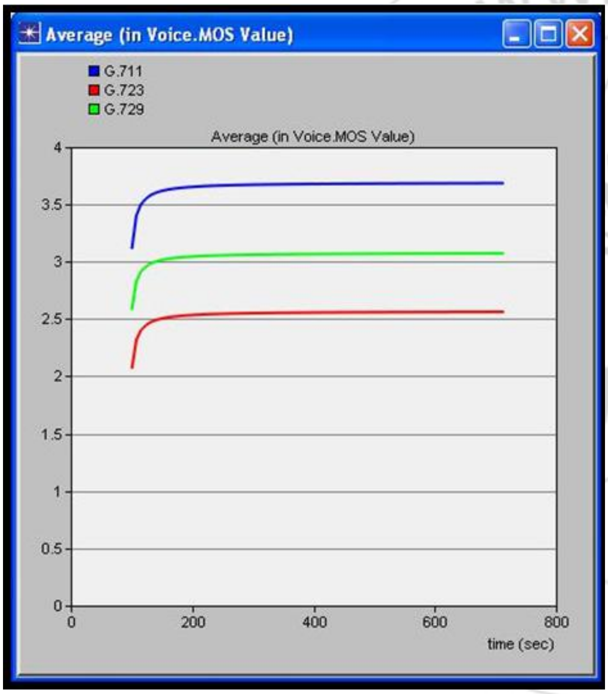

Figure 4: Average voice MOS under various codecs

Average end to end delay metric is shown in Figure 5. G.729A presents the best performance with respect to other codecs. These results are due to transfer rate and packet size. The low packets transfer and the larger packet size, the more time is required to process them. The relatively high transfer rate (8 kbps) and low packet size (20 bytes) for G.729A make G.729A the ideal codecs. Otherwise, G.723.1 and G.711 suffered the highest delay than G.729A for the reason that it has the lowest bit rate (5.3 kbps for G.723.1) and larger packet size (160 bytes for G.711). In turn, the end to end delay is increased with transfer rate and packet size.

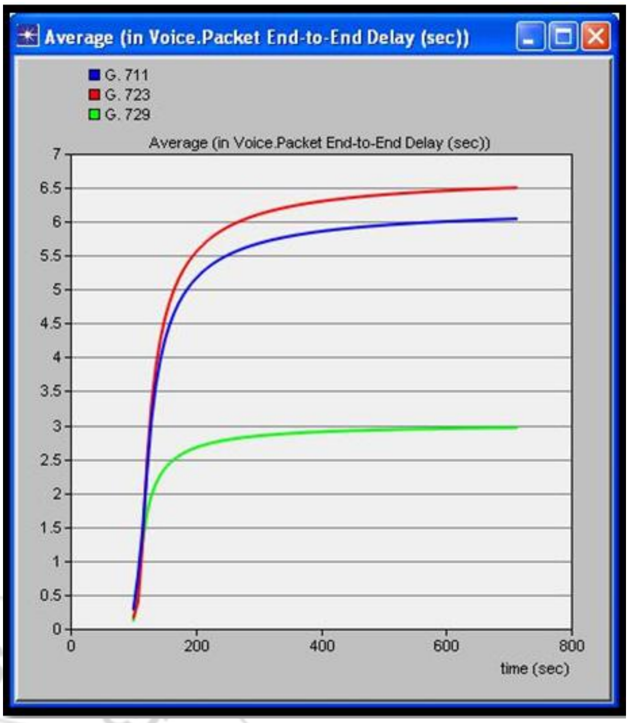

Figure 5: Average voice packet End to End delay (sec) under various audio codecs

Figure 6 describes the average voice jitter comparison using different codecs. From the Figure, the variation of the codec G.729A is minimum and approximately constant throughout the simulation. The average voice jitter variation in case of codec G.723.1 is higher than the other two codecs at the earlier time of simulation, but after some time it falls down. The jitte variation in case of G.711 lies between two other audio codecs. The voice jitter threshold for smooth communication in VoIP network is about 1 ms [11] so audio codec G.729A gives better results than audio codecs G.711 and G.723.1 respectively. So there is a high increase in jitter as audio codecs G.711 and G.723.1 are added to the network. This increase in voice jitter makes the voice difficult to understand due to arriving packets at the different time . The use audio codec G.729A will make the jitter less and best performance of VoIP application in Integrating Wireless LAN and WAN.

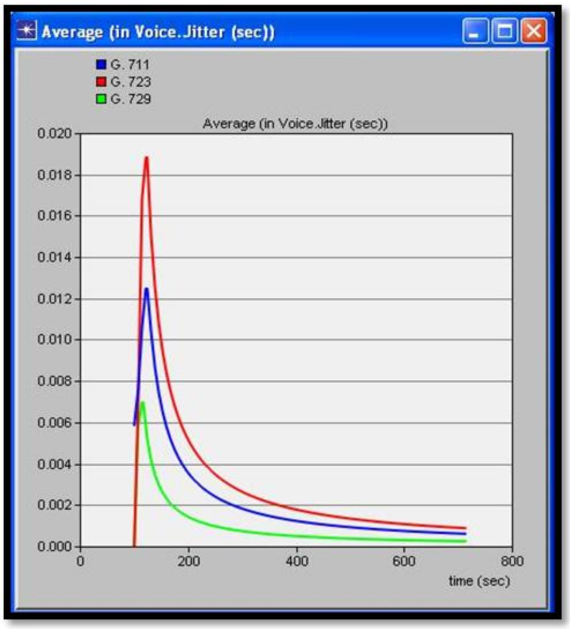

Figure 6: Average Jitter under various audio codecs

6. Conclusion

In our paper, analysis and evaluation of the QoS performance for VoIP traffic under various voice codecs were carried out. The use of codecs appropriately is very important in the implementation of VoIP to generate maximum QoS value. The result shows a selection of G.729A codec in a simulation gives a significant result for the performance of VoIP that codec G.729A has acceptable MOS value and less deviation of received to transmit packet as compared to G.711 and G.723.1 also average delay like end to end delay and Voice jitter is lesser in codec G.729A as compared to the other two referenced codecs.

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