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Performance Evaluation of VoIPv6 Service for Wireless Network Using OPNET Software

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Abstract— *Due to increase in the number of Internet users and the expansion of the newly applied services on the Internet such as Internet of Things (IoT) technology, the Internet Protocol version four (IPv4) has been developed to support this development of the Internet. Internet Protocol version six (IPv6) supports these capabilities by increasing the number of addresses available to connect large number of users to the global network. Deploying Voice over Internet Protocol (VoIP) service through wireless networks is a challenge for many network engineers. In this work different voice codec (coder-decoder) technologies (G.711, G.723 and G.729) are tested using Optimized Network Engineering Tool (OPNET) environment and their results are compared based on MOS (Mean Opinion Score), voice end to end delay and jitter parameters to investigate the quality of voice over IPv6 based wireless topology. Simulation results suggest that G.711 codec can be adopted to offer a best voice quality approach for VoIPv6 through Wi-Fi networks.*

Keywords— Codec, IPv6, OPNET, VoIPv6, Wi-Fi.

1. INTRODUCTION

In the previous couple of years, Internet has turned out to be mainstream and pervasive in our life. Internet is an IP-based network that works under the TCP/IP layering model. Internet protocol (IP) is a set of standards that characterizes how devices are impart over the network. The main function of the Internet protocol is to identify connected hosts dependent on their addresses in order to transfer information between them over the network. The first presentation of Internet protocol is Internet protocol version four (IPv4) was presented by the Internet Engineering Task Force (IETF). IPv4 has some limitations such as in addressing issue, it has just 32 bits for network addressing and appears insufficient for the future expanding Internet clients. Aside from this restriction of address issue, other restrictions for example, less security support and less mobility support will be issues in the future network condition. So IPv4 is probably not going to be adequate to work as Internet protocol in the future [1].

To fix of the issues in the Internet Protocol version four (IPv4), the Internet Engineering Task Force (IETF) produced another edition of the Internet Protocol (IP) known Internet Protocol version six (IPv6). IPv6 has more benefits especially in addressing, it provides very wide range of logical addressing space. It has auto-configuration and supports security and mobility technologies. Also IPv6 has simpler header than IPv4 and supports QoS technology which will be used in real time communications [2].

As of now, the Internet is works in blends of IPv4 based networks and IPv6 based networks. Later on IPv6 will be actualized in all Internet networks.

2. IPV6 OVERVIEW

Internet Protocol version six or IPv6 (also called IP new generation (IPng)) is the up and coming age of IP and it is the successor of Internet Protocol version four which is utilized today. IPv6 gives address space a lot bigger than the address space of IPv4, it went from four billion delivers to three hundred and forty trillion trillion trillion of extraordinary IP address [3]. IPv6 address is represented in eight sections, each section made of four hexadecimal digits separated by colons. IPv6 utilizes another configuration in which options are isolated from the base header and inserted, when needed, between the base header and the data. This streamlines accelerates the header handling time and increment the execution.

The expected growth of IPv6 networks will be increased greatly, in 2026 the IPv6 networks will be pleased instead of IPv4 networks all over the world. Fig. 1 shows the expected growth of IPv6 network traffic with respect to IPv4.

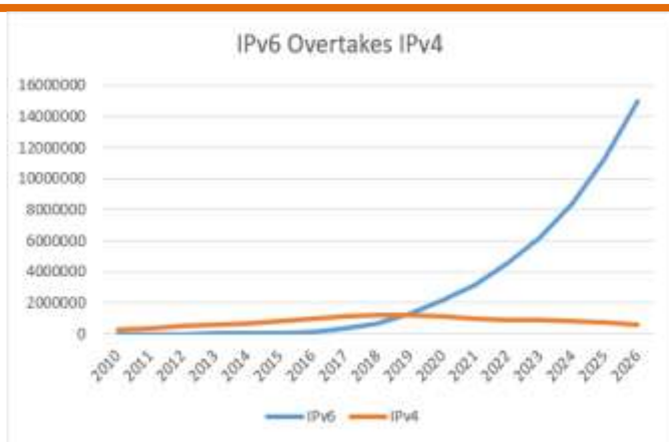

Fig. 1. *IPv6 growth [4]*

3. Wi-Fi Technology

Wireless Fidelity (Wi-Fi) is one of the Wireless Local Area Network (WLANs) technologies that are utilized in IP based networks. Wi-Fi technology is established by IEEE 802.11 standard and optimized for a high speed data rate wireless networks. It works in tow bands (2.4 and 5 GHz band) and gives maximum data transfer rate of (54 Mbps).

The orthogonal frequency division modulation (OFDM) system is utilized in Wi-Fi networks which gives data rates of (6, 9, 12, 18, 24, 36, 48, and 54 Mbps). Wi-Fi has two types of parts: wireless access point (Router) and wireless client terminal. The access point goes about as a bridge between fixed and wireless networks and the wireless client terminal is any end user device such as PC, workstation, mobile phone [5].

4. Voice Quality

To test the voice quality over IPv6 networks, standard parameters must be considered to evaluate the performance of VoIPv6 service. The major QoS parameters of VoIP traffic includes the choice of codec, Mean Opinion Score (MOS), voice end to end delay and delay variation (jitter). The following parameters are considered to test the quality of voice in IPv6 based network:

4.1 Voice Coder-Decoder (CODEC)

Voice codec (coder-decoder) technology was developed to change the analog voice signals into a set of digital pattern and after that reconstruct these patterns back into audible voice. There are various voice codec refined by ITU-T (International Telecommunication Union -Telecommunication) such as (G.711, G.723 and G.729). Each code has diverse payload size, data rate and perform differently. The table bellow shows the features of voice codecs.

Table 1: The Features of Voice Codecs [6]

Codec Type	Algorithm	Data rate (Kbps)	Payload size (Bytes)
G.711	PCM	64	160
G.723	ACELP/MP-MLQ	30	20/24
G.729	CS-CELP	10	20

4.2 Mean Opinion Score (MOS)

MOS provides an aggregate numerical measure of the quality of VoIPv6 networks. The MOS parameter is a recommendation of ITU, the following table shows the voice quality for MOS factor.

Table 2: MOS Rating Measurements [7]

MOS	Voice performance
5	Excellent
4	Good
3	Fair
2	Poor
1	Bad

4.3 End to End Delay

End to end delay is defined as the time taken for transition from the sender to the receiver. In IP based networks, the end to end delay means the elapsed time for a packet to pass from the source through the network to the destination. Delay range from (0-150ms) provides good voice quality over the network, the range from (150-300ms) provides acceptable voice quality and poor voice performance for delay above 300ms [8].

4.4 Jitter (Delay Variation)

Jitter is an important parameter when talking about voice quality in IPv6 networks. Jitter describes the variability of the network delay. When there is an acceptable network delay average but with high variability, there is a higher chance that packets will arrive out of order. For successful VoIPv6 Implementation, the jitter value should not exceed (20-50) milliseconds.

5. Network Model Simulation

To test the quality of the voice over IPv6 networks, proposed IPv6 Wi-Fi network is built and simulated using OPNET software program. The proposed network topology consists of one main switch connects three routers, and four wireless workstations are connected to each router to generate traffic load over the network. IPv6 is used as a network layer protocol to customize ipv6 address for each connected device. The following figure illustrates the network topology which is built and simulated in this work.

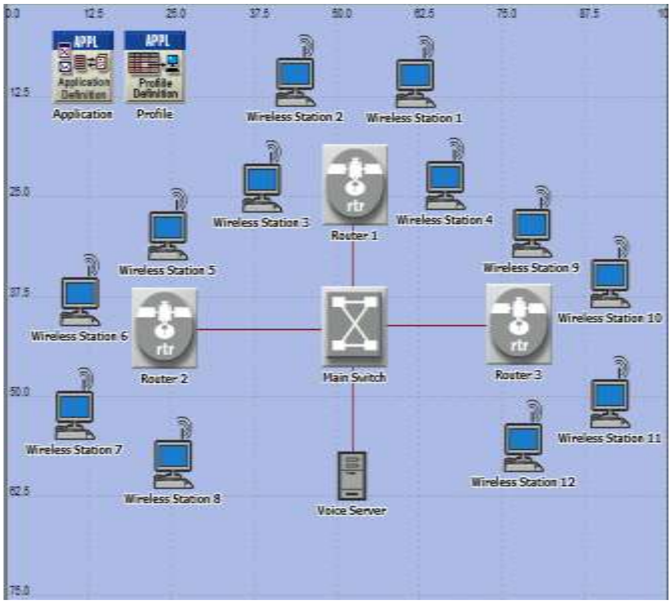

Fig. 2. *Network Topology*

Each network device should be configured to work properly over the simulated network.The table bellow shows the network parameters for the proposed simulated network.

Table 3: Network Simulation Parameters

Network parameter	**Value**
Topology	WiFi
Size	100*100 m^2
Network protocol	IPv6
Number of Routers	3
Number of wireless workstations	12
Number of servers	1 voice server
Number of layer2 switches	1 main switch
Wireless standard	802.11b
Wireless data rate	11Mbps
Supported application	Voice over IP service
Duration of simulation	10 minutes

Three scenarios were simulated, in each scenario different voice codec type was used. In the first scenario, G.711 voice codec was used as voice coding-decoding technique. In the second scenario, G.723 codec was used and in the third, G.729 codec was considered. Each scenario was simulated for ten minutes to examine the behavior of voice quality. Voice quality statistics were collected through the simulation time such as MOS, voice delay and jitter for all scenarios.

6. Simulation Results and Discussions

After running the simulation for ten minutes, obtained results for the three scenarios are collected and discussed in the following sections.

6.1 MOS Factor

G.711 codec type gives the best MOS value (3.7) with respect to G.723 and G.729 codec. Fig. 3 shows the MOS value for the three codec types.

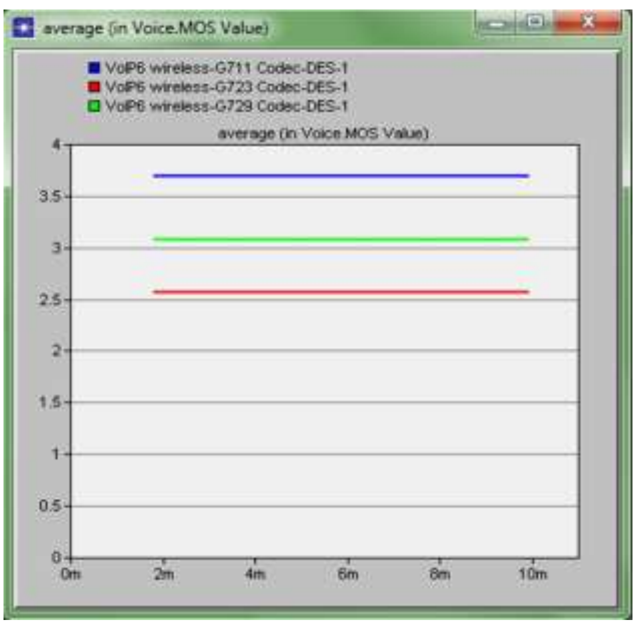

Fig. 3. *Average MOS Factor*

6.2 End to End Delay

The obtained results show that the voice end to end delay for different codec types is in the acceptable range. G.711 and G.729 give the minimum delay value (about 60 ms), the end to end delay for G.723 is (100 ms), fig. 4 shows the voice end to end delay values for the three codec types.

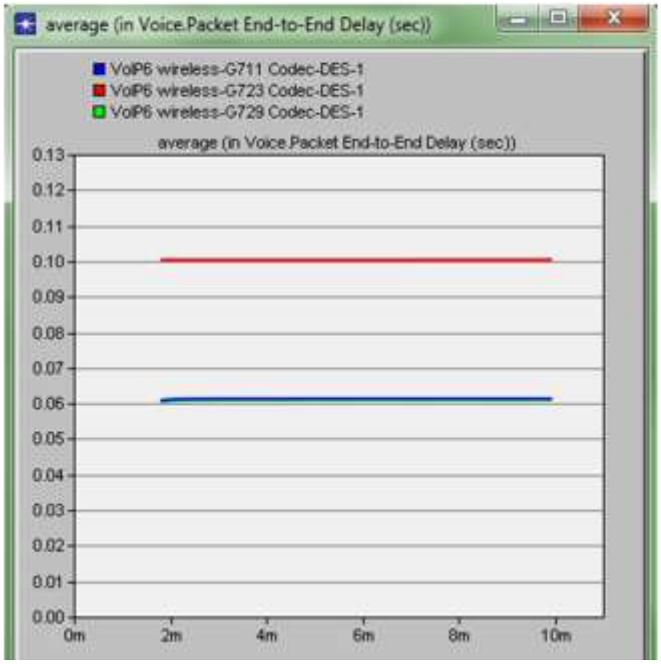

Fig. 4. *Average Voice End to End Delay*

6.3 Voice Jitter

The results for the voice jitter show that there is no Jitter effect for the three codec types over the simulated IPv6 based network as shown in fig. 5.

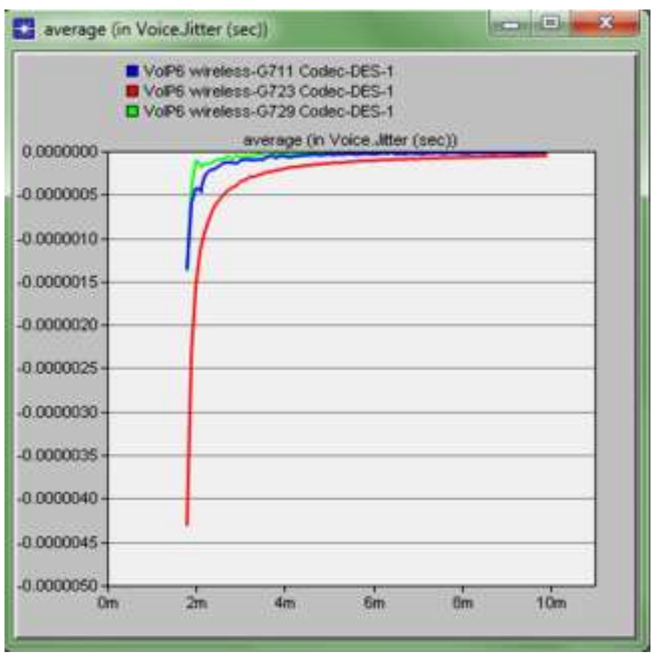

Fig. 5. Average Voice Jitter

7. Conclusion

In this work, we successfully evaluated the effects of various codec types (G.711, G.723 and G.729) for IPv6 Wi-Fi network. Various parameters such as MOS, delay, and delay variation are considered in our simulation to analyze the voice performance. Our results from the simulated network show that the best voice codec for IPv6 Wi-Fi network is G.711 codec type which gives the best voice quality. With respect to other codec types, G.711 gives the best MOS score (3.7) and minimum voice end to end delay (about 60ms) and without any jitter effect. Obtained results for G.711 audio codec are collected in the following table.

Table 4: G. 711 Codec results

Quality factor	Value	Test result
MOS	3.7	Good
End to end delay	61ms	Good
Jitter	0ms	Excellent

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