

End-to-End (e2e) Quality of Service (QoS) For IPv6 Video Streaming

Rosilah Hassan*, Rana Jabbar*

*Research Centre for Software Technology and Management (SOFTAM), Faculty of Information Science and Technology (FTSM)

Universiti Kebangsaan Malaysia, 43600 UKM Bangi Selangor, MALAYSIA

rosilah@ukm.edu.my, ranajabbar87@yahoo.com

Abstract— With continuous progress and advances in Internet Technology (IT) devices like smart mobiles, tablets and laptops, an increasing demand for real time applications have emerged. Reliable Quality of Service (QoS) mechanism over IPv6 for video streaming is required. New advances resulted in huge real time traffic on the internet and a reliable (QoS) mechanism has become an urgent need to meet the requirements of new technologies and network complexity. Furthermore, Internet Protocol (IP) which IPv4 address space has been exhausted and IPv6 address are now widely deployed. Real time video streaming demands required increased efforts to meet the requirements of end users. A new QoS approach for IPv6 video streaming traffic using flow Label field to control network parameters has been proposed. Proposed methods have been compared with different scenarios including best effort scenario and Differentiated Service (DiffServ) QoS approach. In this paper, we present the result of best effort scenario against different quality of streamed videos.

Keywords— Flow Label, Best effort, DiffServ, OPNET, IPv6 Header.

I. INTRODUCTION

Enormous number of ongoing activity application are currently accessible at versatile markets for download (i.e. You tube, sound cloud, live TV, and so on) [1]. These outcomes in gigantic constant movement on the web and a solid QoS instrument turned into a pressing need to meet the necessities of new innovations and system complexities [2]. IPv4 address space has been depleted and IPv6 address are currently generally conveyed [3]. Distinctive QoS approaches has been proposed to give a solid end to end (e2e) correspondence for constant activity, be that as it may, the system level QoS [4] approaches have given the most dependable QoS ensures over end to end correspondence [5]. A system solid QoS is proposed in this conceptual in view of the Flow Label field of the IPv6 header for conveying QoS for video spilling applications [6]. Existing QoS arrangements gives general answers for various sorts of movement for IPv4 Addressing space. New fields and capacities has been given by the new form of IP address space IPv6. Solid QoS instrument over IPv6 for video spilling is required with proceeded with requests on video gushing administrations to meet the necessities of end clients. Web incorporate huge activity sorts and expanding volumes of various sorts of information. Video Quality is additionally

expanded. This quality should be kept up everywhere throughout the way from source to goal. The primary objective of this exploration is to give a dependable end to end QoS over IPv6 for video spilling which keep up the Quality of various classes of recordings over the way from gushing source to goal. IPv6 20bit Flow mark field will be used to handle the principle three parameters which control the nature of video: least transmission capacity, least deferral and bundle misfortune.

The proposed system is recreated against best exertion and DiffServ instrument utilizing OPNET test system. Proposed technique has given a dependable level of QoS for top notch video. At the point when system get to be distinctly congested. Proposed instrument has kept up the postponement at required points of confinement where end to end defer has been diminished up to 4.7% than DiffServ system. Jitter likewise has been diminished to 15% superior to DiffServ system.

The remainder of this paper is organized as follows. In Section II presents the details of the simulation topology. In Sections III the used component of the OPNET simulator has been illustrated. In Sections IV, the performance evaluation metrics has been defined and presented. Results analysis and justification has been investigated in Section V, Section VI, and in the end in section VII the while conclusions of this paper is clarified.

II. NETWORK TOPOLOGY

To evaluate the proposed QoS protocol, a network topology has been built, as shown in the Figure 1. The topology contains three main Ethernet networks (Network 1, Network2, and Network 3) which are connected to Router2, Router3, and Router4 correspondingly. To simulate real traffic for QoS model two types of traffic has been generated:

- 1) FTP traffic is used to handle file sharing and file transferring between network clients.
- 2) Video Conferencing traffic which is one kind of video streaming application.

Router2, Router3, and Router4 are the corresponding gateways for network 1, network 2, and network 3. These gateways are connected to the Router 1 which is connected to both FTP server and Video streaming servers. These two

servers are responsible for generating the required FTP and video streaming traffic. Each topology networks includes 10 clients and these clients are divided into two equal groups to access both FTP and video streaming servers. Reliable server specification has been assigned for both FTP server and Streaming servers with the following specification:

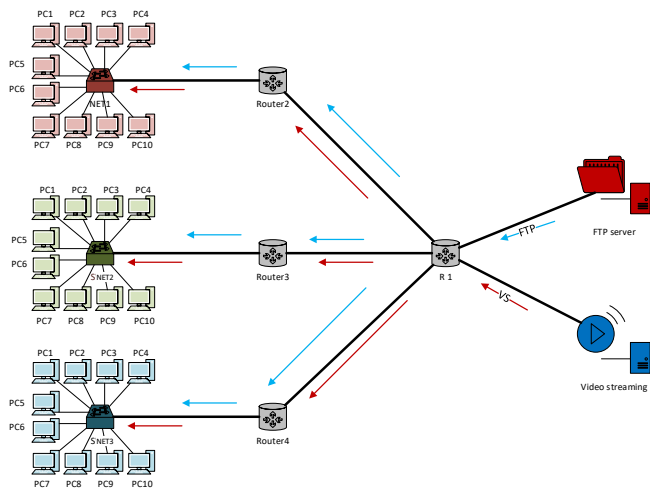


Figure 1. Network topology for simulation

TABLE 1. SERVERS SPECIFICATIONS

Servers Specifications	
Server Name	Dell PowerEdge 2900
CPU	2 Processors
Cores	4 cores/ Processors
Processor Speed	2666MHz
Operating System	Linux Operating System

Router 1 represents the bottleneck of the connection from networks clients and destination servers. Each network will be assigned to an IPv6 address bulk and the link which is connected to the router will be assigned to the gateway address. All Interfaces on the topology router will also be assigned to IPv6 address ranges which guarantee client to server connectivity. As shown in the routing table which has been used to route data between network nodes is OSPF_V3 routing protocol which supports IPv6 environment.

III. OPNET NETWORK COMPONENT

OPNET modeler[7] provides reliable models which can be used to simulate the system behaviour in an efficient manner, OPNET is a huge and powerful simulator which provides the ability to simulate different networks with different types of protocols. OPNET is a commercial simulator but an educational license is free for educational purposes. OPNET include different components which can be used to build different network topologies, these components include server, stations, networks, switches, hubs, routers and a group of different links types[8].

A. Network Simulation parameters:

The simulated topology uses a simulation time of 100 second, two different type of traffic will be generated which are

FTP traffic and video streaming traffic with the following specifications:

- 1) **FTP traffic:** This traffic will be generated from the FTP server which is connected to the main gateway router. The FTP traffic is assigned to high load.
- 2) **Video Streaming:** Different level of video conferencing qualities will be used to generate different load on the network as illustrated in TABLE 2.

TABLE 2. VIDEO TRAFFIC QUALITY SPECIFICATIONS

item	Video Quality	resolution	frames / sec
1	Low Resolution Video	120*128	10
2	Medium Resolution Video	240*128	15
3	High Quality Video	240*128	30

In this scenario, the network topology will run without any QoS configuration the traffic generated from FTP server and video streaming traffic will be treated the same way. No delay, bandwidth or packet loss will be guaranteed. Different video quality is used to confirm the results. As shown in the figure below. The IPv6 header will not be modified and all traffic will use the traditional IPv6 header without any updates. The traditional IPv6 will be used as shown in the figure below, there is no updates to any of the IPv6 header which related to the QoS including both traffic class and Flow label fields three different level of video streaming will be sent as shown in video streaming traffic table with different traffic rate. Both FTP traffic and video streaming traffic will compete for network resources, no bandwidth limitation, delay or packet loss are guaranteed. As we can see all traffic will be treated the same way by the generated servers and route paths routers. Data is delivered in the best effort model.

IV. EVALUATION METRICS

To evaluate the experimental procedure different metrics[9] will be measured including:

A. Video Streaming Throughput

Throughput refers to the number of streamed packets that pass successfully from streaming server to destination clients during a specific period of time. Throughput can be calculated by summation of the total number of packets that have been successfully sent to destination over the simulation time. Throughput is measured as an indicator for good performance where its value is increased when the number of the streaming videos packets delivery is increased. Throughput is represented in term of bits per second (bit/s or bps). Throughput can be calculated mathematically using the following equation.

$$\text{Throughput} = \frac{\text{No. of successfully pkts} * \text{Pkt size} * 8}{\text{Total simulation time}}$$

B. Video streaming delay variation

Delay variation is the arrival time's variation for a packet stream which has at least some known packet arrival times, it is

also referred to as jitter [10]. This metric is very critical for the Quality of streaming video where high jitter values can lead to anything from multiple performance problems like lip-sync or packets loss which results from buffer overflow or underflow. Jitter is reported and measured in milliseconds or even in nanoseconds.

C. Video streaming end to end delay

The third metric is the end-to-end delay which represents the average time which is consumed by each video streaming packets to traverse the trajectory from video streaming source to its destination client. e2e delay involve buffering delay, processing delay for route discovery and routers forwarding delay. e2e delay is a significant indicator for the performance of the network where small values e2e delay represent better network performance. To calculate the average e2e delay, the following equation can be used,

$$T_{E2E} = \frac{(T_R - T_S)}{\text{Total number of packets}}$$

Where, T_{E2E} is the Average End to End Delay, T_R is video streaming packets received time at destination node, T_S is video streaming packets sent time from source node.

V. SIMULATION RESULTS

A. Video Streaming Throughput

As shown in Figure 2, the number of bytes has been increased while increasing the quality of streamed video. With low resolution (LQ) quality video, the throughput of video streaming is about 2Mbps, but when the video quality changed to high quality (HQ) the throughput has been increased up to 7Mbps. On the other hand, VCR traffic which has the highest streaming quality the throughput has been increased up to 13 Mbps and the media of 100 Mbps and network became overloaded.

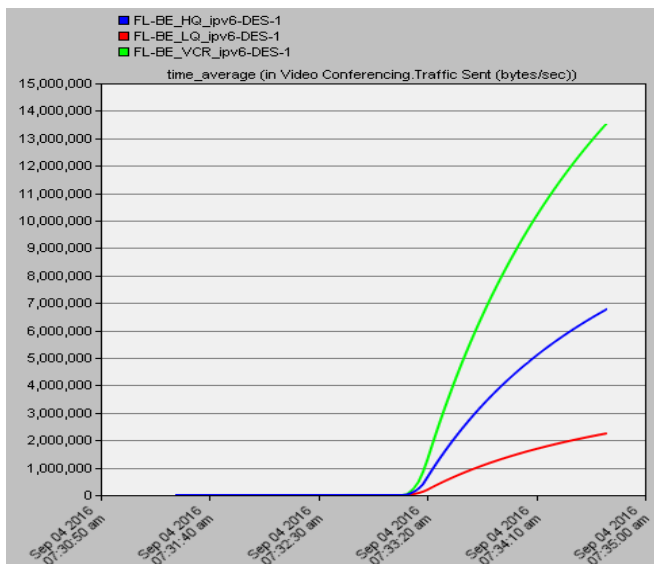


Figure 2. Time average Video streaming throughput for three different video qualities

B. Average end to end Delay

As we can see in figure3, when the quality of streaming video increased, the network load became more congested where the end to end delay is increased. For low quality videos, the average end to end delay is about 10ms, however it increases up to 200ms for the high-quality video. In the last part of the figure the end to end delay has been exponentially increased up to 8 second.

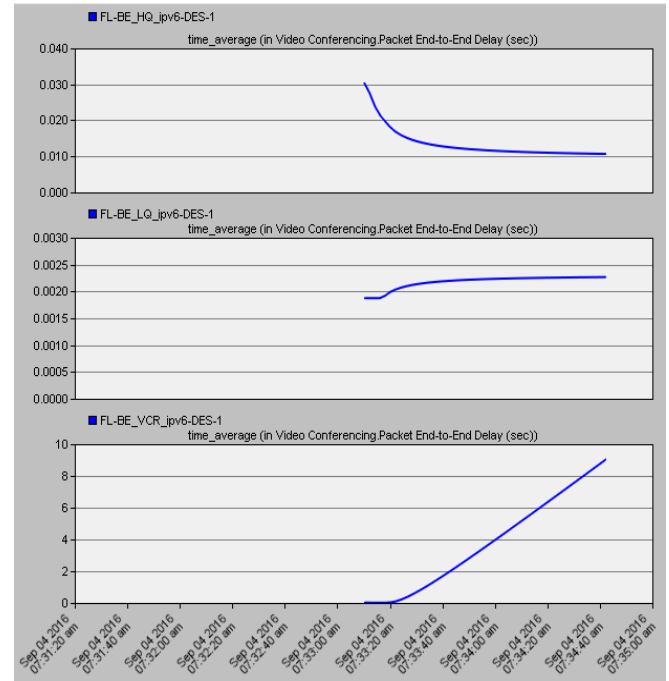


Figure 3. Time average end to end (e2e) delay for LQ, HQ and VCR video streaming

C. Delay Variation (Jitter)

Jitter values are very small for low Quality video streaming packets where it doesn't pass the value of 1 ns, however it has increased up to 8,0ms when video quality is high resolution. On the other hand, jitter for VCR has been massively increased. The value of jitter has exponentially increased to 14 seconds where the network became congested.

VI. RESULT ANALYSIS

As illustrated in the previous chapter the quality of video mainly affected the network performance. When the quality of video increase the size of the streamed video is increased. Increasing traffic volume make the network congested which results in high end to end delay and high jitter. As we can see in the streaming video throughput the size of traffic reach 12Mbps where 100Mbps Ethernet links become fully loaded. This high load traffic causes high end to end delay for video streaming which reaches up to 8 seconds and a video packet jitter up to 14 seconds. This performance is very bad for video streaming applications and doesn't meet its requirement. Based on this results and increasing video quality and application over networks and the internet, a reliable QoS mechanism is required which can maintain the quality of service parameters guaranteed all over the path from source to destination. In the

next step, DiffServ approach and our proposed mechanism, will be implemented and results will be compared against this best effort scenario to illustrate the significance of our proposed work.

VII. CONCLUSION

In this paper, we present the effect of increasing the quality of video streaming on the network infrastructure and results shows that increasing qualities results in increasing load on networks which can result in degraded performance including high end to end delay and jitter values. Proposed mechanism provides reliable end to end QoS for video streaming application since the QoS parameters is guaranteed over all network nodes from source to destination. QoS has been achieved without extra configuration or bandwidth overhead. The computing overhead has been minimized where routers can read directly the values of traffic requirements from the flow label field of the IPv6 header.

ACKNOWLEDGMENT

The authors would like to acknowledge the assistance provided by the Network and Communication Technology (NCT) Research Group, FTSM, UKM in providing facilities throughout the research. This project is partially supported under the Fundamental Research Grant Scheme FRGS/1/2015/ICT03/UKM/02/2.

REFERENCES

- [1] O. J. S. Parra, A. P. Rios, and G. L. Rubio, "Quality of Service over IPV6 and IPV4," in *Wireless Communications, Networking and Mobile Computing (WiCOM), 2011 7th International Conference on*, 2011, pp. 1-4.
- [2] S. Giordano, "Advanced QoS provisioning in IP networks the European premium IP projects ,*Communications Magazine, IEEE*, pp. 30 - 36.
- [3] M. M. Yousafzai, N. E. Othman, and R. Hassan, "Toward IPv4 to IPv6 migration within a campus network," *Journal of Theoretical and Applied Information Technology*, vol. 77, pp. 209-217, 2015.
- [4] O. M. Daoudeyeh and R. Hassan, "The necessity of integrating security as a QoS parameter in Mobile Ad Hoc Networks," *Research Journal of Applied Sciences*, vol. 9, pp. 466-473, 2014.
- [5] J. Williams, "Troubleshooting IP Video Quality of Service," *JDSU*, 2005.
- [6] M. M. Yousafzai, R. Hassan, N. E. Othman, S. S. Hasan, and U. K. Ali, "Performance evaluation of IPv6 test-bed network at UKM," in *Engineering Technology and Technopreneuship (ICE2T), 2014 4th International Conference on*, 2014, pp. 63-66.
- [7] *Opnet Simulator*. Available: <http://www.riverbed.com/products/performance-management-control/opnet.html>
- [8] O. Modeler, "OPNET Technologies Inc," ed, 2009.
- [9] L. Zhang, L. Zheng, and K. S. Ngee, "Effect of delay and delay jitter on voice/video over IP," *Computer Communications*, vol. 25, pp. 863-873, 2002.
- [10] J. G. Apostolopoulos, W.-t. Tan, and S. J. Wee, "Video streaming: Concepts, algorithms, and systems," *HP Laboratories, report HPL-2002-260*, 2002.



Rosilah Hassan received PhD in mobile communication from University of Strathclyde, United Kingdom in May 2008. She obtained Master of Electrical Engineering in Computer and Communication from Universiti Kebangsaan Malaysia, Malaysia in 1999. Her first degree is BSc. in Electronic Engineering from Hanyang University, South Korea in 1997. She is currently an Associate Professor in Universiti Kebangsaan Malaysia, Malaysia in School of Computer science, Faculty of Information Science and Technology. She has been serving Universiti Kebangsaan Malaysia since 1997. Dr. Rosilah is also a members of IEEE and IET. She was a member of program committees and reviewers in 2011 4th International Conference on Computer Science and Information Technology (ICCSIT 2011), Chengdu, China and 2010 International Symposium in Information Technology (ITSim), Kuala Lumpur, Malaysia, and also an editor for ICACTION Transactions on Advanced Communications Technology.



Rana Jabbar is still studying master in University Kebangsaan Malaysia (UKM) . Her specialisation is Computer Science (Network Technology) .She enrolled for master in (UKM) in 2014. She obtained on bachelor's degree in Computer science from Al-Rafidain university in Baghdad-Iraq in 2009. Her reserch in master was about enhancing the quality of video streaming using IPv6 header flow lable field.