QoS PERFORMANCE EVALUATION OF VIDEO STREAMING ACROSS IPv6

Efiyeseimokumo Sample Ikeremo
University of Africa, Toru-Orua, Faculty of Basic and Applied Science
Department of Computer Science
ikeremo.efiyeseimokumo@uat.edu.ng, 08039375701

Mc-Kelly Tamunotena Pepple
Federal Polytechnic of Oil and Gas Bonny
Mckelly2014@gmail.com

Abstract
The emerging internet video technologies provide the broadcasting industries a platform to support video streaming over the internet for viewers using the Internet Protocol version 4 (IPv4). Sequel to address shortage in IPv4, the Internet Protocol version 6 (IPv6) was invented to solve the IP addressing problem prevalent in IPv4. Meanwhile, IPv6 has not been fully deployed and it is rarely known in video streaming. This paper evaluates the Quality of Service (QoS) performance metrics of video streaming across IPv6 such as delay variation, end-to-end delay and packet loss. In this study, the network models investigated are compatible with industrial standards and the results are validated for a scenario 1. The Network model designed involved servers, workstations, switches and routers, and other technologies for video streaming, with the key objective of examining the QoS performance metrics and the impact of QoS modifications on various parameters, and the total impact on the quality of video content transmitted across the IPV6 network. During simulation and to understand the varying effect of video frame rates and the bandwidth, the OPNET software is used in the simulation of the network model. Results of the simulation show the effect of varying the different parameters of the video frame rate and bandwidth of the network; the lesser the value of the QoS performance metrics, the better the quality of the video streamed. The results obtained quite agree with the ITU (Y.1541) and British (BS EN50132-5-1:2011) standards for video streaming.

Key words: End-To-End Delay, Delay Variation, Frame Rate, Ipv6, Packet Loss, Type of Service.

INTRODUCTION
The Internet protocol has many utilities for transporting different signals such as data, voice, text and video etc., and this paper will evaluate the performance metrics of Quality of Service of the Internet Protocol (IP) video transmission in Internet Protocol version 6(IPv6) network because IPv6 has not been fully deployed in video streaming.

1.1 Background
In 1890s the Lumiere brothers and Edison introduced cinematography which led to motion image, and after fifteen years of the concept, television was also launched and it became popular for individual and public usages. Ampex introduced the videotape recorder which

stock picture streams that enable broadcaster to choose live broadcast or to play pre-recorded programs from tape. (Austerberry, 2002).

The streaming media was introduced by Rod Glaser in 1994, a former vice president of multimedia and consumer systems at Microsoft. The Realnetworks were the first to streamed live broadcasting between the Yankees and Seattle mariner baseball match on 5th September 1995, two years later streaming video technology came on board and was originated by the same company, but Realnetworks witness challenges from the arrivals of Xing and Vivo, which announced their video solutions.

The streaming media market were populated with these four companies: Realnetwork (Realplayer), Microsoft (Windows media series), Apple Computer (QuickTime) and Macromedia (flash XM), and these four products can be found in PC, Desktop and laptop (Follansbee, 2004). With competitive innovation in the streaming media field, five companies: Netflix, Amazon prime, Hulu plus, Vudu and iTunes are presently considered the most popular and offer good selection and are simple to use on wide variety of devices (Warren, 2011 and Smith, 2012).

In 2000 Macromedia obtained by Adobe systems gradually but certainly crumbled windows media market share in the influence of its progressively famous flash player which integrated interactivity, web 2.0 and streaming media. Also, the issue of bandwidth, scalability still continuous, but large quantities of network traffic were HTTP-based and content delivery network (CDN) with User datagram protocol (UDP) to transport content to large of number viewers. (Zambelli, 2013).

The evolution of these new technologies has enabled large number of viewers to access and view very high quality video content by using of internet protocol (IP) networks connectivity. Video transport which has very large scale of application in area such as education, sport, entertainment and peer to peer communication. There are several fields that video content is presently applied. The internet protocol (IP) technologies progresses with innovative evolution that has altered the internet protocol technologies for video transport which also lead to more of communication methods such as video delivery that which actually involved video streaming over internet protocol (IP) network, in the place of IP networks. This advancement in technology will incorporate the entire phases of the formulation of video content and delivery, starting from the video camera and the end public viewer’s video equipment. This modern flow of advancement in this aspect shows a fast growth of multimedia communication over internet Protocol (IP). There is a high rush of audio and video on demand over the internet which started in early 1990s from CNN.com and MSNBC.com and were presently the main collection of recent music and video merchandises from Napster to iTunes. MTV's promote utility, that will be increasing speedily to a whole movie delivery on request, like sites such as YouTube which helps to upload and distribute personal user-generated video content and produces internet video producers and consumers that made publishing videos on request which is accessible to any person with access to internet connection with the aid of home video, iPod or mobile phone (Van Der Schaar 2007, Kurose and Ross, 2013). Furthermore the network application like Skype, Google Talk, and QQ (mostly use in china) which enable individual to make not only internet telephony over the internet protocol network but it also improve the internet telephony with video as well as multi-person or group video conferencing and text message can be send at the same time.

2.0 VIDEO FORMATS.

The various video formats presently today as well as the difference among them are basically reliable on some operational constrain such as specification needs for video quality. Steadily innovation of technology which results to the evolution of modern as well as more efficient standards (Simpson, 2008, Angelides and Harry, 2011). Schreer et al 2013 suggested that TV industry can use universal video production format that will unify the world and support large range of multimedia operation. For example, the 3 D video format require a specific glass to view (Seungsin et al., 2012), but the different standards are only effective in particular format and have not supported these unify operation of broadcasting applications, (Pulipaka et al., 2013, Gabriellini and Mrak, 2013). Emergence of current video compression technologies are the same like MPEG4 AVC makes it possible to compress real time industrial high-quality video into moderate bandwidth of about 1Mbits/s that will be easily sent across within fixed capacity (Van Der Schaar, 2007).

STANDARD DEFINITION: The Standard definition is generally apply in digital coding format that mostly used for Television production, the luminance as well as chrominance components of the video signals which are sampled at 13.5MHz and 6.7MHz respectively to generate a 4:2:2 Y: Cr:Cb component signal. The way at which sampled digital signal are based on the chosen video frame rate that can neighbour than sub-QCIF, QCIF, CIF either

4CIF as well as the use of 30Hz for, an NTSC signal also 25Hz for both PAL/SECAM signal (Poynton, 2012).

HIGH DEFINITION: Generally High-definition video signal are in two major formats video streaming methods that include 720p and 1080i, which are gettable at distinctive format rates. It is the choice of the broadcasters in several countries to use either 1080i or 720p whereas others permit only one format. Countries that adopt NTSC make usage of 1080-line interlaced video at the frame rate of 30 as well as 720-line progressively scanned video at the frame rate of 59.94, others countries which uses PAL/SECAM function or apply at 1080i of 25 frames rate per second as well as 720p at 50 frame rate per second. Standard definition video signal applies the aspect ratio of 4:3 whereas High-definition video signal operates at the aspect ratio of 16:9 (El-Hajjar and Hanzo, 2013, Poynton, 2012, Simpson, 2008).

2.1 Video Streaming and Technologies

Video streaming is the technique of transmitting video content, it can be live either pre-recorded to a viewing gadget over a transport channel for quick display, and it entails the concurrent sending of audio and video to the viewer for playback over IP network. it is suitable in various business application aims which are communication, education, Training, Advertising and customer support; for this usages video producer are will to control several areas of user experience and control also administer their individual private video streaming equipment. The video producers have two cardinal methods in which for hosting content for streaming delivery, the primary hosting of content on mainly web-based utilities like YouTube, this method, the content provider has little restricted regulation over the outcome of how the content is been change, stocked or dispatched. The other method is private video streaming which the content provider fully regulates and agree on the conversion formats, the delivery techniques as well as how viewers need to experience video content (Simpson, 2008). Video streaming over the IP network include compressing of digital video signal as well as dividing it into IP packets (IP packet is any data unit that is transmitted using IP between the source and destination on the Internet). The compression of video content which decrease the quality of the video content that decreasing the set of bits applied to reveal individual pixel in the image (particularly the use of some pattern of MPEG or numbers of encoding formats like Real video either windows media). The uncompressed IP can be streamed over the data rate which meets the rate of the video. The packets will be admitted by applying a player software either particular or assigned circuit in the set top box that can convert the image for displaying. Hence regard to video streaming, on time dispatch of a video content is the utmost priority.

The content has to reach at the displaying gadget precisely before it required to be streaming to operate better. The period of content delivery may be affected with several factor which mostly cause by network traffic. The network should have the capacity to transport the video to the viewer without any loss of packet either time delay.

2.3 Streaming System Structure

Video streaming on internet protocol (IP) networks demands good amount of framework to be in place. A set of these very essential frameworks includes the streaming server as well as the media player software.

An illustration of the comprising elements for streaming system structure is depicted in Figure 2.1, the streaming server is vital for content depository transfer to any single displaying gadget as well as the media player software is important for conversion of the image on the screen of the user gadget, the architecture also consist of some different essential components which includes the content preparation station and the delivery network (Austeberry, 2005, Simpson, 2008).

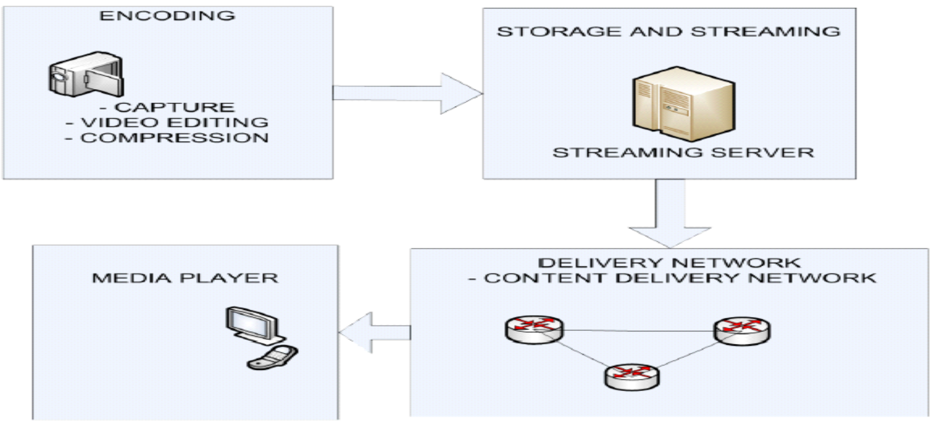

Figure 2.1 Component of the Video Streaming Structure.

2.4 Compression

Compression is the method of decreasing the number of bits applied to display any pixel in the image (Mozammil, Zakariya and Inamullah, 2012). The video signal of the standard definition television camera which is 166Mbit/s video rate is not possible to be circulated over many of the IP network due to the accessible bandwidth as well as thus it needs to decreased to lower rate like 30bit/s for delivery to dial-up modems, Figure 2.2 display relative size of encoded media.

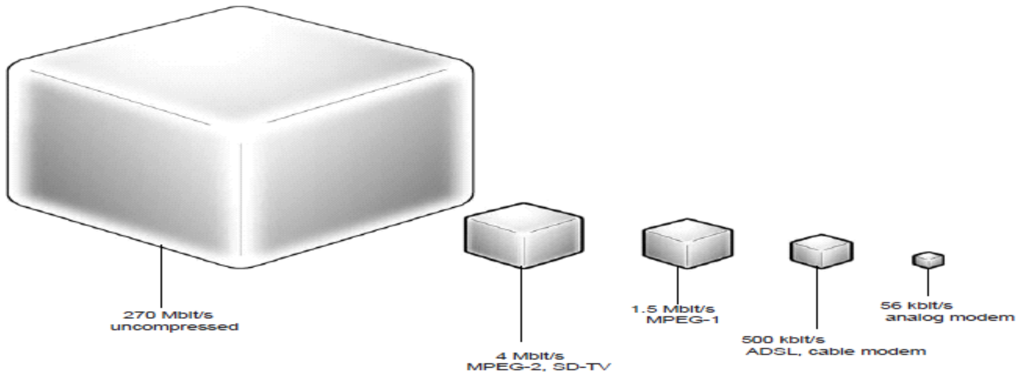

Figure 2.2 Relative file size (Austeberry, 2005)

2.5 Video Compression Technologies

The international Telecommunication Union developed standards known as codec for videophone as well as video conferencing were H.261. It applies progressive scan as well as 4:2:0 sampling support data rates starting from 64Kbit/s toward 2Mbit/s which depended upon CIF (Common Intermediate format) that functions at 30 frames per second containing a picture size of 352 pixels wide by 288 lines. Although presently, there are several various codec that are grouped in any of the three categories; includes international standards, proprietary as well as open source as shown in Figure 2.4.

The evolution of several multimedia usages which required for additional adjustable coding platform that brings about MPEG 4 standard which gives mass different functioning profiles also reducing of the bandwidth for the same video when contrast to MPEG-2, as well as allow the delivery of HD Video over the internet and IPTV networks. MPEG-4 integrates variability in the size of the macro-blocks and fractal compression that outcomes decrease the requirement bandwidth for Transmission (Hosik et al., 2010).

Compression format	ISO/IEC number Issue date	Target bandwidth bit/s	Typical resolution pixels	Application
H.261	1988–1990	384 k–2 M	176 * 144 or 352 * 288	Video Streaming Low Delay
H.263	1992	28.8 k–768 k	128 * 96 to 720 *480	Video Streaming
MPEG-1	11172 1993	400 k–1.5 M	352 * 288	CD-ROM
MPEG-2, MP@ML	13818 1994	1.5 M–15 M	720 * 480	Broadcast television, DVD
MPEG-4	14496 1998	28.8 k–500 k	176 * 144 or 352 * 288	Fixed and mobile web
AVC, H.264	14496–10 2002			General purpose

Figure 2.4 Compression Standards (Austeberry, 2005).

2.6 Stream Serving Using the Streaming Server

The streaming serving includes the storing, recovery and sharing of media streams to audiences. These streams may be either unicast or multicast as well it based on various mechanisms. There are other methods of streaming; true streaming which required no storing due to video signal transported across the IP network at a data rate that meets the rate of the video also the simultaneously displaying of the transported video to the watcher at the far terminal.

2.6.1 Streaming Server

The streaming server is applied to circulating media streams to end gadget by making packets for any outgoing stream in real-time with any IP packet that consist of both source and destination IP address. If a server is transmitting normal RTP packets, the audio and video streams which are transfer as distinctive packet stream toward divided sequence (Dilger, 2010, Van Der Schaar and Chou, 2007).

The main characteristic of streaming server is the generation and streaming of video file to end user gadget over networks, with distinctive and dynamic network associated speed, which any of the speed needs distinctive play back speed (Hwang, 2009). Lesser bandwidth causes network congestion which the streaming server need to accept the network speed modifying the video bit rate while network congestion alternates. This is very essential to restrain suspending streams also assure better video quality.

2.6.2 Internet Protocol

"The internet protocol provides a uniform addressing so that computers on one network can communicate with computers on the distant network" (Simpson, 2008). The effectiveness of internet protocol supports various operation, which simply the application of simultaneous activities upon a computer, which made communication for various types of computer. The internet protocol improves efficient transmission which dividing transferred along datagram that have being transferred into small packet networks. Internet datagram are transmitted from source address to the destination address among various interconnection of networks, and its function to deliver the datagram. The general operation of IP includes addressing and fragmentation

- **Addressing**: - IP addresses of the header applied in sending internet datagram to their destination with chosen path is known as routing

- **Fragmentation**: - specific fields of the header applied toward fragmenting and

3.0 QUALITY OF SERVICE (QOS)

The existence of the outbreak in multimedia communication across IP, whereas demonstrated in the growth of video and audio required across the internet as its usages in associate communications. The solution to supplies featured traffic kinds require mechanism, that categorize and manage various network completely is known as Quality of service QoS (Mark, 2006, Van Der Schaar and Chou, 2007). Because these usages can only be improved by QoS traffic flow controlling capabilities, facilitating the processing of these sensitive flows of packets in the core routers, and integrating QoS techniques to separate packet flows of various classes (Fgee et al., 2005, I-Hsuan, et al., 2013, Perea, 2008,).

3.2 IP QoS Architectures

There are different QoS mechanisms which may be accomplished in a network to generate eminent guarantee of terminal-to-terminal video transport, they normally stated by distinctive QoS architectures, which supplies various mechanism to fulfill specific effective video streaming performance that involves policing dropping, marking, Queuing, and Classification.

3.2 Type of Service (TOS)

The mission of the type of service field had to establish when to remit packets required for particular utilities form IP network, not deciding on queue experience by video streaming packets either, it detects the route from the IP network that the video packet can pass completely, few numbers of protocol included in type of service routing for instance the use of TOS field value to decide specific IP packet with destination IP address for routing. And there is no difference between IPV6 and IPV4 in terms of QoS (Marchese et al., 2006, Ping and Desheng 2010). There are used in exactly same way.

3.2.1 Integrated Service

TOS model consists of few deficient impacts, which integrated service model is created toward solving these deficient impacts that improves the efficiency of its usages such as video and voice as the outcomes of its eminent requirement (Evans and Filsfils, 2007, Marchese et al., 2006, Ping and Desheng, 2010). Intserv accomplish resource guarantee by applying the idea of per-flow resource reservation demanding usages to create a reservation easily, then transmit packet to network involving the resource reservation protocol (RSVP), whereby the end-to-end signaling protocol applied in fixing up Intserv stipulations (Mark, 2006). It consists of two service model which choice can be made: -

- The Guaranteed service mainly created fundamental for the usages such as video streaming due to massive intention of quality from user, it supplies active acceptance control to the network as well network queuing is regulated.

- The load control service also supplies utilities like low loaded best effort network (Wang, 2001).

3.2.2 Differentiated Service

As the result of scalability restriction related to the integrated service model which led to the creation of differentiated service model employed the integration as follows (Marchese et al., 2006, Ping and Desheng, 2010): -

-Edge function: packet classification and traffic conditioning: the entering edge of the network, arriving packets are marked most specific the differentiated service (DS) field in the IPV4 or IPV6 packet is fix to few values.

- DSCP marking: - packets would neither pre-marked applying the Diffserv code point (DSCP) among the DS field from IP packet header nor marked upon entrance of diffserv network, shown a specific class of a member packet, whereas Diffserv nodes particularly require carry-out easily classification applying the DSCP in way to decide class of packet.

- Per-hop behaviour: - eventually marking from the edge of the network, the diffserv network confirms which of the per-class organizing as well queuing control mechanism used toward the traffic classes which depended upon the DSCP marking in a way to assure per-class service quality differentiation (Evans and Filsfils, 2007).

4.0 METHODOLOGY

The overview of the methodology used in building the network model to run different simulation scenarios as designed, to use it to evaluation the performance metric of Quality of Service in line with industrial standards. A brief discussion of how the network model was created and configured and the OPNET software; the video packets network interconnection bandwidth and type of QoS parameters as follows: -

4.1 OPNET Modeller

OPNET was introduced in 1986 by OPNET Technologies Inc., it is a network simulation software that provides solution to the area of communications network including application performance management, Engineering, Planning research and development, it is object-oriented of GUI and with capability of simulating both explicit DES and hybrid simulation modes, which also assists co-simulation, parallel simulation, supports small and large networks (Lu and Yang, 2012, Sethi and Hnatyshin, 2012).

4.2 Hypothesis

Earlier, the IPv6 has been reported to exhibit similar throughput performance compared to IPv4 (BS EN 50132-5-1:2011). Both IPv4 and IPv6 have some similarities in the IP-header of the traffic class. For instance, if there are 8-bit traffic class field used to identify a flow in IPv4, then the QOS of IPV6 will be strictly equivalent to that of IPv4 (Marchese et al., 2006, Ping and Desheng 2010). In this work, it is anticipated that the IPV6 will provide similar QoS performance statistics with IPv4 in using IPv6 for video streaming.

4.3 Simulation Objectives

The objectives of the simulation is to examine and determine, the outcome value of the Quality of service (QoS) Performance metric (delay variation, end-to-end delay and packet loss) from various scenarios. With specify video frame parameters, bandwidth and the impact of type of service (TOS) of the QoS on the entire activities of the network. As a result of transferring video packets from server to client in IPV6 network. And to evaluate these results with BS EN 50132-5-1:2011 and ITU-T Y.1541 industrial standards. These standards validate the results that obtained from simulation to show how the quality of video is affected in video streaming. Based on the literature review of video, technologies and in section 3 QoS as well as TOS discussed. The following parameters are been considered to examined and investigated in this simulation.

- Video frame rate
- Video frame size
- Type of service (TOS)
- Network interconnection bandwidth
- Buffer size of Routers

4.4 Performance Metrics

The configuration of Discrete Event Simulation is to check set of different Statistics which show the Quality of the transported video stream.

- Packet loss: - this can be determine by the number of traffic sent and traffic received, packet loss illustrates the situation that packets are been transferred accurately from source and do not reach its destination, the underlying transport protocol UDP has no capability of retransmission of loss packet, thus, Quality video can only be obtained when packet loss is below 1%.(Namee, 2009).

- Packet end to end delay: - it is the variation of time between the video packets received by the destination as well the time its transferred from source. Video packet are very sensitive to end-to-end delay, and it is measured in range of millisecond (150ms, 200ms, 250ms and 300ms respectively (Cranley and Liam 2009, Perea, 2008, Van Der Schaar and Chou, 2007). Hence had to be kept very minimal to assure good Quality of video.

- Delay variation: - it is the fluctuation of latencies in packets within specifies data stream (Cranley and Liam 2009.)

4.5 Stages Performed for Simulation

To design a network model, the following stages are performed: -

- Application Definition: - it is a device in object palette to set attribute that define different application of the parameters of video size and frame rate.
- Profile Definition: - it is applied to specify each profile of any user in the network.
- TOS Definition: - it defines various classes of service and type of service in the network.

- IP QOS: - it is a network device in the object palette which is used to configure quality of service attribute of the network as TOS class.
- Designing of network topology by applying network node from the predefine devices in the object palette.
- Operation of Discrete Event simulation to determine selected individual DES statistic of various node results.
- Viewing and analyze Simulation Results.

4.6 Network Model Simulation

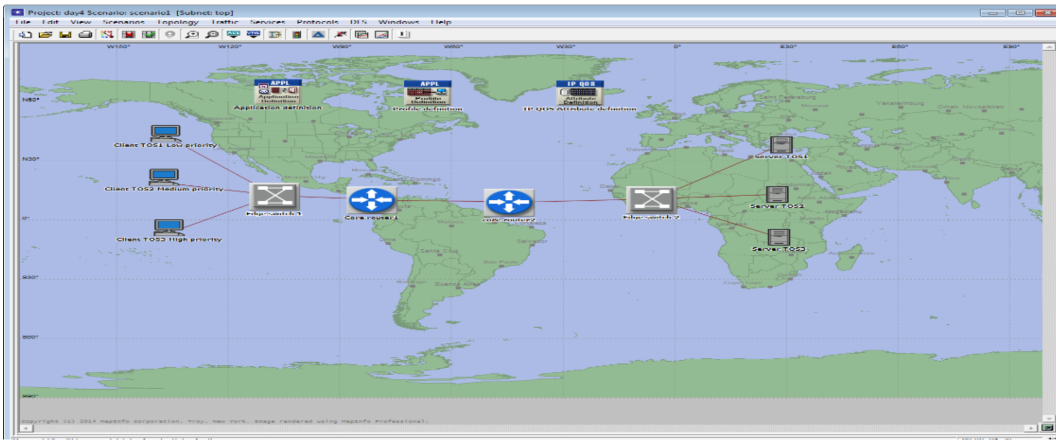

Figure A1.1: Network Model Topology

The Network model shown in Figure A1.1 was setup using the OPNET Modeller 16.1, to evaluate the Quality of Service (QoS) performance metrics of video streaming across IPv6. The model was simulated for scenario 1. These QoS performance metrics are packet delay variation (Jitter), and packet end-to-end delay (Latency) and packet loss (packet dropping). The parameters affect the quality of video when streaming. They are determined by the duration of delay and percentage of loss experienced by the traffic. Hence the value of these QoS performance metrics will determine whether poor or good quality video. The Network model in Figure A1.1 is built using Internet Protocol (IP) video transmission standards in accordance with British (EN 50132-5-1:2011) and ITU (Y.1541) standards. Each of these devices are obtained from the OPNET software standard application library to design the network model. These devices were used in real time network setup which include, servers, switches, Routers and link model (network cable) with workstations. These can be client (laptop, mobile phones, iPad, etc.) or personal computers. In the OPNET software, these devices are generally known as nodes while the cables are called link models. The workstation (client) sends a request to server and it starts to send the video packets to the client. The time it takes for the client to receive the data from the server is the end-to-end delay; the processing time of data to be transmitted is the delay variation. Also, the loss of data experienced while transporting the packet form the server to client is the packet loss (i.e. inability of packet or traffic to reach its destination either drop of packet as a results of congestion). Additionally, all clients and the servers communicate in this way.

The network model in Figure A1.1 also contained global model (Application definition node, Profile definition node and IP QoS Attribute definition node) that are used to define the applications of each of the network devices. Later, these applications were deployed to the clients and the servers, and then to define the profile which contain these applications. The profile describes the event methods of a user or group of users in terms of the applications used over a period of time. Applications are the predominant source of traffic in the network. The traffic generated by the applications load the network, makes request on the bandwidth and the underlying network technology, and create loads in the server. To control/manage congestion in the network (such as to avoid collision of packet in the model), policing is setup on how different devices will send traffics to each other through priority queuing rules.

It is the hierarchical Network Model Topology that composes of various network devices each shows as well simulating all parts of the video streaming designed architecture across IPv6 Network.

- The Routers display geographically with different core routers for routing video stream packets from the streaming server to the streaming client with a specify path which is accomplished by the interconnection of two routers that support the forwarding of video data packets.
- The streaming server consist of multimedia content which all the server generates video stream by various define parameters namely video frame size, video rate and type of service.
- The Network connection links simulates the features of a specific network layer and links within the client/streaming server as well as access layer consist of switches; links within switches and router, router and router showing distribution layer and core layer, the various layers featured with bandwidth.

4.7 Simulation Parameters

TOS; IPv6 header comprise diffserv code point (DSCP) in service of Traffic class (Mark, 2006,). The Traffic class is featured with a mutual meeting of the DSCP field and is used to set particular precedence or DSCP values. There are used identical as in IPv4. No difference

(Marchese et al 2006, Ping and Desheng, 2010). IP QOS categorization is applied in traffic class field of IPv6 to configured video packets by ascribed one of the three various TOS categories of diffserv code point (DSCP) below: -

- Background QOS(AF31)
- Excellent effort QOS(AF32)
- Streaming Multimedia QOS(AF41)

Quality of Service profile were designed dependent upon the IP QOS parameter of priority Queuing and the priority has been specified with TOS value of highest value has the high priority. The DSCP or the class value ascribed to the video packets from the streaming server decides the regulation of routing of the video packet by the router when transferring via IPv6 network. The specifications of the features of TOS classes are as follows: -

- Background QOS(AF31): It is recognized with the TOS value of 1 as well as the in-between routers within the routing path were configured to ascribe packet service labeled with TOS1(Background TOS) were also allocated to a Queue containing 20 packets, thus, the Queue cannot receive more than the defined packet any order that will be discarded.
- Excellent Effort QOS (AF32) It is recognized with the TOS value of 2 as well as the in-between routers within the routing path were configured to ascribe packet service labeled with TOS2(Excellent Effort TOS) were also allocated to a Queue containing 60 packets, thus, the Queue cannot receive more than the defined packet any order that will be discarded.
- Streaming Multimedia QOS(AF41) It is recognized with the TOS value of 4 as well as the in-between routers within the routing path were configured to ascribe packet service labeled with TOS3(Streaming Multimedia TOS) were also allocated to a Queue containing 80 packets, thus, the Queue cannot receive more than the defined packet any order that will be discarded.

The in-between routers were configured as priority Queuing profile with queues featured with distinctive priorities, were made to control packets with various TOS labels. The Queue priority are determined by the type of service (TOS) in a way that packets labeled as TOS1 (Background QOS) were ascribed to Queue 1; packets labeled as TOS2 (Excellent QOS) were ascribed as Queue 2; packets labeled as TOS 3 (Streaming multimedia) were ascribed as Queue 3.

The features of Queue:

- Queue 3 transfers packets as far as it has not finished.
- Queue 2 transfers packets if Queue 3 finishes
- Queue 1 transfer packets if all the both Queues finishes.

When network congestion occurs this specified categorization. Video packet that has the highest values conceives good delay values.

Video frame rate: The variations of the video frame rate are 10fps, 15fps and 30fps.

Video frame size: In a way to make QOS analyze very simple the video frame size was made constant.

Bandwidth: the network connection bandwidth links that selected from the object palette in OPNET are 10base-T (10 Mbps), 1000base-T (1 G bps), and 100base-TX (100 Mbps), respectively.

4.8 Simulated Scenario

The scenario1 were designed as shown in Figure A1.1. Which in line with BS EN 50132-5-1:2011 and ITU-T Y.1541 industrial standards in video streaming, to collect statistics of QOS Performance metric and determine and analyze the impact on the video quality as well as network parameters upon the outcome of QOS with various video frame rate, video frame size, Type of service (TOS) and the bandwidth. The scenario1 would consist of value of parameter stated above and single value of each parameters will be applied to examine the impact of the parameter on the Quality of service performance metrics such as end-to-end delay, delay variation and packet loss.

The servers are configured to produce label video packets with various TOS labeling as well individual client are configured based on TOS of QOS to receive video packet from corresponding TOS label server's illustration of the connection between the server and the client are: -

- Server TOS1 and client TOS1(Background TOS); server TOS1 generates video stream labeled TOS1 and the client TOS1 only received this labeled TOS1 packet from server

TOS1, the packets send within these nodes are handled by the in-between routers based on its TOS value as well as accepting service specially to packet in Queue 1.

- Server TOS2 and client TOS2(Excellent effort TOS); server TOS2 generates video stream labeled TOS2 and the client TOS2 only received this labeled TOS2 packet from server TOS2, the packets send within these nodes are handled by the in-between routers based on its TOS value as well as accepting service specially to packet in Queue 2.
- Server TOS3 and client TOS3(streaming media TOS); server TOS3 generates video stream labeled TOS3 and the client TOS3 only received this labeled TOS3 packet from server TOS3, the packets send within these nodes are handled by the in-between routers based on its TOS value as well as accepting service specially to packet in Queue 3.

5.0 RESULTS AND DISCUSSION

The results of simulation scenario1 with varying parameters on the effect of video Quality are as follows below:

5.1 Results

Scenario 1

Scenario 1 follows a conventional star topology IP network, where packets are sent from IPV6 source to the corresponding IPv6 destination through the IPv6 core domain, with the background Type of Service (ToS) of QOS. To examine the effect of low bandwidth on the Quality of service performance metric. Based on this parameters video frame size: 1500bytes, video frame rate 10fps and Bandwidth 10baseT. the step by steps configuration of the network model as in Figure A1.1.

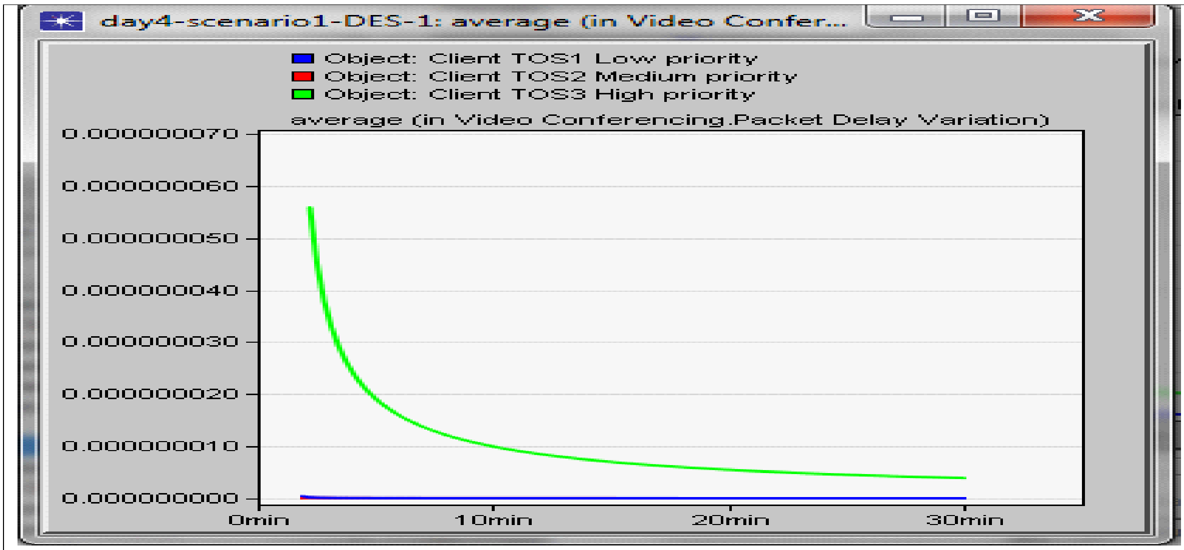

Figure 4.1: Packet Delay Variations.

The Figure 4.1 shows the delay variation of scenario 1 with the input parameter stated above, the y-axis represented delay variation in second and x- axis shows the simulation period.

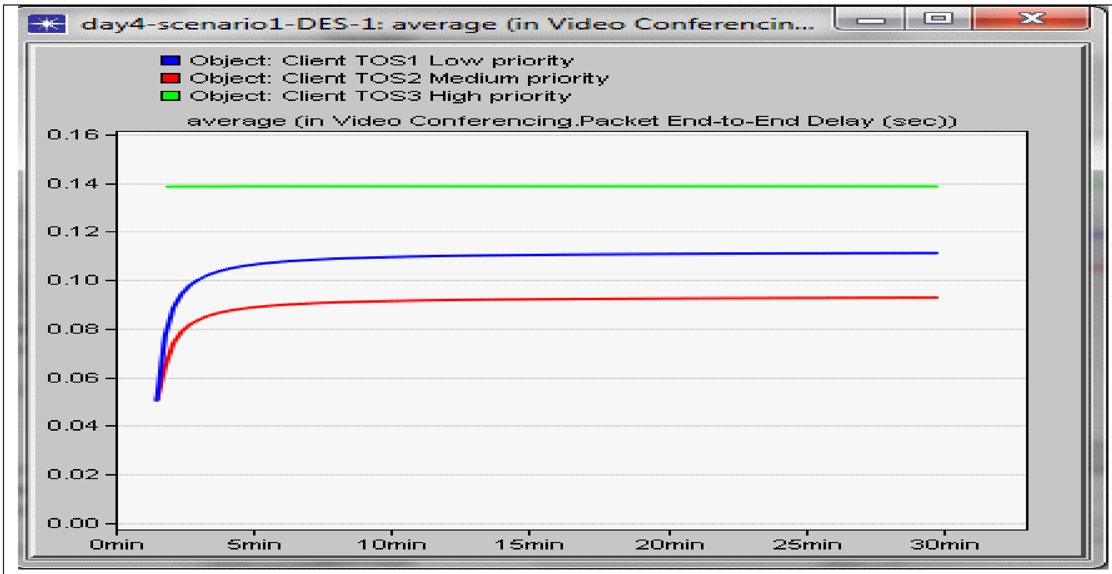

Figure 4.2: Packet End-to-End Delay.

The Figure 4.2 shows the End to End delay of scenario 1 with the input parameter stated above, the y-axis represented End to End delay in second and x- axis shows the simulation period.

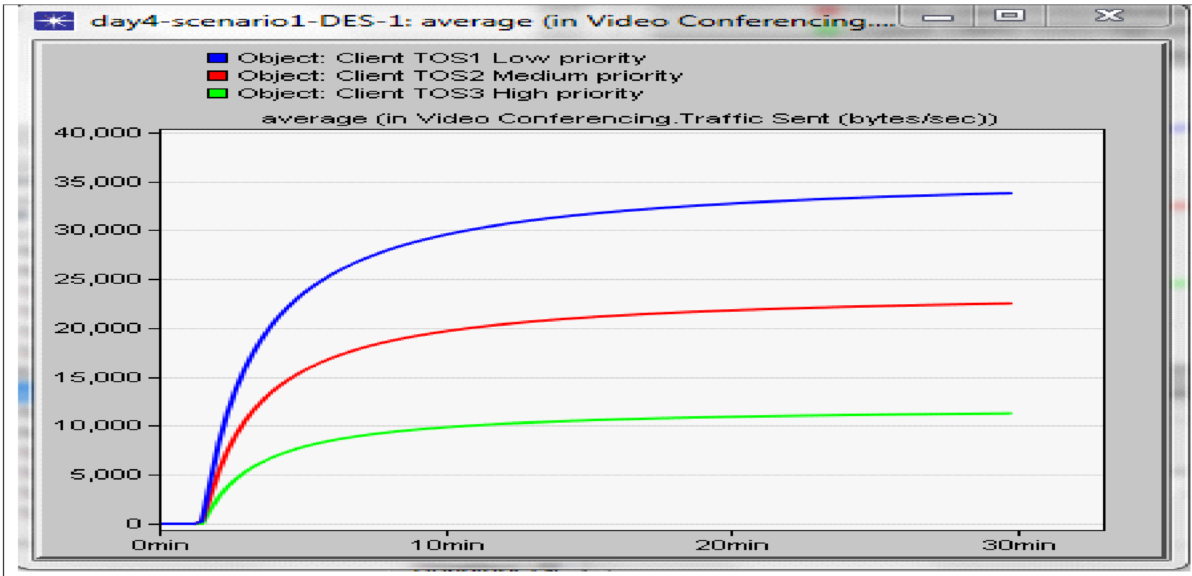

Figure 4.3: Traffic sent (bytes/sec)

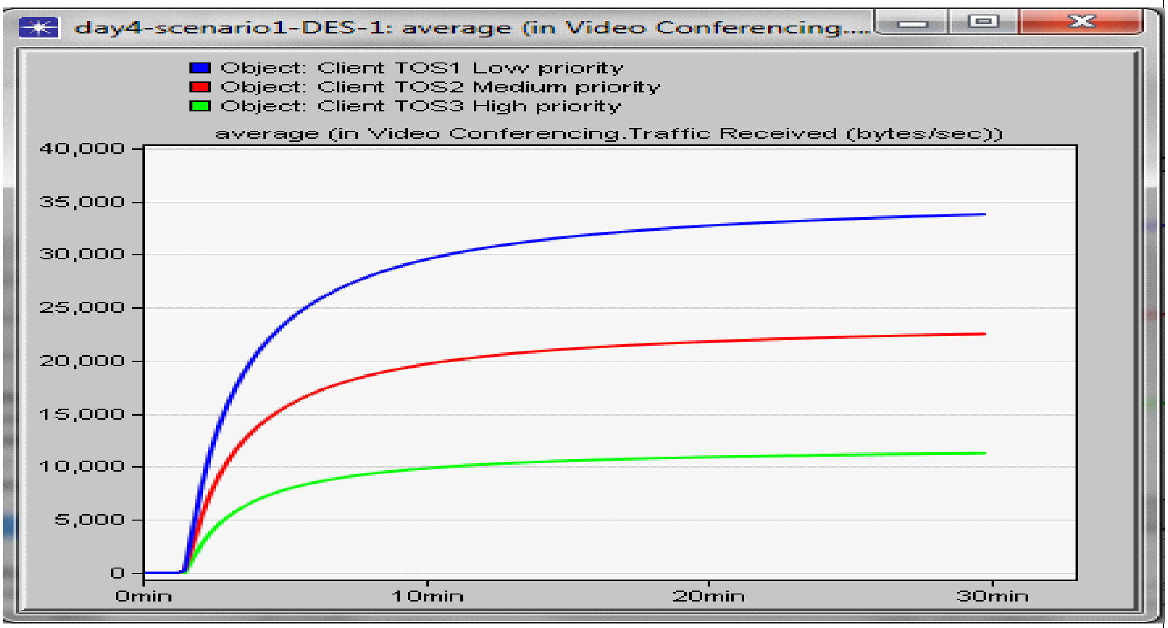

Figure 4.4: Traffic received (Bytes/sec)

The Figure 4.3 and Figure 4.4 shows the traffic sent per bytes and Traffic received per bytes respectively in scenario 1 with the input parameter stated above. The y-axis represented the amount bytes per second and x- axis shows the simulation period, in the graphs result. From the both graph of Figure 4.3 and Figure 4.4. Three colours indicating the amount of traffic sent and the amount of traffic received, based on the priority queuing polices that used in the IP QOS of the in-between routers. The green curve from Figure 4.3 shows 12000 bytes was sent from server TOS3 to client high priority with the same configured policy that support the service application received 12000 bytes as shown in Figure 4.4. The blue and the red curves from Figure 4.3 shows 23000 bytes was sent from server TOS2 to client medium priority with the same configuration policy of queuing to support the service application received 23000 bytes displayed by the red curve in Figure 4.4. The blue curve indicated that 34000 bytes was sent was from server TOS3 to client low priority with configuration to support the service application received 34000 bytes as shown by the blue curve in Figure 4.4.

5.2 Discussion

The evaluation and validation of the obtained results from the simulation scenario 1(one) of the Network model in Figure A1.1 this discussion of results will follows as a results of scenario 1 outcome based on the Quality of Service performance metric to validate with the British standards as outlined in BS EN 50132-5-1:2011 and International Telecommunication Union (ITU) standards specified in ITU-T Y.1541 for IP video streaming and the general video transmission performance metrics requirements of Quality of Service.

The scenario 1 results from the graph in Figure 4.1 of packet delay variation shows that the packet delay variation value was 58 ns delay, this is the highest delay variation values in the graph of Figure 4.2 which means that the packets experiences 58 ns delay, this delay is acceptable because its falls within the allowable delay variation range of British standards (BS EN 50132-5-1:2011) and ITU-T Y.1541 for IP video transmission and performance metrics of Quality of service requirements delay variation should not exceed 75ms (Cranley and Liam 2009), hence the Quality of video will be smooth when transmitting.

The scenario 1 results from the graph Figure 4.2 of packet end-to-end delay, shows that packet end-to-end delay was ranged from 50ms to 140ms. The end-to-end delay from the graph of Figure 4.2 with respective type of service from the Quality of service performance metric, this packet end-to-end delay experienced by the packet falls within the British standards (BS EN 50132-5-1:2011) and ITU-T Y.1541 values for IP Video transmission of performance requirements which the end-to-end delay values are 150ms, 200ms , 250ms and 300ms respectively (Cranley and Liam 2009, Perea, 2008, Van Der Schaar and Chou, 2007), and its effect on the video quality. Therefore the obtained results of 50ms -140ms of packet end-to-end delay shows in Figure 4.2 will not degrade and cause video impairment as result of Quality video.

The scenario 1 results of graph of Figure 4.3 and Figure 4.4 shows the amount of traffic sent in bytes per seconds and the traffic received in bytes per seconds, which both graphs shown the same amount of traffic sent and traffic received which means that there was no packet loss during the transmission period of simulation. That falls within the British standards (BS EN 50132-5-1:2011) and ITU-T Y. 1541 for IP video transmission but the both standards allows maximum of 1% loss of traffic based on different class of transmission (Cranley and Liam 2009, Fragkiadakis, Tragos and Askoxylakis, 2011, Lei and Xuan, 2009, Namee, 2009). Thus, in this result of 0 % loss. The video Quality will not be affected by freezing or artifact but the transmission will experience smooth video Quality.

The results obtained from the scenario 1 falls within the British standards (BS EN 50132-5-1:2011) and ITU-T Y.1541 standards for IP video streaming in the internet but companies like Hulus and Netflix adopted this standards in their day to day business in the multimedia industry, also there is different bandwidth allocation with different companies and different content application in transmission of videos usages because companies subscribe for bandwidth and IP address from internet service providers (ISP), which use already existing line like T1, T2, E1, E2 and ADSL etc. These lines have a specific bandwidth based on their applications, and the network link that is cables which supports this bandwidth in video transmission but bandwidth is cost intensive, therefore companies may use minimum bandwidth of 1.5Mbit/s in IP video streaming depending on their application.

6.0 CONCLUSION

The QoS performance evaluation of video streaming in the IPv6 has been studied. This was motivated by the IP addressing shortage in the earlier IPv4 and that IPv6 has rarely been exploited in the industries for video streaming. In Section 4, the network model created in OPNET environment was simulated with different parameters to investigate the performance metrics of classes of service in the quality of service of IPv6 in terms of video streaming services. The results obtained from the studied network model topology was simulated in scenario 1; it was found that the chosen video as well as network parameters alter the quality of the terminal streaming media accepted from the video streaming client. The following are the brief summary of the impact of these distinctive parameters upon the video streaming across IPv6 network obtained from the designed network model:

Frame rate: - it is the number of delivered video frames per second. It depicts the smoothness in the movement of the video. From the simulation results of the network model, it was discussed that the frame rate describes the end-to-end delay as well as delay variation in the network. These can be compensated with the increment of frame rate that is generated from the streaming server so that more traffic will be sent; this will require increment in the bandwidth to entirely convey the video stream to the client. This observation supports the section 2 of video that high definition video required more bandwidth than standard video to display the high-quality video. Thus, increasing the frame rate in IPv6 video streaming will require more bandwidth.

Frame size: - the frame size is relied upon to take advantage of the video resolution that decides the measure of feature or specific aspect which can be shown in the video frame with the high resolution. In section 2, greater resolution of video are the outcome of the increased frame size. The frame size depends on the readily available bandwidth which is directly proportional to delay variation as well as the end-to-end delay. Consequently, varying the frame size is thus directly proportional to the traffic transported from the streaming server. So, in IPv6 video streaming, the resolution of the video will be affected by the frame size sent.

Type of Service: - the Quality of service performance metrics are dependent on the value of type of service (TOS) in the IP header field. Thus, it decides the way the video frame or video packet is controlled from the in-between routers within the chosen path. The simulated network model results showed that the greater TOS value lead to good quality of service performance. This is because the chosen queue features were improved with the increment in the TOS value of the studied model. The chosen TOS value of any video impacts the end-to-end delay metric of the video streamed. Thus, the TOS value of video sent will determine the quality of IPv6 video received.

Bandwidth: - the bandwidth has a notable relationship with the performance metric of the quality of service such as end-to-end delay, delay variation, traffic sent, traffic received. If such relationship exits, then it can be inferred that the video quality that is delivered to the client to display depends on the entire designed network model topology. Hence from the discussion of the simulation results in Section 5. These results were found to be consistent with recommended QoS performance metrics for video streaming of ITU (Y.1541) and British (BS EN50132-5-1:2011) standards.

REFERENCE

Angelides, M.C., and Harry A. (2011). *The Handbook of MPEG Applications: Standards in Practice*. John Wiley & Sons (UK). Books24x7 [Online] Available at: http://common.books24x7.com/toc.aspx?bookid=43305 (accessed 27 June, 2013)

Amoss, J.J and Minoli, D. (2008). *Handbook of IPv4 to IPv6 transition: methodologies for institutional and corporate networks.* Auerbach Publications Taylor and Francis Group, Boca Raton.

Austeberry, D. (2002) *The Technology of Video and Video Streaming.* Burlington: Focal Press.

Austeberry, D. (2005) *The Technology of Video and Video Streaming.* 2nd edn. Burlington: Focal Press.

British Standards Institution (2011) *Alarm systems – CCTV surveillance systems for use in security applications- Part 5-1: Video transmission – General video transmission performance requirements. BS EN 50132-5-2011. British Standards Online* [Online]. Available at: htt://bsol.bsigroup.com (Accessed: 17 January 2014).

Brosh, E., Baset, S.A., Misra, V., Rubenstein, D. and Schulzrinne, H. (2010) 'The Delay Friendliness of TCP for Real-Time Traffic,' *Networking, IEEE/ACM Transactions on, IEEE*, 18(5), pp.1478-1491, *IEEEXplore* [Online]. DOI: 10.1109/TNET.2010.2050780. (Accessed: 8 October 2013).

Cranley, N. and Liam, M. (2009) *Handbook of Research on Wireless Multimedia: Quality of Service and Solutions. (eds) IGI Global*. Books24x7 [Online]. Available at: http:// common.books24x7.com.ezproxy.brad.ac.uk (Accessed: 25 August 2013).

Christ, R. D. and Wernli SR, R. L. (2014) Chapter 8 – Cables and Connectors. In: Christ, R. D. and Wernli, R. L. (eds) The ROV Manual (Second Edition). Oxford: Butterworth-Heinemann.

De Vogeleer, K., Popescu, A., Fiedler, M. and Erman, D. (2012) 'Content dependency of the traffic control in the darwin streaming server'. *Next generation internet (NGI), 8th Euro-NGI conference on* 25-27 June 2012. DOI: 10.1109/NGI.2012.6252166. (Accessed: 16 February 2014).

Dilger, D. E. (2010) *Snow Leopard Server*. John Wiley & Sons. Books24x7. <*http://common.books24x7.com.ezproxy.brad.ac.uk/toc.aspx?bookid=40811*> (Accessed: 18 February 2014).

Ding, Y., Du, Y., Hu, Y., Liu, Z., Wang, K., Ross, W., and Ghose, A. (2011) *Broadcast yourself: Understanding YouTube Uploaders*, proc. 2011 ACM Internet Measurement Conference (Berlin).

El-Hajjar, M. and Hanzo, L. (2013) 'A Survey of Digital Television Broadcast Transmission Techniques'. *Communications Surveys & Tutorials, IEEE*, 15(4), pp. 1924-1949, *IEEEXplore* [Online]. DOI: 10.1109/SURV.2013.030713.00220. (Accessed: 16 February 2014).

El Mahdi, B. M., Idboufker, N. and Ouahman, A. A (2012) ‘Evaluation of startup delay of video on demand in IMS’. *Networks and services (NGNS Next generation)*, 2-4 December 2012. DOI: 10.1109/NGNS.2012.6656098. (Accessed: 15 February 2014).

Evans, J. W., and Filsfils, C. (2007). *Deploying IP and MPLS QoS for Multiservice Networks: Theory & Practice*. Morgan Kaufmann.

Fgee, E.-B., Kenney, J.D., Phillips, W. J.; Robertson, W. and Sivakumar, S. (2005) ‘Comparison of QoS performance between IPv6 QoS management model and IntServ and DiffServ QoS models,’ *Communication Networks and Services Research Conference, Proceedings of the 3rd Annual*, 16-18 May 2005. DOI: 10.1109/CNSR.2005.28. (Accessed: 15 February 2014).

Fragkiadakis, A.G., Tragos, E.Z. and Askoxylakis, I.G. (2011) ‘Video Streaming Performance in Wireless Hostile Environments,’ *Multimedia and Ubiquitous Engineering (MUE), 5th FTRA International Conference on*, 28-30 June 2011 DOI: 10.1109/MUE.2011.54. (Accessed: 15 February 2014).

Gabrillini, A. and Mrak, M. (2013) ‘On coding and resampling of Video in 4:2:2 Chroma format for cascaded coding applications. *Image Analysis for Multimedia Interactive Services (WIAMIS), 14TH International Workshop* on 3-5 July 2013. DOI: 101109/wiamis.2013.6616153 (Accessed: 17 February 2014).

Hosik, S., Hana, Y., De Neve, W., Cheon, S. K. and Yong-Man, R. (2010) ‘Full-Reference Video Quality Metric for Fully Scalable and Mobile SVC Content’, *Broadcasting, IEEE Transactions on IEEE* 56(2), pp. 269-280, *IEEEXplore* [Online]. DOI: 10.1109/TBC.2010.2050628. (Accessed: 15 February 2014).

Hwang, J. (2009) *Multimedia Networking: From Theory to Practice*. Cambridge University Press. Books24x7. <*http://common.books24x7.com.ezproxy.brad.ac.uk/toc.aspx?bookid=31233*> (Accessed: 18 February 2014).

H3C Technologies (2010) *RTP packet format* [Online image] [Diagram] Hong Kong: H3C. Available at: http://www.h3c.com/portal/Products_Solutions/Technology/QoS/Technology_Introduction/200701/195599_57_0.htm (Accessed: 28 June 2013).

I-Hsuan, P., Meng-Hsien, L., Yen-Wen, C., Fong-Ming, Y. and Su, A. Y. S. (2013) Improvement of Streaming Video in Differential Service Networks by Using RED Mechanism, *complex, intelligent, and software intensive systems (CISIS), seventh international conference on* 3-5 July 2013. DOI: 10.1109/CISIS.2013.116. (Accessed: 16 February 2014).

International Telecommunication Union (2011) *SERIES Y: Global Information Infrastructure, Internet Protocol aspects and next – generation Networks, Internet protocol-Quality of Service and network performance. Y. 1541. ITU-T: Telecommunication Standardization Sector of ITU Online* [Online]. Available at: http://http://www.itu.int/ITU-T/recommendations. (Accessed: 25 August 2013).

Jinyao, Y., Muhlbauer, W. and Plattner, B. (2012) ‘Analytical Framework for Improving the Quality of Streaming over TCP,’ *Multimedia, IEEE Transactions on, IEEE,* 14(6), pp.1579-1590, *IEEEXplore* [Online]. DOI: 10.1109/TMM.2012.2187182. (Accessed: 8 October 2013).
Kurose, J.F. & Ross, K. W. (2013) *Computer networking. A top-down approach*. Boston, MA; London; Pearson Education.

Learn networking (2008) *How the Internet layer works* [Online image] [Diagram]: Learn-Networking.com. Available at: http://learn-networking.com/tcp-ip/how-the-internet-layer-works (Accessed: 28th June 2013).

Lei, Y. and Xuan, C. (2009) ‘Business video ready enterprise IP network architecture’, *Broadband Multimedia Systems and Broadcasting, BMSB '09. IEEE International Symposium on*, 13-15 May 2009. DOI: 10.1109/ISBMSB.2009.5133808. (Accessed: 19 February 2014).

Lu, Z. and Yang, H. (2012). *Unlocking the Power of OPNET Modeler*, Cambridge University Press.
Marchese, M., Raviola, A., Mongelli, M., Garibbo, A. and Gesmundo, V. (2006) ‘IPv4 versus IPv6 interworking with QoS guarantees. *Military communication s conference, 2006. MILCOM 2006. IEEE*, 23-25 October, 2006. DOI: 10.1109/MILCOM.2006.302082. (Accessed: 17 July 2013).
Mark Blanchet (2006) Migrating to IPV6 (*A Practical guide to implementing IPV6 in mobile and fixed networks*). John Wiley & sons offices.

Mesnik, B. (2010) *MPEG frames* [Online image] [Drawing]: Kintronics.
Available at: http://kintronics.blogspot.com/ (Accessed: 28 June 2013).

Mozammil, M., Zakariya, S. M. and Inamullah, M. (2012) ‘Analysis of Video Compression Algorithm on Different Video Files,’ *computational intelligence and communication networks (CICN), fourth international conference on* 3-4 November 2012. DOI: 10.1109/CICN.2012.49. (Accessed: 15 February 2014).

Namee, K. (2009) ‘Performance evaluation of multimedia application QoS over wireless and wired IPv6 Networks’. *Communication software and networks, ICCSN ’09. International conference on*, 27-28 February 2009. DOI: 10.1109/ICCSN.2009.158 (Accessed: 18 July 2013).

Negreiros, M., Klein, H. A., Bonatto, A. C., Soares, A. B. and Susin, A. A. (2012) Towards video processing architecture for SBTVD. *Programmable Logic (SPL) VIII Southern Conference on*, 20-23 March 2012. DOI: 10.1109/SPL.2012.6211801 (Accessed: 17 February 2014).

Perea, R. M. (2008). *Internet Multimedia Communications Using SIP: A Modern Approach Including Java® Practice*. Morgan Kaufmann.

Ping, G. and Desheng, F. (2010) ‘The Discussions on Implementing QoS for IPv6’. *Multimedia technology (ICMT), International Conference on,* 29-31 October 2010. DOI: 10.1109/ICMULT.2010.5630961. (Accessed: 19 August 2013).

Pulipaka, A., Seeling, P., Reisslein, M. and Karam, L. J. (2013) ‘Traffic and Statistical Multiplexing Characterization of 3-D Video Representation Formats’. *Broadcasting, IEEE*

Transactions on, IEEE, 59(2), pp. 382-389, *IEEEXplore* [Online]. DOI: 10.1109/TBC.2013.2244792 (Accessed: 17 February 2014).

Poynton, C. (2012) *Digital video and HD: algorithms and interfaces*. 2nd edn. Waltham: Elsevier Morgan Kurfmann.

Sethi, A. S. and Hnatyshin, V. Y. (2012) *The Practical OPNET User Guide for Computer Network*
Simulation, Taylor & Francis

Richardson, I. E. (2011). *The H. 264 advanced video compression standard*. John Wiley & Sons.

Seungsin, L., Seok, L., Byungtae, O., Kwan-Jung, O., Ilsoon, L., Jin, Y. L. and Kim, J. D. (2012): '3D video format and compression methods for Efficient Multiview Video Transfer' *Consumer Communications and Networking Conference (CCNC) IEEE,* pp. 10-14 14-17 January 2012. DOI: 10.1109/CCNC.2012.6181054 (Accessed: 17 February 2014).

Schreer, O., Feldmann, I., Weissig, C., Kauff, P. and Schafer, R. (2013): 'Ultrahigh-Resolution Panoramic Imaging for Format-Agnostic Video Production', *Proceedings of the IEEE,* 101(1) pp. 99-114, IEEEXplore [Online]. DOI: 10.1109/JPROC.2012.2193850. (Accessed: 18 February 2014).

Simpson, W. (2008) *Video over IP: IPTV, Internet Video, H.264, P2P, Web TV, and Streaming: A Complete Guide to Understanding the Technology, Second Edition*. Focal Books24x7. *<http://common.books24x7.com/toc.aspx?bookid=32178>* (Accessed: 27 June 2013)

Smith, M. (2012) *6 0f the Best Streaming Media Service compared*, www.makeuseof.com/tag/6-streaming-media-service-compared (Accessed: 29 July 2013)

Schulzrinne, H., Casner, S., Frederick, D. and Jacobson, V. (1996a). RTP: *A Transport Protocol for Real-Time Applications,* Internet RFC 1889. Available at: http://www.ietf.org/rfc/rfc 1889. (Accessed: 27 June 2013)

Shulzrinne, H., Rao, A., and Lanphier, R. (1996b) Real Time Streaming Protocol (RTSP). Networking Group. Request for Comments: 1996-RFC 2326. Available at: http://www.ietf.org/rfc/rfc2326.txt. (Accessed: 29th June 2013).

Sun Microsystems (2003) *IPv6 Administration Guide*. Santa Clara: sine nomine.

Tanenbaum, A.S. (2003) *Computer networks*. 4th edn., New Jersey; Prentice Hall.

Tronco, T. (2010). *New Network Architectures: The Path to the Future Internet*, Springer-verlagberlin Heidelberg.

Van Der Schaar, M. and Chou P.A. (2007) *Multimedia over IP and Wireless Networks: Compression, Networking and Systems*. Burlington: Academic Press

Warren, C. (2011) *5 of the Best streaming media services compared*, www.mashable.com/2011/02/14/streaming –media-comparison (Accessed 29 July 2013).

Wang, Z. (2001) *Internet QoS: Architectures and mechanisms for Quality of Service.* San Diego: Academic Press.

Wenzel, G., Gehrke, G., Belzner, M., Stoll, D. and Thomas, W. (2009) Deduction of video quality degradation in Transport networks, *photonic networks, ITG symposium on* 4-5 May 2009. Available at: URL: http://ieeexplore.ieee.org/xpl/login.jsp?tp=&arnumber=5755123&isnumber=5755099&url=http%3A%2F%2Fieeexplore.ieee.org%2Fstamp%2Fstamp.jsp%3Ftp%3D%26arnumber%3D5755123%26isnumber%3D5755099 (Accessed: 27 February 2014).

Wilcox, J.R. and Gibson, D. K. (2005) *Video communications: The Whole Picture.* San Francisco: CMP books.

Zambelli, A., (2013) *A history of media streaming and the future of connected TV*, www.guardian.co.uk/media-network/media-network-blog. (Accessed 29 July 201).

Zha, Z., Wang, M., Zheng, Y., Yang, Y., Hong, R. and Chua, T.-S., "Interactive Video Indexing with Statistical Active Learning," *Multimedia, IEEE Transactions on*, IEEE (14)1, pp.17, IEEEXplore [Online]. DOI: 10.1109/TMM.2011.2174782. (Accessed: 19 February 2014).

Zhang, X., Xu, Y., Hu, H., Liu, Y., Guo, Z., and Wang, Y. (2013). 'Modeling and Analysis of Skype Video calls: Rate Control and Video Quality'. *Multimedia, IEEE Transactions on, IEEE* PP (99), pp. 1-1, *IEEEXplore* [Online]. DOI: 10.1109/TMM.2013.2247988. (Accessed: 8 October 2013).

Zhenhua, W., Qiong, S., Xiaohong, H and Yan, M. (2010) 'IPv6 end-to-end QoS provision for heterogeneous networks using flow label'. *Broadband Network and Multimedia Technology (IC-BNMT), 3rd IEEE International Conference on*, 26-28 October 2010. DOI: 10.1109/ICBNMT.2010.5704882. (Accessed: 19 August 2013).

Zhenqi, W. and Mei, Y. (2011) 'The Research and Application of Internet Protocol Version 6(IPv6)'. *Intelligent computation technology and automation (ICITA), International Conference on,* 28-29 March 2011. DOI: 10.1109/ICICTA.2011.79. (Accessed: 16 August 2013).

Yamagishi, K. and Shan, G. (2013) 'Light-weight audiovisual quality assessment of mobile video: ITU-T Rec. P.1201.1,' *Multimedia Signal Processing (MMSP), IEEE 15th International Workshop on*, 30 September - 2 October. DOI: 10.1109/MMSP.2013.6659333. (Accessed: 21 February 2014).

Yong, L., Bin, D., Shuling, W., Haibo, Y. and Xingwei, W. (2010) "Design and Implementation of Performance Testing Utility for RTSP Streaming Media Server," *Pervasive Computing Signal Processing and Applications (PCSPA), First International Conference on*, 17-19 September 2010. DOI: 10.1109/PCSPA.2010.55. (Accessed: 17 February 2014)