Throughput Performance Evaluation of Video/Voice Traffic in IPv4/IPv6 Networks

Md. Tariq Aziz
School of Computing
Blekinge Institute of Technology
Karlskrona, Sweden

Mohammad Saiful Islam
School of Computing
Blekinge Institute of Technology
Karlskrona, Sweden

Md. Nazmul Islam Khan
Dept. of Electrical & Comp. Eng.
Presidency University
Dhaka, Bangladesh

ABSTRACT
This paper presents a study of throughput performance of voice and video traffic based on simulative and analytical methods in IPv4/IPv6 networks. This research aims to find out what Internet protocol performs better in terms of throughput under congested circumstances in IPv4/IPv6 networks. In doing so three different network loads in both scenarios are considered. In the context of network load, the importance of varying network load is realized while configuring and simulating the network models. For instance, a medium network load, high network load then a worst possible network load are considered to understand the impact on the performance of throughput in IPv4/IPv6 networks. The network topology scenarios are partially meshed on the implication with a small ISP domain as this is an ideal choice of IP domain corresponded to a realistic network topology. Two network models are defined which allow us to compare the obtained results. In addition, IPv4 network model is used and extended in terms of configuring IPv6 network model as. The simulation has been carried out using Optimized Network Engineering Tool (OPNET). This paper shows that analytical and simulative approaches produce same results in terms of throughput performance for video/voice traffic. From Internet Protocol performance perspective, IPv6 experiences more throughput than IPv4.

General Terms
Performance evaluation of Computer Networks and Communications.

Keywords
Throughput, Video, Voice, IPv6, and OPNET.

1. INTRODUCTION
Internet Protocol version 4 (IPv4) is one of the key foundations of the Internet, which is currently serving up to four billion hosts over diverse networks. Despite this, IPv4 has still been successfully functioned well since 1981. Over the last couple of years, the massive growth of the Internet has been evident requiring an evolution of the whole architecture of the Internet Protocol. There-fore, in order to strengthen the existing architecture of Internet Protocol, IETF has developed Internet Protocol version 6 (IPv6) [1]. IPv6 offers a significant improvement of IPv4 when it comes to the unlimited address space, the built-in mobility and the security support, easy configuration of end systems, as well as enhanced multicast features, etc [2]. On the other hand, due to the fascination of end users of the World Wide Web (WWW) and the popularity of real-time applications, we can now observe new increasing demands on real-time multimedia services over the Internet.

Due to the practical difficulties in obtaining “large” blocks of new, unassigned IPv4 addresses, major organizations in the fast-growing markets of Asia and Europe, as well as mobile service providers worldwide are under increasing pressure to migrate from the entrenched IPv4 standard to the emerging IPv6 one. As we continue to see increasing global scale deployments of IPv6 networks, there has also been an increasing interest in measuring the performance of these IPv6 networks [3]–[10].

The study [11] undertaken by the authors, C. Bouras, A. Gkamas, D. Primpas, and K. Stamos followed an experimental measurement approach to evaluate Quality of Service aspects in an IPv6 domain. The results from this study have shown the transmission rate (throughput) where IPv6 maintains higher the transmission rate than that of IPv4.

Three methods are available for packet-level performance evaluation in IP networks which include: mathematical analysis, measurement and computer simulation [12]. From the above described related work, it is observed that most of the work that have been done so far by following the method as an experimental measurement. None of the above research work has done a simulative and analytical evaluation of real-time applications such as video and voice performance in terms of throughput in relation to Pv4/IPv6 networks. In this work, realizing the simulation and analytical approaches a comparative performance analysis of video and voice conferencing in conjunction with IPv4/IPv6 networks has been complemented.

The outline of this paper is organized as follows; in section 1, the introduction of this research is discussed. Section 2 is dedicated to outline the research methodology. The detail steps to design and implement a network model using OPNET is discussed in section 3. It also describes how the statistics in OPNET was collected. Section 4 describes the simulation results followed by section 3. Finally section 5 concludes the research work with possible future work.

2. RESEARCH METHODOLOGY
This section provides the key points involved with the research methodology in conducting this research work. Within such points, one of them is the chosen evaluation methods compared with other possible methods and another one is the justification about choice of them. Additionally, in detail section III illustrates the platforms and possible network scenarios applied to investigate the means of mechanism attempted by this work.

2.1 Justification of the Method of Study

This section briefly discusses the different possible methods of research on networks and explains the choice of simulation as the appropriate method of study for the purpose of this work. Likewise, this section justifies the use of OPNET as the selected simulator and provides information on the procedures followed in order to reduce the possibility of simulation errors. Three methods are available for packet-level performance evaluation in IP networks which include: mathematical/analytical analysis, direct measurement and computer simulation [12]. After thoughtful consideration, simulation and mathematical were found to be the suitable method of study in this work.

At first sort of characteristics associated with the mathematical model need to be addressed prior to give a reason behind our choice of methods being made as mathematical analysis. In that context, advantage of using mathematical analysis lies in the cost, time, and ability of providing better predictive results. That is what led us to choose mathematical analysis.

In this work, as a choice of method, direct measurement could be another alternative to the simulation. According to the intended course of work for this thesis, several number of network models in a small scale were to be studied. In the case of direct measurement, the analysis has to be conducted on an operational network which may lead to a disruptive situation. It is generally too expensive to build an operational network in conjunction with configuration complexity. Nevertheless, using this method one can also perform a realistic observation and achieve fairly accurate results. Therefore, particularly expense and additional configuration complexity in modeling an operational network has not allowed us to select this method in our work.

There was one more selection alternative of the simulators e.g., NS-2, QualNet, OPNET and OMNeT++. In order to perform the simulation work we had to select the suitable simulator. For our work the suitable choice was the popular OPNET simulator introduced by the OPNET Technologies, Inc [11] [12]. The OPNET modeler is an object oriented and a Discrete Event System (DES) based network simulator, which is highly reliable and efficient simulation tool for modeling a network. DES is widely used in the performance evaluation of complex networks and communication systems. The OPNET simulator is well-known for network design and attractive features.

After selecting the simulator, it is important to investigate the obtained results. Simulation investigations form a vital part of networking research, and have long been used by the research community in, for example, the design of protocols and evaluation of quality of service mechanisms. But many theoretical problems lie in the simulation results, which need to be taken in account while performing simulation [12]. In that case, for validation and verification of simulation results, we have considered the guidelines given in [15] [16].The detailed description of the network modeling and implementation using OPNET has been presented in III.

3. NETWORK MODEL AND IMPLEMENTATION

3.1 Network Model Configuration

Being OPNET as a choice of simulators in favor of our intended work described in Section III, the following subsequent sections discuss about the network components used in the network models, assumption and voice and video traffic generation.

3.2 Network Components

This section discusses about the following network components used in the suggested network models running on OPNET [17].

The ethernet2_slip8_ler (Label Edge Router) and ethernet2_slip8_lsr (Label Switched Router) node models are used to represent an IP-based gateway running MPLS and supporting up to two Ethernet interfaces and up to 8 serial line interfaces at a selectable data rate. IP packets arriving on any interface are routed to the appropriate output interface based on their destination IP address.

The ethernet16_switch node model is used to represent a switch supporting up to 16 Ethernet interfaces. The switch implements the Spanning Tree algorithm in order to ensure a loop free network topology. Switches communicate with each other by sending Bridge Protocol Data Units (BPDU's). Packets are received and processed by the switch based on the current configuration of the spanning tree.

The ethernet_wkstn_adv node model is used to represent a workstation with client-server applications running over TCP/IP and UDP/IP.

The 10BaseT and 100BaseT full duplex links are used to represent the Ethernet connections operating at 10 Mbps and 100Mbps, respectively. These links can connect any combination of the nodes such as Station, Hub, Bridge, Switch and LAN nodes (except Hub-to-Hub, which cannot be connected).

The ppp_adv, point-to-point full duplex link is used to connect two nodes with serial interfaces (e.g., routers with PPP ports)) at a selectable data rate.

The Application_Config includes a name and a description table that specifies various parameters for the different applications (i.e. video conferencing and voice applications). The specified application name is used while creating user profiles on "Profile_Config" object.

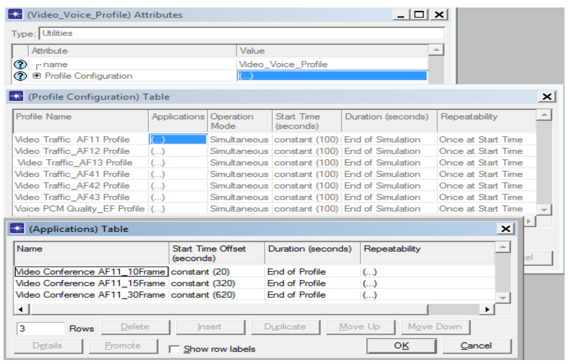
The Profile_Config is used to create user profiles. These user profiles can be specified on different nodes in the network to generate application layer traffic. The applications defined in the Application_Config are used by this object to configure profiles. Traffic patterns can be specified followed by the configured profiles and the applications.

3.3 Network Traffic Generation

Detailed information about the configurable parameters for voice applications is given in Table 1, 2 and 3. In voice applications, voice traffic configuration we have set the codec bit rate at 64 Kbps and codec sample interval 10 ms whereby codec sample size is calculated using 64,000*10/1000 = 640 bits (e.g., codec bit rate=sample interval/sample size). Thus the sample size is 80 bytes. For 10 ms sample interval 100 packets per second needs to be transmitted [18].

Video and voice conferencing profiles are defined in the source workstations while corresponding destination workstations are enabled with their respective supported services. In OPNET terminology, in order to generate voice and video traffic, voice and video conferencing profiles are configured in such a way where video and voice applications can be controlled in terms of their start, end times and repeatability. This is done by adding this profile to each workstation's lists of supported profiles. The start time and offset time for the video_and_voice_profile configuration parameters are presented in Table 1. It is noted that while configuring the profile for video and voice conferencing; the first voice and video calls by each designated workstation start at 120 seconds (Phase-1) (i.e., start time of 100 seconds with offset time of 20 seconds) and it continues till 420 seconds, while the second call is added at 420 seconds of simulation time (Phase-2), and finally the third call is added at 720 seconds of the simulation time (Phase-3). Which follows each designated workstation is having three interactive video and voice conferencing sessions running simultaneously during the simulation period (i.e. 720-1800 seconds).

Table 1. Voice and video profile configuration parameters



(Video_Voice_Profile) Attributes

Type: Utilities

Attribute	Value
name	Video_Voice_Profile
Profile Configuration	(...)

(Profile Configuration) Table

Profile Name	Applications	Operation Mode	Start Time (seconds)	Duration (seconds)	Repeatability
Video Traffic AF11 Profile	(...)	Simultaneous	constant (100)	End of Simulation	Once at Start Time
Video Traffic_AF12 Profile	(...)	Simultaneous	constant (100)	End of Simulation	Once at Start Time
Video Traffic_AF13 Profile	(...)	Simultaneous	constant (100)	End of Simulation	Once at Start Time
Video Traffic_AF41 Profile	(...)	Simultaneous	constant (100)	End of Simulation	Once at Start Time
Video Traffic_AF42 Profile	(...)	Simultaneous	constant (100)	End of Simulation	Once at Start Time
Video Traffic_AF43 Profile	(...)	Simultaneous	constant (100)	End of Simulation	Once at Start Time
Voice PCM Quality_EF Profile	(...)	Simultaneous	constant (100)	End of Simulation	Once at Start Time

(Applications) Table

Name	Start Time Offset (seconds)	Duration (seconds)	Repeatability
Video Conference AF11_10Frame	constant (20)	End of Profile	(...)
Video Conference AF11_15Frame	constant (320)	End of Profile	(...)
Video Conference AF11_30Frame	constant (620)	End of Profile	(...)

3 Rows Delete Insert Duplicate Move Up Move Down
Details Promote Show row labels OK Cancel

Table 2. Voice application parameters

Attribute	Value	
Silence Length (s)	Incoming Silence Length (s)	Exponential (0.65)
	Outgoing silence Length (s)	Exponential (0.65)
Encoder scheme	G.711	
Voice Frames per packet	1	
Type of Service	Best Effort (0)	
Compression Delay (s)	0.02	
Decompression Delay (s)	0.02	

Table 3. Voice and video application configuration parameters

Video_and_Voice_Profiles	Frame Size (Bytes)	Bit Rate (Kbps)	Total Offered Load (Kbps)	Start-time (s)
VideoConference_AF11_10Frame	4000	320		120
VideoConference_AF11_15Frame	4000	480	1760	420
VideoConference_AF11_30Frame	4000	960		720
VideoConference_AF12_10Frame	3000	240		120
VideoConference_AF12_15Frame	3000	360	1320	420
VideoConference_AF12_30Frame	3000	720		720
VideoConference_AF13_10Frame	2000	160		120
VideoConference_AF13_15Frame	2000	240	880	420
VideoConference_AF13_30Frame	2000	480		720
VideoConference_AF41_10Frame	3500	280		120
VideoConference_AF41_15Frame	3500	420	1540	420
VideoConference_AF41_30Frame	3500	840		720
VideoConference_AF42_10Frame	2500	200		120
VideoConference_AF42_15Frame	2500	300	1100	420
VideoConference_AF42_30Frame	2500	600		720
VideoConference_AF43_10Frame	1500	120		120
VideoConference_AF43_15Frame	1500	180	760	420
VideoConference_AF43_30Frame	1500	360		720
Voice PCM Quality_EF	80	64		120
Voice PCM Quality_EF	80	64	270	420
Voice PCM Quality_EF	80	64		720

3.4 Simulation Scenarios

OPNET Modeler 14.0 [17] has been used for the simulation analysis. This section explains the network model used in this study. Two network scenarios have been prototyped as follows, which will be elaborately demonstrated in the up-coming sections. Scenario 1 is modeled as an IPv4 scenario while scenario 2 serves as another IPv6 scenario to demonstrate traffic delivery in a best-effort IPv6 network under congested condition in which no QoS is configured. It is important to mention that in scenarios, the routers ethernet2_slip8_ler and ethernet2_slip8_lsr [17] correspond to the LERs and LSRs, respectively. These routers are interconnected via ppp_adv point-to-point link operated at 4Mbps data rate. The links used to connect switches with the routers (i.e. LER1 and LER2) are 100Base-T, while 10Base-T is to connect the workstations with the switches. The switches namely switch_1 and switch_2 (i.e. ethernet16_switch) are connected with routers ((i.e. ethernet2_slip8_ler and ethernet2_slip8_lsr)) using 100Base-T. The scenarios to be modeled in this work are outlined as follows:

3.4.1 Scenario 1: IPv4

Scenario 1 follows a typical meshed IP network where packets are forwarded from IPv4 source to the corresponding IPv4 destination through the IPv4 core domain with the best-effort policies. In this scenario, each pair uses a best-effort service as a Type of Service (ToS).

The reference network topology depicted in Fig. 1, is composed of six pairs of video conferencing workstations and a pair of voice workstation. The core network consists of nine LSRs (i.e. Label Switched Router) and two LERs (i.e. Label Edge Router). All the LSRs and LERs of the core network are interconnected using the point-to-point link (ppp_adv) operated at the data rate of 4Mbps. In our reference network topology (Fig. 1), OSPF [19] routing protocol is used under normal condition without considering load balancing feature. The purpose of not considering load balancing is that congestion in network can be better understood.

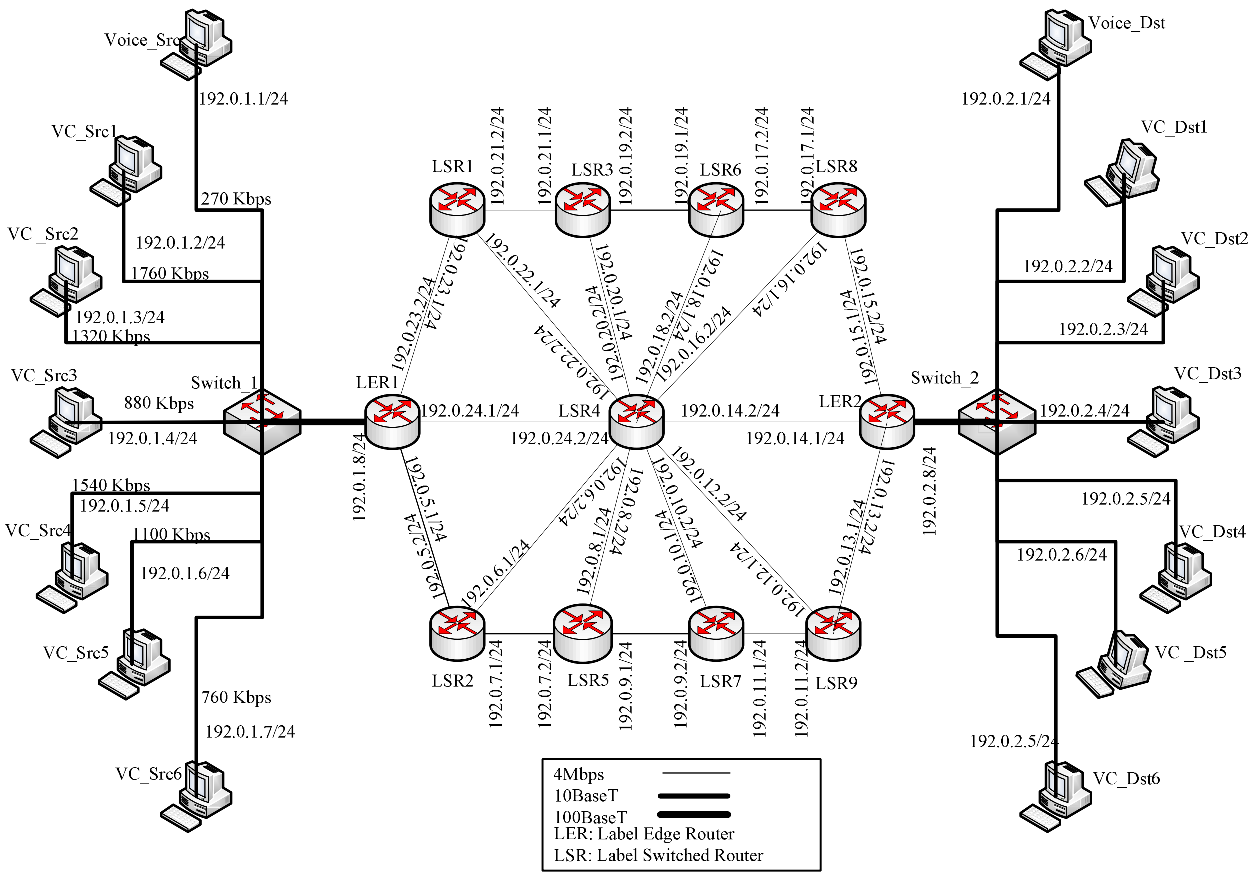


Fig 1: IPv4 network topology

3.4.2 *Scenario 2: Baseline_IPv6*

Topology depicted in Fig. 2 represents scenario 2 which is same as the scenario 1, but IPv6 is configured in this. All IP nodes in the scenario 1 are dual-stack capable that means they can support both IPv4 and IPv6. In this scenario, to manually configure an interface to support IPv6 only but not IPv4, the IPv4 address of the interface is set to “No IP Address”. IPv6 link-local and global addresses on interfaces of all nodes in the network have manually been configured. In order to configure IPv6 in the network, Link-Local Address attribute is set to Default EUI-64 while Global Addresses is set to EUI-64 with the specification of the first 64 bits of the address. The remaining 64 bits of the address are set to an interface ID unique to the interface. With regard to routing protocol configuration of IPv6 network, as the process v2 of OSPFv2 is already running for IPv4 network (scenario 1). In this scenario, the process v2 has been disabled instead another process version (v3) [20] is enabled to the OSPF parameters configuration.

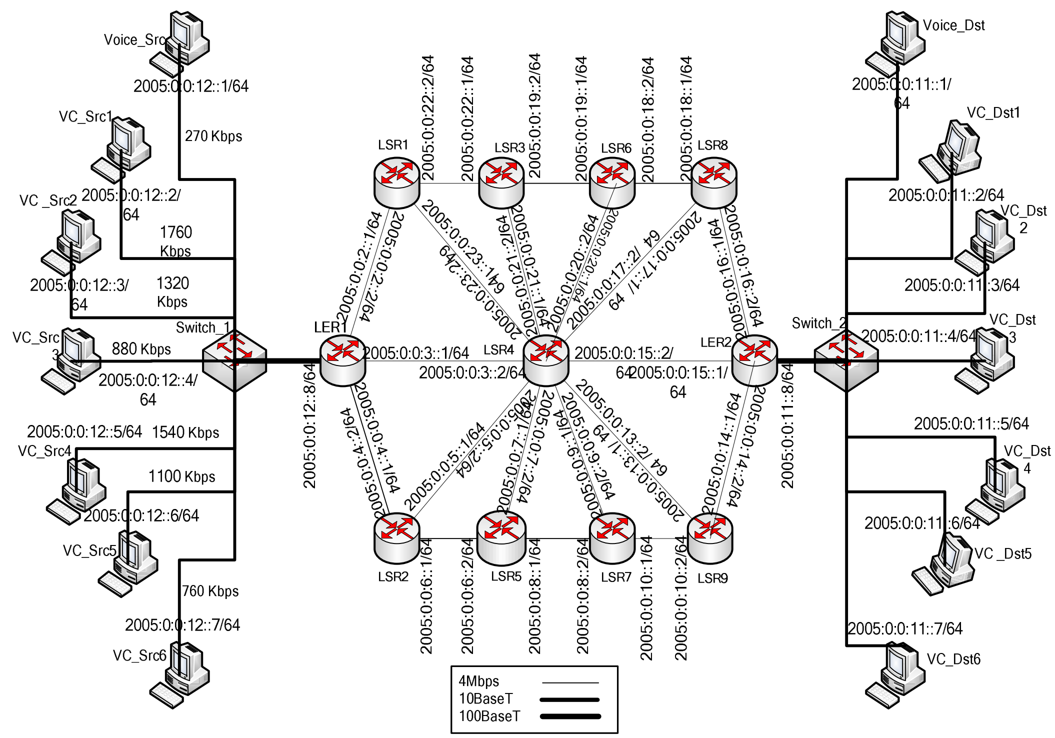


Fig 2: IPv6 network topology

3.4.3 Simulation run-time

All the simulations run for 1800 seconds, and all applications that generate the traffic (i.e. voice and video conferencing) start simultaneously at 120 seconds of the simulated time, that is, every event has the same probability to occur at every value at 120 seconds. The simulation is implemented in OPNET Modeler 14.0 running on a HP laptop with Windows 7, Pentium IV 1.7 GHz with 2GB of RAM. For all of scenarios the information about simulation run in OPNET has been shown in Table 4. These information regarding two different scenarios have been collected from OPNET after the simulation run is finished.

Table 4. Simulation run-time Information.

Scenarios	Duration of simulation phases			Total Simulation Time(s)
	Duration of Phase-1 (s)	Duration of Phase-2 (s)	Duration of Phase-3 (s)	
Scenario 1	120–420	420–720	720–1800	1800
Scenario 2	120–420	420–720	720–1800	1800

3.4.4 Collecting statistics in OPNET

Before running the simulation, OPNET needs to be configured to obtain the desired statistics. Once the desired statistics have been specified, OPNET automatically produces those statistics. However, there is a key factor associated with statistics collection. First of all we need to look at how the simulator processes the generated statistics during the simulation. OPNET Modeler has three modes of the statistics collection for processing the statistics, which include All Values, Sample and Bucket. Details of these statistics collection modes can be found in OPNET tutorial.

In our case we have collected our intended statistics using bucket mode because the results of this mode are useful as they show the general trend of the statistic's variations, even if they do not capture any of the rapid changes [17]. Bucket mode is also called as the default mode in OPNET. Now we will discuss how does bucket mode processes the statistics before being stored in file of the simulation results. In order to explain that here we have considered the specified values in bucket mode configuration made in our simulated scenarios. For instance, bucket mode has one important parameter called bucket width which is measured in seconds. This can be obtained by setting up the Values per statistics and simulation time. Values refer to how many values of the statistics will be reported in the graph and stored in the simulation results file. In that case, we have set 100 as the Values per statistics parameter under Configure Simulation Parameter before running the simulation with simulation time 1800 s (seconds). After setting up these values, we can get the bucket width in way that total simulation time i.e., 1800 s is divided by the Values 100 which gives 18 s.

4. RESULTS AND ANALYSIS

The IP Performance Metrics (IPPM) Working Group of IETF has defined throughput referring to the amount of data packet successfully received by the destination node [21]. The throughput is usually measured in bits per second (bits/sec).

4.1 Throughput Performance in Scenarios 1 and 2

In this experiment, throughput has been measured at the physical layer. Using the conventional routing protocol such as OSPF under normal condition (with-out considering load balancing feature of OSPF), all the traffic follows the shortest path where the link from LER1 to LER2 (LER1→LSR4→LER2) carries all the traffic flows. No traffic passes through the paths, LER1→LSR1→LSR3→LSR6→LSR8→LER2 and LER1→LSR2→LSR5→LSR7→LSR9→LER2) either in IPv4 network or IPv6 network. All real-time traffic flows are being transmitted throughout the entire network with best-effort treatment.

During phase-1, the different UDP CBRs (Constant Bit Rates) generated at the seven different sources i.e., voice_src, vc_src1, vc_src2, vc_src3, vc_src4, vc_src5, and vc_src6 include 64,000, 320,000, 240,000, 160,000, 280,000, 200,000, and 120,000 bps, respectively. These CBRs generated at the application layer don't include any protocol overhead that turning out to be the total injected traffic as payload of about 1,384,000 bps.

Looking at the Fig. 3 and Table 5, during phase-1 the throughput is appeared to roughly be 1,457,000 bps and 1,507,000 bps for IPv4 and IPv6, respectively. From the achieved throughput and the injected payload, we can determine the total overhead added by layer protocols (i.e., RTP + UDP+IP + PPP) (1,457,000−1,384,000) bps= 73,000 bps while the overhead is added by layer protocols in IPv6 is about 123,000 bps (i.e., (1,507,000−1,384,000) bps). The difference between total overheads of IPv4 and IPv6 becomes about 50,000 bps.

To analytically determine the total overhead, one of the important things that we have to identify how many packets for voice and video traffic are transmitted throughout the network. According to our voice traffic configuration, we have set the codec bit rate at 64000 bps and codec sample interval 10 ms whereby codec sample size is calculated using 64,000*10/1000 = 640 bits (e.g., codec bit rate=sample interval/sample size). Thus the sample size is 80 bytes. For 10 ms sample interval 100 packets per second needs to be transmitted. Therefore, the total bandwidth required for voice traffic is (80+48)*100*8= 102,000 bps followed by ((sample size in bytes + header overhead of layer protocols)*(number of transmitted packets per second)*8) in one direction. The overhead added by voice traffic for 100 packets is about 38,000 bps (i.e., (102,000−64,000) bps).

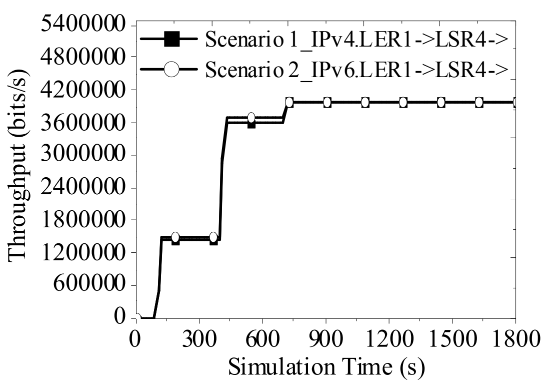


Fig 3: Throughput in bits/s for scenarios 1 and 2

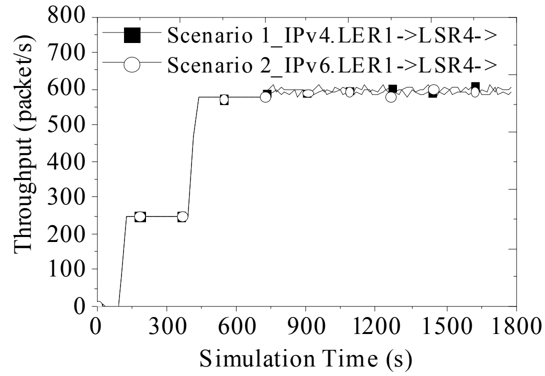


Fig 4: Throughput in packets/s for scenarios 1 and 2

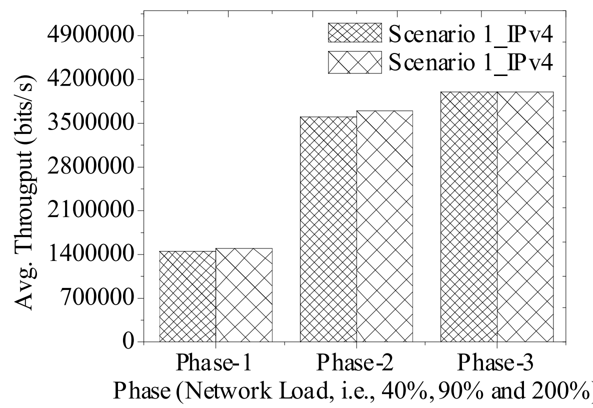


Fig 5: Average throughput for scenarios 1 and 2.

At the same time, the video traffic has been configured in way that this has to do with different frame sizes for the different traffic sources (Table 3). For example, in the course of phase-1 of the simulation, source vc_src1 generating video traffic at the CBR of 320,000 bps of payload at the application layer where incoming/outgoing frame size is set at 4000 bytes with 10 frame/s. For IP layer and PPP (Point-to-Point Protocol) at physical layer, MTU (Maximum Transmission Unit) has been set at 1500 bytes. As a result, there will be fragmentation for the video traffic. In the case of different frame sizes for all of the video traffic, analytical calculation of those fragmented packets would be very complicated for each traffic sources. With that said, the obtained statistics of throughput in packet/s depicted in Fig. 4 is favored in roughly determining the total number of video packets transmitted during the phase-1.

Total number of packets for both video and voice are found to roughly be 250 packets. 100 packets per second for voice traffic already determined earlier and the rest of the packets around 150 are for video traffic.

Once the number of packets transmitted by video and voice sources during the pahse-1 is determined, we can calculate the total overhead added to total payload by different protocols. The different layer protocols consist of (RTP+UDP+IP + PPP), which include total header sizes (12+8+20+8)*8=384 bits for each packet in IPv4, and (12+8+40+8)*8= 544 bits for each packet in IPv6 [22]. From our analytical calculation, we now roughly determine the total bandwidth required in IPv4 and IPv6. This is obtained followed by the number of voice and video packets multiplied by total overhead and 8. That means the required bandwidth for IPv4 is 250*384= 96,000 bps and 250*544=136,000 bps in IPv6.

We have already calculated the total injected traffic rate, 1,384,000 bps of payload thereby we get about 1,480,000 bps (i.e., (1,384,000+96,000) bps) as theoretical throughput for IPv4 while throughput for IPv6 is about 1,520,000 bps (i.e., (1,384,000+136,000) bps), which correspond to the obtained throughput from the simulation results.

It is noticed that comparing the theoretical throughput and the resulting throughput from OPNET simulation, it is found that the theoretical throughput is little bit higher than simulation results for both IPv4 and IPv6 networks. This is due to the fact that particularly for video traffic, the total number of packets considered from the OPNET simulation results are not an absolute value of the packets. The reason lies on how OPNET process the statistics, which has elaborately been described in section III under statistics collection in OPNET section.

During the phase-2 (i.e., medium load; simulation time: 420-720 s), the throughput for IPv4 and IPv6 is found to be about 3, 592, 000 bits/s and 3,708, 500 bits/s, respectively. In this case, the throughput for IPv6 is 3.14% higher than IPv4.

Now, turning to the third phase-3 (i.e., high load (offered traffic load (200%; 8Mbps); simulation time: 720-1800 s), it is apparent that the egress interface of the ingress router LER1 gets saturated to be served as a bottleneck link for all the traffic flows. This is attributed to the fact that the injected traffic load increases with the passage of time. Therefore, at the state of the congested circumstances, the throughput for IPv4 and IPv6 are mostly alike as it reaches the maximum link capacity of the followed route. From the discussed analysis and the obtained results reported in Table 5 and plotted in Fig. 5, it is reasonable to conclude that the average throughput for IPv6 is roughly 0.85% higher than its counterpart IPv4 under the varying network load.

Table 5. Summary statistics of throughput for scenarios 1 and 2

Phase-1(40% Network Load)			
	Forward Direction	**Scenario 1**	**Scenario 2**
Avg.(bits/s)	LER1←→LSR→	1,46E+06	1,51E+06
Std Dev.(bits/s)	LER1←→LSR4→	3,24E+01	6,15E+01
Phase-2 (90% Network Load)			
Avg.(bits/)	LER1←→LSR4→	3,59E+06	3,71E+06
Std Dev([bits/s)	LER1←→LSR4→	7,35E+02	7,27E+02
Phase-3 (200% Network Load)			
Avg.(bits/s)	LER1←→LSR4→	4,00E+06	4,00E+06
Std Dev.(bits/s)	LER1←→LSR4→	2,48E+02	2,68E+02

5. VALIDATION OF THE SIMULATION RESULTS

There is couple of techniques in relation to the validation, verification and testing (VV&T). In order to verify accuracy of the network models and interpreted simulation results we have considered analytical approach, which is described in section IV. In the course of simulation results verification, throughput performance obtained from the analytical analysis and the simulation is found to be almost similar. Therefore it can be ensured that network models and the simulation results are correct.

Furthermore, a detailed discussion on the subject of taxonomy of VV&T is presented in [23]. With the aim of validating of our simulation results, the techniques which have been used in the analysis are statistical outcomes and graph-based results obtained from previous work. The course of action in validating the simulation results offered in [24] [25] have also been followed in this work. Over the course of simulation analysis, the six experimental network models are replicated five times by changing the initial seeds and simulated keeping the confidence interval at ± 95%. While simulating replicated models in OPNET, it is found that both network models generate nearly same results.

6. CONCLUSION

In this thesis paper, we have evaluated the throughput performance of video/voice based on simulation and analytical methods in IPv4/IPv6 networks. Two network scenarios called IPv4 network and IPv6 network have been simulated. The simulation has been carried out by using OPNET. Comparative investigation of throughput performance based on simulative and analytical approaches was carried in both network scenarios.

Based on this research, research question was aimed to understand and investigate how much difference in the throughput is experienced in IPv4 over IPv6 network. The simulation results related to this question was presented in section 4 where the average throughput in the suggested IPv6 network was found to be roughly 3% higher compared with IPv4 networks. This is due to the fact that IPv6 has a bigger header size. Moreover, as we claim, that is not significantly enough to make a difference in the quality of video and voice traffic.

7. REFERENCES

[1] S. Deering and R. Hinden, “Internet Protocol, Version 6 (IPv6) Specification,” RFC 2460 (Draft Standard), Internet Engineering Task Force, Dec. 1998, updated by RFCs 5095, 5722, 5871. [Online]. Available: http://-www.ietf.org/rfc/rfc2460.txt

[2] A. Liakopoulos, D. Kalogeras, V. Maglaris, D. Primpas, and C. Bouras, “QoS experiences in native IPv6 networks,” International Journal of Network Management, vol. 19, no. 2, pp. 119–137, 2009.

[3] Y. Wang, S. Ye, and X. Li, “Understanding Current IPv6 Performance: A Measurement Study,” in 10th IEEE Symposium on Computer Communications,Jun. 2005.

[4] X. Zhou, R. E. Kooij, H. Uijterwaal, and P. van Mieghem, “Estimation of Perceived Quality of Service for

Applications on IPv6 Networks," in ACM PM2HW2N'06, Oct. 2006.

[5] D. P. Pezaros, D. Hutchison, F. J. Garcia, R. D. Gardner, and J. S.Sventek, "Service Quality Measurements for IPv6 Inter-networks," in 12th IEEE IWQoS, Jun. 2004.

[6] T.-Y. Wu, H.-C. Chao, T.-G. Tsuei, and Y.-F. Li, "A Measurement Study of Network Efficiency for TWAREN IPv6 Backbone," International Journal of Network Management, vol. 15, no. 6, pp. 411–419, Nov. 2005.

[7] K. Cho, M. Luckie, and B. Huffaker, "Identifying IPv6 Network Problems in the Dual-Stack World," in ACM SigComm Network Troubleshooting Workshop, Sep. 2004.

[8] S. Zeadally and L. Raicu, "Evaluating IPv6 on Windows and Solaris," IEEE Internet Comput., vol. 7, no. 3, pp. 51–57, May/Jun. 2003.

[9] W. Matthews. (2000, Oct.) IPv6 Performance and Reliability.[Online]. Available:http://www.slac.stanford.edu/grp/scs/net/talk/ipv6-oct00/IPv6-2000files/v3document.htm

[10] Y. Law, M. Lai, W. Tan and W. Lau, "Empirical Performance of IPv6 vs. IPv4 under a Dual-Stack Environment", in Proceeding of IEEE International Conference of Communication, 2008, pp. 5924-5929.

[11] C. Bouras, A. Gkamas, D. Primpas, and K. Stamos, "Quality of Service aspects in an IPv6 domain," in 2004 International Symposium on Performance Evaluation of Computer and Telecommunication Systems (SPECTS f04), San Jose, California, USA, 2004, pp. 238–245.

[12] R. Mondragon, A. Moore, J. Pitts, and J. Schormans, "Analysis, simulation and measurement in large-scale packet networks," Communications, IET, vol. 3, no. 6, pp. 887–905, 2009.

[13] P. Farrington, H. Nembhard, D. Sturrock, G. Evans, and X. Chang, "NETWORK SIMULATIONS WITH OPNET," 2009.

[14] OPNET. OPNET Modeler 14.0. [Online]. Available: http://www.opnet.com/solutions/network_rd/modeler.html

[15] R. Sargent, "Verification and validation of simulation models," in Proceedings of the 37th conference on Winter simulation. Winter Simulation Conference, 2005, pp. 130–143.

[16] A. Law, "Statistical analysis of simulation output data: the practical state of the art," in Simulation Conference, 2007 Winter. IEEE, 2008, pp. 77–83.

[17] OPNET, Modeler Release, 14th ed. [Online]. Available: http://www.opnet.com/solutions/network_rd/modeler.html

[18] K. Salah, P. Calyam, and M. Buhari, "Assessing readiness of IP networks to support desktop videoconferencing using OPNET," Journal of Network and Computer Applications, vol. 31, no. 4, pp. 921–943, 2008.

[19] J. Moy, "OSPF Version 2," RFC 2328 (Standard), Internet Engineering Task Force, Apr. 1998, updated by RFC 5709. [Online]. Available: http://www.ietf.org/rfc/rfc2328.txt

[20] R. Coltun, D. Ferguson, J. Moy, and A. Lindem, "OSPF for IPv6," RFC 5340 (Proposed Standard), Internet Engineering Task Force, Jul. 2008. [Online]. Available: http://www.ietf.org/rfc/rfc5340.txt

[21] P. Chimento and J. Ishac, "Defining Network Capacity," RFC 5136 (Informational), Internet Engineering Task Force, Feb. 2008. [Online]. Available: http://www.ietf.org/rfc/rfc5136.txt

[22] K. Salah, P. Calyam, and M. Buhari, "Assessing readiness of IP networks to support desktop videoconferencing using OPNET," Journal of Network and Computer Applications, vol. 31, no. 4, pp. 921–943, 2008.

[23] O. Balci, "Principles and techniques of simulation validation, verification, and testing," in Simulation Conference Proceedings, 1995. Winter. IEEE, 2002, pp. 147–154.

[24] K. Pawlikowski, H. Jeong, J. Lee et al., "On credibility of simulation studies of telecommunication networks," IEEE Communications Magazine, vol. 40, no. 1, pp. 132–139, 2002.

[25] A. Law, W. Kelton, and W. Kelton, Simulation modeling and analysis. McGraw-Hill New York, 1991, vol. 2.