



An Optimized Model for Transition from Ipv4 to Ipv6 Networks in a Cloud Computing Environment

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ABSTRACT

The paper proposes a novel optimized model for transition from IPv4 to IPv6 networks. The aim of this paper is to design the Original IPv6 Transition Controller Application and compare its performance with 6to4 tunneling using OPNET 17.5 with identical traffic and network loads. The analysis is based on an experimental design with simulation for better, faster and a more optimized solution through empirical measure of the process. The OPNET Modeler simulator was used to accurately model and predict the behavior of the real world system. The purpose of such detailed modelling efforts and analysis was evaluated using distinct performance metrics: data access (measured through response times of database queries), video conferencing (measured through video packet delay variations and end-to-end packet delays), and IP telephony voice communications (measured through jitters, mean opinion score, packet delay variations, and end-to-end packet delays). The paper presents the redesigned network model for ensuring that the IPv6 Transition Controller can control all the voice, video, and data sessions through “traffic recognition and routing” and also verifying the ToS status of the traffic and controlling the prioritization accordingly using IPv6 Transition App. The final results show that the model is optimized besides, focusing on traffic recognition, routing and prioritization, it ends up being the best design concept to achieve IPv6 deployment and high quality of service.

Keywords: *IPv4, IPv6, 6to4 Tunneling, NAT, OPNET, ToS.*

1 INTRODUCTION

The IPv4 protocol was introduced more than three decades ago with approximately 4 billion addresses. The addresses cannot cater for the needs of modern internet, with address depletion problem [1]. Some short-term antidote solutions were such as, such as Network Address Translator (NAT) or Classless Inter Domain Routing (CIDR), however work began on a new Internet Protocol, namely IPv6. The main reason for a new version of the Internet Protocol was to increase the address space. According to [2], detailed modelling and analysis of schemes for transiting from IPv4 to IPv6 in a cloud computing environment was carried out. The schemes modelled were dual-stack, IP tunneling, and network address translation.

The purpose of such detailed modelling efforts and analysis was to explore the performances of data access (measured through response times of database queries), video conferencing (measured

through video packet delay variations and end-to-end packet delays), and IP telephony voice communications (measured through jitters, mean opinion score, packet delay variations, and end-to-end packet delays). Special care was taken to ensure that the amount of traffic and network loads remained identical in the three scenarios. Further, it was found that network throughput remained comparable in the three scenarios, perhaps because the network is a cloud environment with high end servers and links of high capacities. From the simulation results, the following observations were made:

a) Dual Stack and NAT having comparable performances for Voice, but IP tunneling returned higher jitters, and more significantly, an unacceptable MOS value (at 2.5 while the acceptable range is between 3 and 4.2).

b) IP tunneling returned a stable performance of video with packet delay variation settling after a small initial variation. However, both dual stack

and NAT resulted in high initial variation before settling at almost the same value of video packets delay as that of IP tunneling.

c) The database query performances of dual stack, IP tunneling, and NAT are comparable.

Can these results be treated as empirical? Given that these results are obtained from modelling and simulations of a small-scale cloud with specific types of routers and servers, they cannot be generalized or treated as empirical. These performances may vary based on numbers and types of switches, routers, and servers.

The comparison reflects that IPv6 transition performance may vary based on many additional factors based on operating systems and their versions. The factors found in the studies compared are based on tunneling mechanisms, and also based on increase in density of customers. Perhaps, many more factors may be found in the future studies. Hence, the argument in previous paragraph against generalization and empiricism may hold to be true. There should not be any bias towards a particular IPv6 transition mechanism. Instead, there should be more design principles than merely evaluating the performance comparisons between the three IPv6 transition mechanisms. Keeping this aspect into account, the designing of original IPv6 Transition Controller Application has been executed in this research.

2 LITERATURE REVIEW

According to [3] comparing performances of dual stack, 6to4 tunneling, and NAT schemes of IPv6 transition by modelling them in Wireshark tool and testing round trip collision delay (latency) using PING and Trace Route processes. The findings returned dual stack with the highest latency compared with tunneling, and NAT schemes. The latencies of 6to4 tunneling and NAT were found to be comparable.

Compared performance of 6to4 tunneling scheme of IPv6 transition between Windows 2003 SP2 and Windows 2008 SP1 servers by testing throughput, jitter and average packet network delay for both TCP and UDP traffic types (configured on arbitrary ports) [4]. The TCP jitter through 6to4 tunnels was found as stable and almost identical in both the operating systems for all packet sizes. However, both the operating systems reflected very high jitters through 6to4 tunnels for small packet sizes that reduced sharply for medium packet sizes and then rose gradually for large packet sizes. In UDP the jitters were small and almost stable for small packet sizes. However, UDP jitter values increased gradually in both operating systems for low and

medium sized packets and returned slightly lower values in Windows 2003 SP2 as compared with those in Windows 2008 SP1 at larger packet sizes. The TCP average packet network delay through 6to4 tunnel in Windows 2008 SP1 was very high at about 1100 milliseconds while the same in Windows 2003 SP2 was found as close to zero at all packet sizes. The UDP average packet network delay through 6to4 tunnel in Windows 2008 SP1 was quite high between 200 and 600 milliseconds for smaller packet sizes, but it later settled between 50 and 150 milliseconds for large packet sizes. The TCP average packet network delay through 6to4 tunnel in Windows 2003 SP2 was almost constant, varying between 20 and 25 milliseconds. The TCP and UDP throughputs through 6to4 tunnels were found as exactly identical in both the operating systems. The TCP throughput varied from 40 to 70 Mbps for small packet sizes and 70 to 90 Mbps for medium to large packet sizes in both the servers. However, for small packet sizes UDP had a higher variation in throughput from 30 to 70 milliseconds in both the servers.

[5] Comparing performance of 6to4 tunneling scheme of IPv6 transition between Linux Fedora 9.10 and Linux Ubuntu 11.0 servers by testing throughput, jitter and average packet network delay for both TCP and UDP traffic types (configured on arbitrary ports). The TCP jitter through 6to4 tunnels was found almost identical in both the operating systems for all packet sizes. However, both the operating systems reflected very high jitters through 6to4 tunnels for small packet sizes that reduced sharply for medium packet sizes and then rose gradually for large packet sizes. In UDP, the jitter was found to be very high in Fedora 9.10 for small packet sizes but settled at comparable values with those in Ubuntu 11.0 for larger packet sizes. Both the operating systems returned an increasing trend of UDP jitters through 6to4 tunnels for large packet sizes. The TCP average packet network delay through 6to4 tunnels was found as close to zero at all packet sizes in both the operating systems. The UDP average packet network delay in Fedora 9.10 6to4 tunnel was quite high between 200 and 300 milliseconds for all packet sizes. The TCP average packet network delay through 6to4 tunnel in Ubuntu 11.0 was found as varying between 0 and 50 milliseconds. The TCP and UDP throughputs through 6to4 tunnels were found as exactly identical in both the operating systems. The TCP throughput varied from 40 to 70 Mbps for small packet sizes and 70 to 90 Mbps for medium to large packet sizes in both the servers. However, for small packet sizes UDP had a higher variation

in throughput from 20 to 70 milliseconds in both the servers.

In a cloud computing virtualization environment, three types of tunnels were configured using Teredo, ISATAP, and 6to4 and the means of throughputs, means and standard deviations of voice over IP jitters, means of end-to-end packet delay, mean of round trip collision delay (Ping process), mean of tunneling overhead, mean of tunnel set up times, and mean of DNS query delays were studied [6]. ISATAP was found to be comparatively better than 6to4 and Teredo in throughput, end-to-end packet delay, round trip collision delay, tunneling overhead, tunnel set up times, and DNS query delays. However, ISATAP was found comparatively poorer than 6to4 tunneling and Teredo in VoIP jitters. 6to4 tunnel was found to be better than Teredo in all variables except VoIP jitters in which, Teredo has the best performance. In general, 6to4 tunneling was found to be having sustained performance in all the performance variables.

In the research by [7], a scenario was configured in which, the clients on IPv4 needed to connect to servers on IPv4 through a cloud service provider on IPv6 only. Three configurations were studied: dual stack at the client and server ends, 6to4 tunnels crossing the cloud service provider, NAT centralisation and NAT distributions crossing the cloud service provider. The three scenarios were modelled in OPNET Modeler. This research found highest performance and reliability by NAT centralization. However, NAT cannot be preferred for high density communications as too many IPv4 addresses will be needed (one each dedicated for a running session). Hence, this research recommended 6to4 tunneling that appeared as second to NAT centralization in performance and reliability.

The research by [8] has similar setup to this research except that the HTTP, E-Mail, DB Query, and VoIP applications were studied for comparing the performances of dual stack, manual 6to4 tunneling, and automated 6to4 tunneling. Dual stack performed best in all the applications and automated 6to4 tunneling performed second to dual stack. Manual 6to4 tunneling performed the worst in voice jitters and MOS value.

3 RESEARCH METHODOLOGY

The paper employed experimental design to ascertain the transition strategies employed by Internet Service Providers running both IPv4 and IPv6. The experimental design with simulation was adopted for better, faster and a more optimized

solution through empirical measure of the process. There are so many objects in real or hypothetical life [9]. The goal of using any simulator is to accurately model and predict the behavior of a real world system [10]. Computer network simulation is often used to verify analytical models, generalize the measurement results, evaluate the performance of new protocols that are being developed and the existing protocols. Several types of simulations do exist for network simulation and modeling. These include discrete-event simulation (event-driven), continuous simulation, Monte Carlo simulation, trace-driven simulation, among others. For computer and telecommunication network simulation the most used method is discrete-event simulation. The discrete-event simulation has been used for research on all seven layers of OSI model. OPNET Modeler uses an object-oriented approach for the development of models and simulation scenarios.

4 OVERALL DESIGN AND ANALYSIS OF AN ORIGINAL APPLICATION FOR IPV6 TRANSITION

The research carried out by [2], tested automated 6to4 tunneling with dual stack and NAT and hence it was preferred in the final design. The criteria for this choice are as follows:

a) The IPv4 addresses are limited to about 4.3 billion and may get exhausted in future. Hence, in an ideal design all the equipment, servers, clients, and interfaces should be configured on IPv6. This is the primary reason an economic IPv6 transition method is needed.

b) Dual stack configuration requires equal number of IPv4 and IPv6 addresses and hence defeats the fundamental purpose for designing an optimal IPv6 transition method. It is not a choice even if it performs the best in most of the cases. Hence, while it may be used for small LANs already having IPv6 addresses, it is not suitable for large networks.

c) NAT may be good for small number of concurrent connections. However, when a large number of client machines are communicating, NAT will require a massive pool of IPv4 addresses and hence may not be suitable. In large networks, a large number of concurrent client connections is expected. Hence, NAT is judged as unsuitable for such networks. Given its lack of viability for long-term usage, it has been rejected in the design of this research.

d) Automated 6to4 tunneling has been recommended by all the research studies reviewed in the literature. The primary advantage of this

technology is that it requires only one IPv4 address per tunnel for unlimited number of concurrent sessions. However, encryption overheads cause increased voice jitters resulting in low MOS values, as was observed in this research.

e) However, if voice traffic and video traffic are segregated flowing to different server farms, then the jitters can be reduced improving the MOS value.

f) Voice performance of IP 6to4 tunneling can also be improved by prioritising voice traffic through Quality of Service (QoS) settings based on Type of Service (ToS).

Keeping the criteria highlighted above in mind, an original IPv6 Transition Controller Application design has been designed, modelled in OPNET, and tested through simulations. Figure 1 presents the redesigned network model for ensuring that the IPv6 Transition Controller can control all the voice, video, and data sessions through “traffic recognition and routing” and also verifying the ToS status of the traffic and controlling the prioritisation accordingly. The ToS status of the running traffic is determined with the help of a ToS DB having relational records defining traffic shapes and their priority levels. The IPv6 Transition Controller is directed by the IPv6 Transition App. The Servers SVR_1 and SVR_2 are now dedicated for data, SVR_3 and SVR_4 are dedicated for voice, and SVR_5 and SVR_6 are dedicated for video. There are additional servers and switches in the design as compared with the previous models.

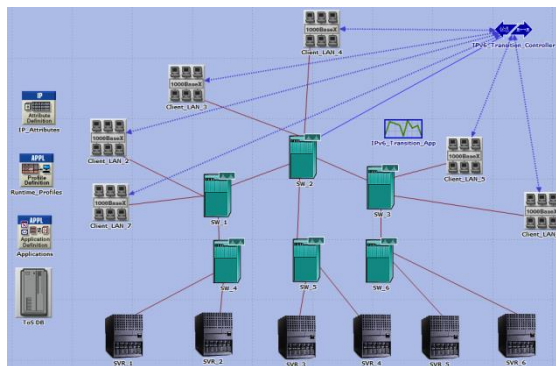


Fig. 1. Network reorganization for running the original IPv6 transition controller application design created in this research for IPv6 transition (Source: Researcher)

All the devices, interfaces, IP address schemes and allocation, and connecting devices (next hops) are connected appropriately with either IPv4, IPv6, or both. The network has now three distinct paths: SW_2 → SW_1 → SW_4 → SVR_1 / SVR_2 for data, SW_2 → SW_5 → SVR_3 / SVR_4 for voice, and SW_2 → SW_3 → SW_6 → SVR_5 /

SVR_6 for video. It may be observed that the number of hops for voice traffic path has been deliberately kept smaller keeping in mind the limitation of IP 6to4 tunneling in handling the voice traffic. Thus, the network is now reorganized to work for the IPv6 Transition Controller, and the IPv6 Transition App. The ToS DB, the IPv6 Transition Controller, and the IPv6 Transition App are new additions in the model. The runtime profiles and the applications are also modified based on the new application and its interactions as defined under the IPv6 Transition Controller and the IPv6 Transition App. This model and its performance analysis are the original contributions of this research study. The three server groups for data, voice, and video are now accessible only through IP 6to4 tunneling. The reasons are explained in the design criteria prior to the configuration and the model design.

It is noted that the ToS DB is a crucial component of this design, which is explained in detail later in this section. The IPv6 Transition App has three workflows of request-response interactions, each for traffic recognition and routing for data, voice, and video requests. The Client LANs are reconfigured (in the destination settings) to first consult the IPv6 Transition Controller before starting any traffic to the servers. Further, it is observed that the IPv6 Transition Controller first consults the ToS DB before responding to the requesting LANs. The ToS DB is the key component in this design for making a decision about traffic recognition and then routing it appropriately.

The entire client LANs are configured as destination preferences for responding to DB destination query (or may be simply named as data destination query). Similar settings are done for responding to voice destination query, and video destination query.

The ToS DB is an integral component of the IPv6 Transition Controller. Hence, the destination setting for sending query to the ToS DB points towards the transition controller itself. This is to ensure that all queries are resolved locally before the requests from the Client LANs are responded after making decisions about traffic recognition and then routing the traffic appropriately. The ToS DB records are presented with appropriate mapping of traffic classification schemes, the priority labels, and maximum queue sizes (similar to records in a relational database table).

There are four priority labels. The queue sizes are reducing with increase of priority labels. The streaming and interactive multimedia traffic are positioned at priority label “Medium” with a

maximum queue size of 40 and the interactive voice and reserved traffic are positioned at priority label “High” with a maximum queue size of 20. Thus, interactive voice and reserved traffic will have higher priority than streaming or interactive video. This means that there may be scenario when the video packets of a running interactive video session may be delayed but the voice gets through earlier. This may result in some time lag and lip syncing issues if there are longer queues of video packets; but the voice session will run smoothly. However, in case of multimedia streaming, both voice and video will be affected simultaneously if there is a longer queue of packets.

Another relational table in the ToS DB in which, the maximum bytes allowed, maximum queue lengths allowed, and the traffic classification schemes are mapped is also presented. In this ToS setting, applications with higher priority will be allowed higher number of bytes in the packets (that is, larger packet sizes will be allowed). As per the settings, voice and reserved categories will be allowed larger packet sizes such that even if their queues are long (say, reaching the maximum queue size of 20), the quality of application delivery will be better because of “more content delivered with each packet delivery”.

With a “reserved category”, a bandwidth of 5 Kbps and a buffer size (in switches) of 5 KB will be reserved per packet for each sender allowed in this category. This setting has been designed keeping a “most probable value” of the packet sizes in mind and may be increased if the applications are expected to send packets of higher sizes more frequently. For example, if the senders in “reserved” category tend to use their maximum quota of 16 KB more frequently (although unlikely; as the network is not designed to reach such high congestion levels), then the buffer reservation may be increased by this value and the bandwidth reservation may also be increased to 16 Kbps.

Finally, traffic limits are defined for average, normal, and excess bursts. If the excess traffic bursts occur on any type of service, application type, or TCP/UDP port, it will be treated as congestion threshold and any further traffic will be dropped. This explains why it is unlikely for the reserved category to send traffic with maximum packet sizes. This rule of traffic limits is applicable for any sender irrespective of the priority labels and privileges. It is like if a motorway is congested, even the vehicles of VVIPs will be stopped at a barrier.

OPNET has in-built parameters for load generation based on such levels. A designer can create custom loading profiles as well, based on

knowledge of loading profiles in an actual organization. However, capturing loading profiles from actual networks requires multiple packet capture probes for data collection, which can collectively capture thousands of packets of different applications and determine the loading profiles running on the network. It is very difficult to get such permissions from an organization for capturing packets. Hence, to demonstrate the design working in a simulation environment OPNET’s internal loading profiles are used. The runtime profile for the IPv6 transition controller application used in the simulation is presented and the duration is fixed at 50 seconds with an offset of 5 to 10 seconds. In reality, the duration shall vary dynamically based on the number of requests received and processed by the IPv6 transition controller application. After completing the above settings, multiple simulations were executed, presented and analyzed.

5 SIMULATION RESULTS’ ANALYSIS OF THE ORIGINAL APPLICATION FOR IPv6 TRANSITION

The simulation results presented include IPv6 transition controller application and the comparisons with dual stack, 6to4 tunneling, and NAT modelled, simulated, and analyzed.

5.1 IPv6 Transition Controller Application Operations Simulation Results’ Analysis

The analysis of operations of the IPv6 transition controller application is presented. Figure 2 presents average phase response times of the three tasks of the application: traffic recognition and routing of data, voice, and video traffic. The phase response times have been returned as 1 second in the three phases. This response time is unavoidable because the response steps involves running a quick query on the ToS DB before making and communicating the routing decision to the requester.

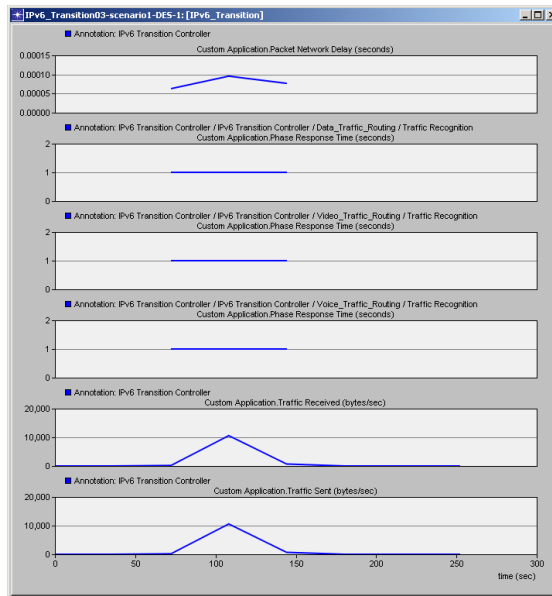


Fig. 2. Packet network delay, response times, and overall traffic of each of the three phases of IPv6 transition controller application executed for traffic recognition and routing of data, video, and voice traffic (Source: Researcher)

This 1 second may extend further if there is a long queue of requests to be processed. However, this should not affect the performance of the overall network because execution of these phases will be needed only once before the traffic volumes are triggered.

As presented in Figure 2, the phases are completed within a short period of between 75 and 150 seconds of the simulation. The curve is triangular indicating that the peak workload of processing the requests occurred between 100 and 110 seconds of the simulation. Once the requests have been processed, there is no traffic related to IPv6 Transition Controller Application. In real world, this triangular pattern may repeat randomly with varying peaks as the requests from clients are expected to arrive randomly in a stochastic manner.

The overall packet network delay for executing the IPv6 Transition Controller Application reported in Figure 2 is 0.1 Milliseconds. This delay is very small and is independent of the actual data, voice, and video delays and will add to the phase execution time for executing the requests. In addition, there are additional delays to be accounted that are caused by introducing the application in the network.

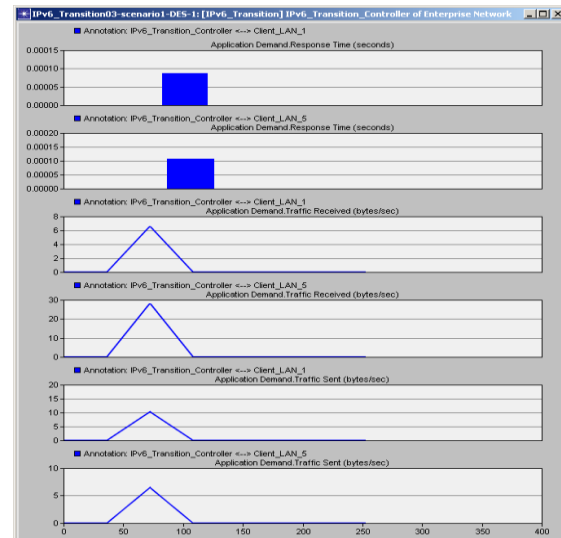


Fig. 3. Response times and traffic of initial application demands raised by two of the Client LANs to the IPv6 transition controller application executed for requesting for traffic recognition and routing of data, video, and voice traffic (Source: Researcher)

Figure 3 presents the delays caused by a few clients in making the requests to the application. These delays are again very small (approximately 0.1 Milliseconds) but will add to delay in actual traffic starting. The traffic volumes are very small for making the requests (between 6 to 30 bps).

Figures 4 to 7 further shows the overall scenario of the client requests made to the IPv6 Transition Controller Application. For every request made, the application sends an acknowledgement to each client such that the client does not repeat the request. Thereafter, the client has to wait till the application provides traffic routing information for the requested traffic: data, voice, video, or a combination of the three that may result in a session split into multiple streams routed through different paths on the network.

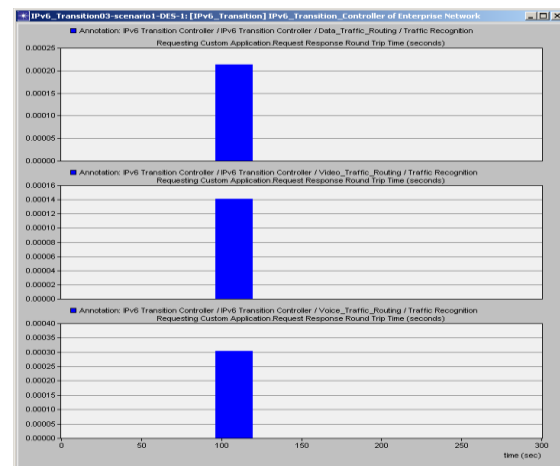


Fig. 4. Request-Response round trip times of each of the three phases of IPv6 transition controller application executed for traffic recognition and routing of data, video, and voice traffic (Source: Researcher)

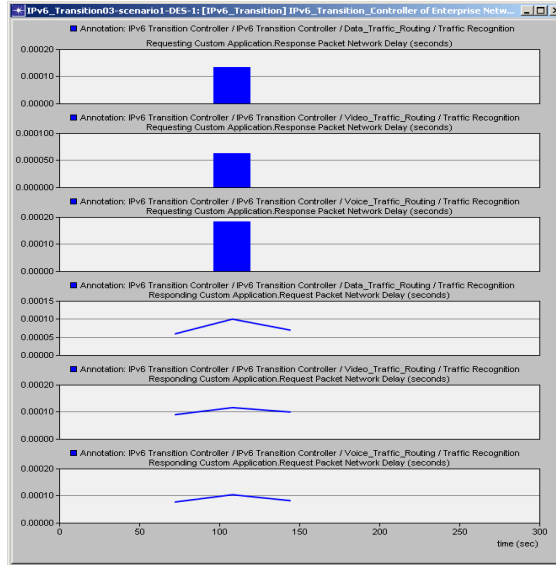


Fig. 5. Request and response packet network delays of each of the three phases of IPv6 transition controller application executed for traffic recognition and routing of data, video, and voice traffic (Source: Researcher)

Figure 4 presents the round trip delay of the request-response cycles being a maximum of 0.3 milliseconds for the requests and decisions made to route data, voice, and video traffic. Further, the traffic routing decisions for the client is based on the query output of the ToS DB. Hence, after this round trip delay, a waiting period for getting the final confirmation from the application is added.

Figure 5 further shows the overall packet network delays of both the request and response cycles. The requests (as observed in Figures 3, 4, and 5) are of short durations (maximum 30 seconds) and the responses are the ones responding with full traffic routing information thus extending to the maximum duration of 75 seconds as observed in Figures 2 and 3.

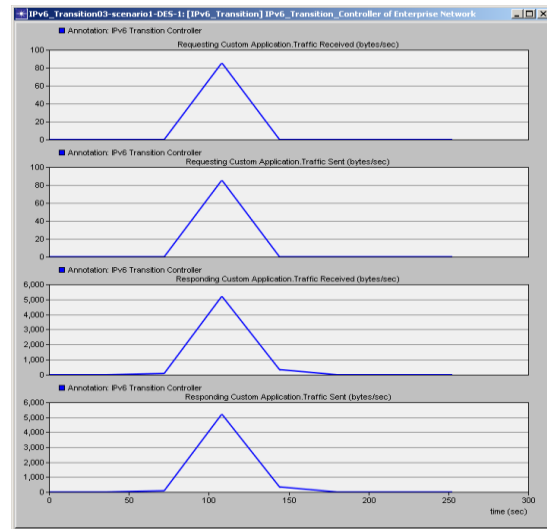


Fig. 6. Overall requests and response traffic to/from the IPv6 transition controller application executed for traffic recognition and routing of data, video, and voice traffic (Source: Researcher)

The maximum packet network delays are recorded at a slightly above 0.1 milliseconds for the overall requests and responses pertaining to routing decision-making of data, voice, and video traffic. The overall traffic for completing the cycles of requests and responses is as presented in Figure 6. The requests from clients comprise of small traffic volumes: between 80 bps to 90 bps. However, the responses from the IPv6 transition controller application comprise of larger traffic volumes: between 5.0 to 5.5 Kbps.

The last simulation report analyzed in this part of simulations is related with the size of packets in the requests and the requests per second handled by the application (Figure 7). The requesting packet sizes are fixed at 1000 bytes. Further, the number of requests per second handled by the application peaked at five requests per second in this simulation.

The request sizes are the same as the minimum sizes defined in the application because there are no overheads in this simulation period. This is because the peak number of requests per second handled by the application is six only. This is why the overall round trip delay of the request response cycles peaked at 0.3 milliseconds only.

In real world networks, there shall be hundreds or even thousands of requests per second for the application. Hence, a single instance of this application may be able to handle the overall load with acceptable performance. For example, keeping in mind the benchmark of 0.3 milliseconds round trip performance for a peak load of six requests per second, the application may be able to handle 6000

requests per second at a round trip request-response time of 300 milliseconds.

Combining with the response time performance of ToS DB query of one (1) second, the average total time taken for the application to complete the processing of a request will be 1.3 seconds. This performance will be quite acceptable because it will occur only once before establishment of the data, video, or voice session. In complex networks, like grid and cloud computing, this application may have to be run in multiple instances such that each instance can undertake the services overhead of its neighbouring virtualized machines.

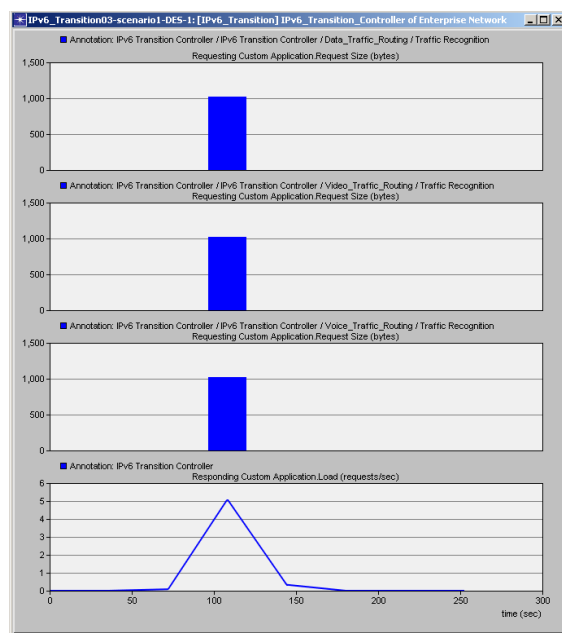


Fig. 7. Overall sizes and load of all requests sent to the IPv6 transition controller application for traffic recognition and routing of data, video, and voice traffic (Source: Researcher)

5.2 Simulation Results' Analysis related to comparison of operations of the IPv6 Transition Controller Application with the previous three IPv6 Transition Designs

The performance comparison with the previous three designs of IPv6 transition is discussed. The simulations have been conducted to collect the results for the 13 performance parameters and later collected for the three IPv6 transition scenarios: dual stack, IPv6 tunneling, and IP NAT. The first performance is related with overall TCP delay and TCP segment delay (Figure 8 compared with dual stack, IP 6to4 tunneling, and NAT).

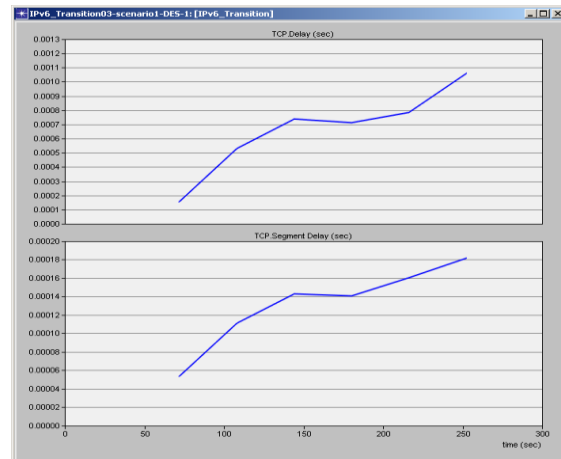


Fig. 8. TCP performance after completing the three phases of IPv6 transition controller application

(Source: Researcher)

The overall TCP delay at a peak value between 0.738 and 1.058 milliseconds is better than dual stack, IP tunneling, and NAT but quite close to the 1.2 milliseconds recorded in IP tunneling. But, TCP segment delay is almost equal to dual stack, slightly higher than IP Tunneling, and slightly lower than IP NAT.



Fig. 9. Database performance after completing the three phases of IPv6 transition controller application (Source: Researcher)

The next performance comparison is with regard to database query performance (Figure 9 compared with dual stack, IP 6to4 tunneling, and NAT). The peak value of DB query response is at 16.66 milliseconds, which is considerably lesser than that in dual stack (34.93 milliseconds), IP tunneling (34.711 milliseconds), and IP NAT (34.991 milliseconds). The traffic volume is the same as that of the three models indicating fair comparisons.

The video conferencing performance of the traffic controlled through the IPv6 transition controller

application is presented in Figure 10 and is compared with dual stack, IP 6to4 tunneling, and NAT. The initial peak of packet delay variation is about 0.859 milliseconds. From this peak, the packet delay variation dropped sharply to about 0.117 milliseconds and then gradually dropped 0.035 milliseconds (almost zero). Similar pattern was observed for video conferencing packet delay variation in dual stack, IP tunneling and IP NAT.

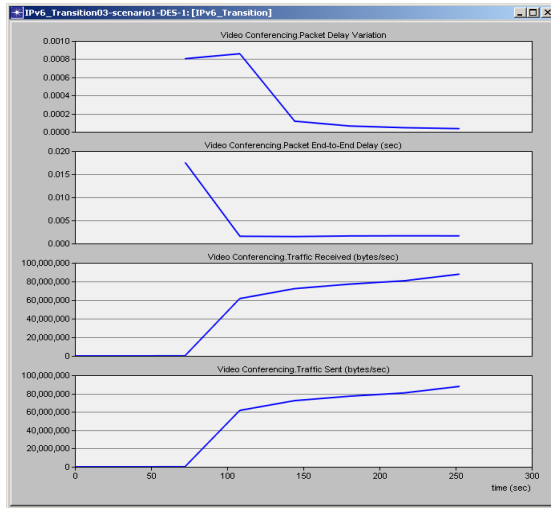


Fig. 10. Video performance after completing the three phases of IPv6 transition controller application (Source: Researcher)

The video conferencing end-to-end packet delay decreased sharply from a peak of 17.573 milliseconds to about 1.644 milliseconds and stabilized at this level. Similar pattern was observed in the three previous IPv6 transition models. Hence, it may be safely concluded that the video conferencing traffic controlled through the IPv6 transition controller application performed at par with the previous three models (dual stack, IP tunneling and IP NAT). The video conferencing traffic volumes reached between 87.768 Mbps and 77.241610666667 Mbps in all the four design scenarios and hence, the comparisons made are fair.

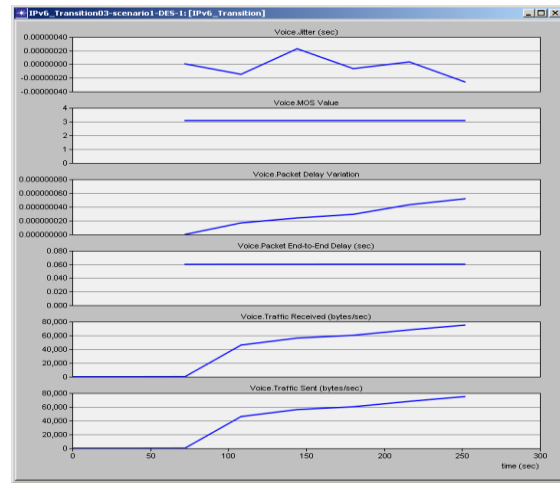


Fig. 11. Voice performance after completing the three phases of IPv6 transition controller application (Source: Researcher)

The summary statistics of the performance parameters comparison against simulation time for 6to4 tunneling (TNL) transition scheme and IPv6 transition controller application (IPTC) is depicted in Table 1.

Table 1: The performance parameters comparison summary statistics against simulation time for dual stack, 6to4 tunneling, Network Address Translation transition schemes, and IPv6 transition controller application.

Scheme	Simulation Time (sec)			
	TCP Delay (sec)			
	1m 12s	1m 48s	2m 24s	3m 0s
TN L	0.000	0.0008	0.0010	0.0012
	455	35	73	03
IPT C	0.000	0.0005	0.0007	0.0007
	158	31	38	12
TCP Segment Delay (sec)				
TN L	0.000	0.0001	0.0001	0.0001
	069	33	53	65
IPT C	0.000	0.0001	0.0001	0.0001
	054	11	43	41
DB Query Response Time (sec)				
TN L	0.033	0.0341	0.0345	0.0347
	509	78	45	11
IPT C	0.004	0.0166	0.0167	0.0155
	74	7	3	5
DB Query Traffic Received (bytes/sec)				
TN L	7,395	242,20	219,09	258,85
	.55555	4.444444	3.333333	8.666667
IPT C	1,123	233,15	216,96	216,16
	.56	9.11	0	3.56
DB Query Traffic Sent (bytes/sec)				

TN L	7,395 .55555 6	242,20 4.444444	219,09 3.333333	258,85 8.666667
IPT C	1,123 .56	233,15 9.11	216,96 0	216,16 3.56
Video Conferencing Packet Delay Variation				
TN L	0.000 096	0.0007 77	0.0000 65	0.0000 42
IPT C	0.000 805	0.0008 59	0.0001 17	0.0000 64
Video Conferencing Packet End-to-End Delay (sec)				
Tun neling	0.006 277	0.0023 31	0.0020 14	0.0019 13
IPT C	0.017 573	0.0015 63	0.0015 01	0.0016 28
Video Conferencing Traffic Received (bytes/sec)				
TN L	21,12 0	58,905, 600	72,771, 840	77,238, 720
IPT C	54,72 0	61,595, 520	72,270, 720	77,086, 080
Video Conferencing Traffic Sent (bytes/sec)				
Tun neling	21,12 1.7777 78	58,909, 559.1111 11	72,771, 847.1111 11	77,241, 610.6666 67
IPT C	55,68 8.89	61,597, 552	72,268, 807.11	77,089, 928.89
Voice Jitter (sec)				
TN L	0.000 000521	0.0000 00084	- 0.000000 311	- 0.000000 050
IPT C	0.000 000005	- 0.000000 147	0.0000 00227	- 0.000000 064
Voice MOS Value				
TN L	2.517 819300	2.5179 18703	2.5179 18703	2.5179 18703
IPT C	3.080 691	3.0807 8061	3.0807 8061	3.0807 8061
Voice Packet Delay Variation				
TN L	0	0.0000 00026	0.0000 00035	0.0000 00040
IPT C	0	0.0000 00017	0.0000 00024	0.0000 00029
Voice Packet End-to-End Delay (sec)				
TN L	0.100 034885	0.1001 44073	0.1001 70904	0.1001 98920
IPT C	0.060 044	0.0601 1545	0.0601 34424	0.0601 59949
Voice Traffic Received (bytes/sec)				
TN L	125.8 3333333 3	44,598. 33333333 3	54,457. 5	60,070
IPT C	113.3 333	45,973. 61	56,113. 33	60,158. 89
Voice Traffic Sent (bytes/sec)				

TN L	125.8 3333333 3	44,598. 33333333 3	54,458. 33333333 3	60,069. 16666666 7
IPT C	113.3 333	45,973. 61	56,113. 33	60,158. 89

Lastly, voice performance of the traffic controlled through the IPv6 transition controller application is presented in Figure 11 and is compared with dual stack, IP 6to4 tunneling, and NAT. There is a finite voice jitter and packet delay variation although the magnitude values are too small to be quantified. Hence, the tangible indicators are MOS value and end-to-end voice packet delay. The MOS and end-to-end voice packet delay are almost identical with the dual-stack model and IP NAT model (3.08078061 and 60.213991 milliseconds) but are noticeably better than the IP tunneling model (2.5 and 100.672 milliseconds). The maximum voice traffic was between 75.031666666667 Mbps and 57.5227777777778 Mbps in all the four models indicating a fair comparison.

It may be recalled that IP 6to4 tunneling was found to be short of acceptable voice performance in the simulation of its voice traffic and analysis. However, IP 6to4 tunneling has been used for data, voice, and video in the optimum design model controlled by the IPv6 transition controller application and still has achieved the performance levels identical to dual stack and NAT. This change has happened because of introducing the IPv6 transition controller application and the ToS DB supporting it. Based on the initial operations analysis and performance comparisons of data, voice, and video with dual stack, IP6to4 tunneling, and IP NAT designs, a critical analysis and summarization is presented.

6 CRITICAL ANALYSIS AND SUMMARIZATION

The original design and modelling of the IPv6 transition controller application was presented. Further, the simulation results were presented in two parts: simulation of the operations of the application and simulation results comparison of data, voice, and video performance with those of dual stack, IP tunneling, and IP NAT model designs.

Has the design of IPv6 transition controller application successfully achieved optimum performances of data, voice, and video communications? At this stage, it appears to be the case. The optimum choice for IPv6 transition is IP 6to4 tunneling given that it employs the least of IPv4 addresses and is very easy to configure and manage. Also, the application has been successful

in achieving results at par with dual stack and IP NAT although IP 6to4 tunneling was found to return relatively inferior voice performance than the other two.

However, the analysis of results of some of the recent studies enumerated in the literature revealed that the results of simulation in a single research study cannot be taken as a benchmark for achieving empirical and optimized results, especially when there is an attempt to ratify a design. Those studies revealed that results obtained through simulations should only be accepted as indicators such that further studies can be conducted to investigate their reliability, validity, and conformability (to the real world networking and the experiences of experts). Hence, a brief analysis of reliability, validity, and conformability of the results is discussed before setting the ground for validation of the findings.

Reliability of a research reflects the consistency of findings if the same methods, techniques, tools, and environment is replicated by another research [11]; [12]. This research is conducted using the academic edition of OPNET Modeler, which is a recognized academic and professional network modelling and simulations software used extensively in advanced networking research [13]; [14]. In addition to recognized corporations and networking vendors, a large number of universities use OPNET for advanced networking education and research [13]. It is affirmed that any researcher following the modelling process, traffic settings, node and link settings, choice of products, and application-level designs and advanced settings as defined in this research will be able to get comparable results. However, this claim is limited to another academic study conducted by an academic researcher using OPNET. This claim cannot be extended to the practical networking systems solely by academic researchers because there is no point claiming confidence on the correctness and accuracy of a network simulation tool (even if it is popular) without inviting criticism and judgement by industry experts.

Validity of a research reflects the relevance of the variables and their measures used in the study such that the research may be perceived as having achieved outcomes based on standardized and trustworthy measures [12]. In this research, the variables and their measures chosen for studying the performances of the four models, as presented are standard as per the networking literature. For example, TCP delays and TCP segment delays are worldwide recognized measures for studying the performance of TCP protocol in a network. Again, how OPNET might have handled these variables and their measures is subject to its reliability and

industry wide acceptance. The academic researchers cannot claim its validity without inviting criticism and judgment by industry experts.

Conformability of a research depends upon three secondary variables: fitment to the subject area, creditability, and auditability [15]; [16]. Fitment to the subject area can be achieved by ensuring transferability of findings from the sample to the population studied with high credibility. Creditability can be established when interested users of the research can recognize the outcomes and relate with their experiences. Auditability of a research can be ensured by maintaining all records and documents created during the progress of the research such that they can be audited by the supervisor of the research and an external examiner.

Review of the three variables is indicating the need for evaluation of the design by industry experts. Input from experts is the only way to establish whether the IPv6 transition controller application is optimal or not. As per experience from comparison conducted in the literature, the outcomes may vary a bit in academic studies depending upon the experimentation environment, test bed used, tools and techniques used, and such other variations. However, experts from industry can directly relate the design and its findings with their experiences and similar applications in action (if any). Keeping this projection in mind, a survey of networking experts was planned and executed using an online survey website. Based on the survey a series of quantitative analysis was done.

7 VALIDATION OF FINDINGS

The reliability, validity, and conformability of the final design is tested with the help of statistical evaluation of the structural used in the design. The data related with the construct is collected through an online survey from Kenyan and Indian IPv6 experts. There were 27 respondents from Kenya (34.6%) and 51 respondents from India (64.4%). The total number of respondents was 78. The survey was sent to 106 individuals through LinkedIn contacts. Hence, the response rate is 73.58%. It is evident that the networking experts surveyed have viewed the final design very critically. The statistical modelling report presented has revealed different results for the four dependent variables, which are analyzed as follows:

a) Bandwidth Performance (BANDP): Variance in BANDP (mean 2.6923, SD 0.67) is found to be statistically significantly influenced by service plan SERVP (mean 3.3718 SD 0.56), network convergence NETWC (mean 2.0897, SD 0.776), and load balancing LOADB (mean 3.73, SD 0.446). Thus, good practical validity of LOADB and

SERVP and fair practical validity of NETWC in the final design are expected to cause significant variations in BANDP at medium level. The experts appear to be impressed with LOADB and SERVP of the final design but their response reflect medium bandwidth performance on the network. The network convergence is low (obviously) because the three forms of traffic (data, voice, and video) are accessible through three independent network paths in the final design. Confidence in service plan seems to be good because of inclusion of ToS DB in the design. When the moderators are applied, only hardware compatibility (mean 3.1282, SD 0.7954) appeared to be influencing these relationships statistically significantly. Hardware compatibility is rated as good in the final design by the experts. The SD in all the variables is quite low indicating high consensus of the mean values among the respondents.

b) Throughput Performance (THRP): Variance in THRP (mean 3.8462, SD 1.082) is found to be statistically significantly influenced by NETWC (mean 2.0897, SD 0.776) and LOADB (mean 3.73, SD 0.446). In spite of ToS DB in place, the experts appear to be unconvinced about influence of service plan on throughput performance. Perhaps, this is because the links have limited bandwidth in the final design and may get overloaded when subscribers' count increases. This anyways can be improved when the ISPs increase the links capacities by adding more switch-to-switch links. Interestingly, technical expertise (mean 4.48, SD 0.5) is rated as excellent in the final design by the experts. They seem to be impressed with the detailed technicalities and their practical feasibility in the final design. Further, technical expertise is found to be a moderator for influencing throughput performance. It appears they are seeking role of good technical experts in design changes (like adding more switch-to-switch links) and network management. It appears natural; technical experts will always support role of technical expertise. One observation is that there is a high SD in THRP. This means that some of the respondents deviated further from the mean, either above or below. This reflects some differences in opinion although not significant enough to impact the outcome.

c) Latency Performance (LATP): The experts appear to be highly critical about LATP (mean 4.0769, SD 0.6982) as its variations are influenced by variations in transition mechanism TRANS (mean 3.7692, SD 0.8962), NETWC (mean 2.0897, SD 0.776), LOADB (mean 3.73, SD 0.446), and SERVP (mean 3.3718 SD 0.56) (from regression analysis). Further, the moderators have a significant influence on LATP as TRANS (mean 3.7692, SD

0.8962) is replaced by physical connectivity PHYC (mean 3.0897, SD 0.7) and all the three moderators HARDC (mean 3.1282, SD 0.7954), TECHE (mean 4.48, SD 0.5), and IMPLC (mean 3.73, SD 0.696) have induced statistically significant variations in LATP (mean 4.0769, SD 0.6982). HARDC and TECHE indicates good choice of servers and switches and good technical expertise influencing high latency performance (which is also reflected in the final simulations). However, the experts also indicated between medium to high implementation cost in achieving this design and the latency performance. Further, replacement of TRANS by PHYC after applying the moderators is slightly confusing. Perhaps, the experts want to give more weighting to physical connectivity as against the IPv6 transition mechanism when the three moderators are considered. Perhaps, this also implies more technical expertise to be hired increasing the implementation cost because the current links capacities are inadequate.

d) Jitter Performance (JITTP): JITTP (mean 4.0128, SD 0.7976) is found to be statistically significantly influenced by TRANS (mean 3.7692, SD 0.8962), ADDP (mean 3.5385, SD 0.5510), and NETWC (mean 2.0897, SD 0.776). This is the first time ADDP has appeared as an influencing variable. However, statistical significance of its influence reduces below 95% after IMPLC (mean 3.73, SD 0.696) takes over as a moderator, which is obvious because IMPLC will be very nominal in this addressing scheme. IPv6 tunneling and the strategy of using IPv4 and IPv6 addresses were found to achieve good jitter performance in the final simulations. The experts have rated jitter performance as excellent. This means that the performance reported in this design has exceeded their expectations. The situation may be poorer in real networks.

8 CONCLUSION

The conclusions of the paper have been drawn based on key findings from the survey analysis and in the context of experiences gained from the simulations. These are from the evaluation and perspectives of the networking experts that have studied the design and its performance statistics generated in OPNET. The key conclusions are:

a) In the proposed design, bandwidth performance shall be influenced significantly by service plan, network convergence, and load balancing (among the IPv6 tunnels based on predictive prioritization, which is the key aspect of the proposed design). The networking experts were quite impressed with load balancing and service

planning of the final design, but their responses reflected that bandwidth performance on the network needs to improve further. Subsequently, they found the design to be low at network convergence as the three service types are accessible through three different IPv6 tunnels. However, this is more of a trade-off than a limitation because the IP tunneling performances for data, voice, and video improved significantly at the cost of reducing network convergence. This may not affect the client because the IPv6 transition controller senses the traffic type and routes them to different IPv6 tunnels. Thus, the destination requests by a mobile subscriber may be split and routed through multiple tunnels ensuring optimum performances for all the traffic types.

b) The networking experts were not convinced about the throughput performance of this design influencing service quality. This is because the network capacity in the test bed modelling is much lower than that in the networks by real world ISPs. This limitation may be overcome simply by investing in higher bandwidth links and increasing their numbers on each of the switches.

c) The networking experts treated technical expertise as a good moderator to improve throughput performance. This reflects that they seek more technical thoughts to improve the network capacity.

d) The networking experts were very critical about latency performance of the design. Their responses reflected that latency performance shall be affected significantly by service planning, load balancing, network convergence, and transition mechanism. They have tried to emphasize that compromising network convergence at the cost of improved load balancing may result in varying latencies for different service types. This may happen because there may be different loadings on different IPv6 tunnels at the time of service requests. Hence, it is possible that a user may experience excellent video performance but relatively lower performance of the file transfer protocol (FTP). A general perception may be that H.264 protocol in MP4 streams has been prioritized over FTP. However, the real reason will be the difference in loadings on the IPv6 tunnels leading to the two service destinations. This will make service quality unpredictable if the design is not enhanced further.

e) The networking experts were highly impressed with the jitter performance. As revealed in simulations of the single IPv6 tunnel configuration, the jitter performance was poorer than the proposed multi-tunnel design. Given that all the results were shared with the experts, they

have tried to communicate that they are impressed with the multi-tunnel configuration and want more enhancements in load balancing and network convergence. They have also projected a need for enhancing physical connectivity, perhaps to ensure that the IPv6 tunnels are divided equally among multiple parallel physical links. They have further reflected appreciation for implementation cost as it proved to be a moderator in influencing jitter performance. It needs to be noted that jitter performance is a very critical aspect as voice requires real time traffic flow with high prioritization. Interactive voice performance can only be achieved with quality of experience if it is awarded the highest priority. In this design, the problem of such prioritization has been solved as voice services can be reached through dedicated IPv6 tunnels, just like data and video. Hence, the challenge is not in prioritizing voice traffic over data and video albeit is to prioritize among multiple voice sessions running parallel.

9 FURTHER RESEARCH

This research has significant scope of future studies. The setting of the research test bed was moderate keeping in view the limitations faced by an academic researcher. However, the test bed can be enlarged significantly in future studies either by using other simulators or by implementing test beds with real world networking equipment. In addition, experts from more countries should be invited to scrutinize test results from the larger test beds. In real world, there may be thousands of IPv6 tunnels in action following the design and its principles proposed in this research. Hence, there is also a scope of artificial intelligence in predictive load analysis going much beyond advisory services of ToS DBs. This aspect may also be examined in future research studies.

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