

Performance Evaluation of OSPF and EIGRP Routing Protocols for Video Streaming over Next Generation Networks

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Abstract

The challenge in performance of Next Generation Networks (NGNs) is a growing concern. Bandwidth limitation for multimedia applications in NGNs such as voice and video telephony along with the increasing number of applications on the Internet, service classification and efficient resource management have all become quite challenging tasks. In order to improve the service quality in video streaming especially, there has to be efficient resource and traffic management, and using routing protocols is one way in which this can be done. We designed three network models that are configured with OSPF, EIGRP and one with both OSPF and EIGRP routing protocols, and then used the QoS parameters of throughput, packet loss, convergence time, mean latency and end-to-end delay as our metrics to evaluate the performance of OSPF & EIGRP. Our main source of network traffic was a typical video conferencing application. The results obtained during our experiments indicate that combining both EIGRP and OSPF is more reliable in providing Quality of Service than OSPF routing protocol when the main traffic used in the network is video, but when dealing with a standalone real time application network, EIGRP is better than OSPF.

Keywords: NGNs, Video Streaming, Routing Protocols, OSPF, EIGRP.

1. Introduction

With recent trends and technology advancement in the development of converged broadband next generation networks (NGNs) and advanced multimedia services, the potential has increased for video streaming services to end users “anywhere, anytime” using the World Wide

Web. A wide variety of these services do exist, this is mainly due to the availability of various systems to deliver the services. These systems are built on top of tools and applications that provide the necessary communications and computer-aided support (e.g., multimedia conferencing/streaming enablers, image analysis and visualization tools, immersive and collaborative virtual environments). In this paper we limit ourselves on evaluating the performance of key protocols using one of the mechanisms used to guarantee efficient delivery of real-time video services to users – *Routing Protocols*.

The rest of the paper is organized as follows: Section 2 presents a background study on the key concepts in this paper, section 3 details the test bed environment and set up, section 4 presents a discussion on the results obtained from the experiments and we finally conclude the paper in section 5.

2. Background

This section presents a brief background on Next Generation Networks, video streaming in NGNs and the main routing protocols used to implement Quality of Service in these networks.

2.1 Next Generation Networks (NGNs)

A Next Generation Network is an interesting innovation that mainly drives to reduce costs on the side of service providers while at the same time enhancing the capability

of a given network to stay open to new services and applications. This innovation basically involves the transformation of public switched telephone networks (PSTN) which are circuit-based networks into packet-based networks that mainly depend on Internet protocol. Therefore, it is one of those innovations to change the telecommunication industry forever.

The development of NGNs has further led to yet another concept – convergence, this represents the shift from the traditional ‘vertical silos’ architecture i.e. a scenario where services were provided through different networks (mobile, fixed, IP) to a situation where communication services are accessed and used seamlessly across different networks and provided over different platforms in an interactive way[1]. The biggest driving force behind this has been the Internet and it has stayed liked that.

Converged NGNs deliver different types of traffic across heterogeneous end-user environments [2]. For example, video and audio streaming have special bandwidth, loss and delay requirements, where data or a video fails to arrive in expected time, play out in a particular application may pause, this becomes annoying to the user. Therefore, in order to meet the requirements of a specific video or audio service traffic delivered over networks in conjunction with other commercial traffic, appropriate routing protocols and QoS mechanisms such as class-based traffic prioritization are necessary. The wide variety of video services imposes different Quality of Service (QoS) requirements on underlying networks. One aspect is delay tolerance, with service requirements ranging from strict real-time and delay-intolerant data transmission to delay-tolerant services. However this paper focuses on evaluating two routing protocols with regard to their performance towards video streaming across Next Generation Networks.

2.2 Video Streaming

Video streaming is the real time delivery process of video to the user's media player services [3]. Normally, streamed video takes on the notion of being played as it downloads, compared to when the user first downloads the video and then plays it on their video application. In the ‘real time delivery process’, the video has various requirements with regards to delay, and bandwidth. Fig 1 is a typical depiction of video streaming architecture.

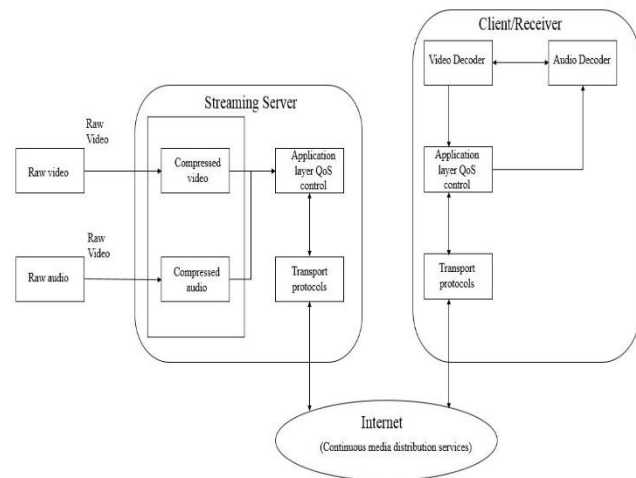


Fig. 1 Video streaming architecture [4]

2.3 Routing protocols

Routing can be defined as transmitting information from a source to a destination by hopping one-hop or multi hop [5]. Routing is carried out by devices known routers whose major objectives are; making routing decisions, construct routing tables to facilitate communication and sharing information amongst neighboring routers in the network. One way in which these objectives are achieved is through routing algorithms, the latter use a variety of metrics as a way of determining the best path of reaching a preferred network.

A routing protocol can either be a distance vector routing protocol or a link state routing protocol; distance vector protocols operate mainly by passing duplicates of their routing tables to neighboring routers while link state routing protocols rely on advertising a list of available neighboring routers to a point where all the routers in the networks have a duplicate of this list; with this information the routers then run algorithms to analyze and select the best path available to transmit the packets.

In this paper, the protocols presented are OSPF(Open Shortest Path first) and EIGRP(Enhanced Interior Gateway Routing Protocol), these are the protocols we used in our experiments (section III), and were evaluated based on the quantitative metrics of convergence time, packet delay variation, end to end delay, traffic sent, traffic received and packet loss. These metrics are the ones we used to further evaluate the performance of 3 different network

environments from which results and conclusions were drawn.

- *Open Shortest Path First(OSPF)*

OSPF is an Interior Gateway Protocol (IGP) which is one of the main protocols used in the Internet Protocol (IP)-based Internetworks. The routing protocol is a public (open standard) that is based on the link state. The various concepts and operations of the OSPF link state are fully described in Request for Comments (RFC) 1583.

OSPF (Open Shortest Path First) is an interior gateway protocol. OSPF is a classless link state protocol. OSPF uses the SPF (Shortest Path First) algorithm to calculate the cost. SPF works in tree structure to calculate the cost from root. Root is the router from which cost is calculated to other routers. This algorithm is known as the Dijkstra's algorithm[6].

$$Cost = \frac{10^2}{bandwidth(bps)} \quad (1)$$

In any given network, a router is supposed to measure the cost involved to route to neighboring routers. This cost measure is mainly used as a metric to find the best route. The cost of a route is calculated using the formula shown in Eq. (1).

- *Enhanced Interior Gateway Routing Protocol (EIGRP).*

EIGRP is an interior gateway protocol developed by Cisco Systems and introduced with Software Release 9.21 and Cisco Internetworking Operating System (Cisco IOS) Software Release 10.0. It is a suitable protocol for a variety of network topologies and multimedia. As we shall later on find out in the simulation stage, EIGRP is one protocol that has quick convergence time. The protocol is considered to be one that lets routers exchange information more efficiently than other network protocols.

EIGRP has five building blocks, these are; neighbor route tables, topology tables, route tagging, route states and routing tables [7].

To compute the routing metrics, EIGRP mainly depends on minimum path bandwidth to the destination network and the total delay. Bandwidth and delay metrics are determined from values configured on the interfaces of

routers in the path to the destination network. Eq. (2) is used to scale the bandwidth to the destination network.

$$bandwidth = \left(\frac{100000000}{bandwidth(i)} \right) * 256 \quad (2)$$

Where *bandwidth (i)* is the least bandwidth of all outgoing interfaces on the route to the destination network represented in kilobits [8]. In order to scale delay, EIGRP employs Eq. (3).

$$delay = delay(i) * 256 \quad (3)$$

Where *delay (i)* is the sum of the delays configured on the interfaces, on route to the destination network in tens of microseconds. Then, having bandwidth and delay scales, EIGRP can now determine the total metric to the entire network. Eq. (4) is used to do this[8].

$$metric = \left[K_1 * bandwidth + \frac{K_2 * bandwidth}{256 - load} + K_3 * delay \right] * \left[\frac{K_5}{reliability + K_4} \right] \quad (4)$$

This is when $K_4 \neq 0$ and $K_5 \neq 0$

BUT when $K_4 = K_5 = 0$,

$$metric = \left[k_1 * bandwidth + \frac{(k_2 * bandwidth)}{256 - load} + k_3 * delay \right] \quad (5)$$

But, the Constant values for *k* are:

$$k_1 = 1, K_2 = 0, k_3 = 1, K_4 = 0 \text{ and } K_5 = 0$$

Therefore, when there is a default or constant behavior the formula becomes;

$$metric = bandwidth + delay \quad (6)$$

3. Testbed environment

OPNET modeler was chosen to be used in the simulation phase of this work. OPNET Technology is the developer of this simulator software and it is considered one of the most powerful software in simulation. The main reason is because one can easily design and study either small or large scale networks with the various devices in these networks, including protocols, applications and above all, the processes of the nodes in the overall network. To sum it up, as indicated earlier, OPNET is a simulator built on

top of a discrete event system and it simulates the system behavior by modeling each event in the system and processes it through user defined processes

3.1 Testbed Set up

The network topology was designed with 5 subnets; the first subnet was deployed in the first city which also acted as the headquarters of the network, the other 4 subnets were deployed in 4 other different cities. This particular network topology was duplicated into three other topologies in order to implement three network scenarios with different routing protocol configuration i.e. one with the OSPF scenario, another with the EIGRP scenario and the last one was configured with a combination of the two routing protocols.

The network topology was made up of different devices and different configuration utilities. These included: The subnet at the headquarters city had a Video Ethernet server, an Http Server, a Voice Server, a fire wall and a router, a switch and a LAN with Ethernet workstations. The other Subnets in the other Cities only had a Video Server but with other components.

To generate our network traffic, we used a video conferencing application with high resolution video and for back ground traffic, heavy Http browsing and Voice applications were used. Below is the full list of the various device and configuration utilities.

- I. The routers are connected with the PPP_DS3 Duplex Links;
- II. Ethernet 100BaseT Duplex Links that connect the various 100BaseT Ethernet workstations LANS to the Switch.
- III. A Failure Recovery Configuration Utility.
- IV. An Application Configuration Utility.
- V. Profile Configuration Utility.
- VI. QoS Attribute Configuration.

4. Results and discussion

We analyzed the following metrics in order to evaluate the performance of the routing protocols in the different network scenarios; Convergence Time, End-to-End Delay, Packet Loss, Traffic Sent and Traffic Received of both the video conference application and back ground traffic applications, i.e. Voice and Http.

4.1 Convergence time

Convergence is the time it takes all the various routers in a network to share specific information, of which this time should always be at a minimal.

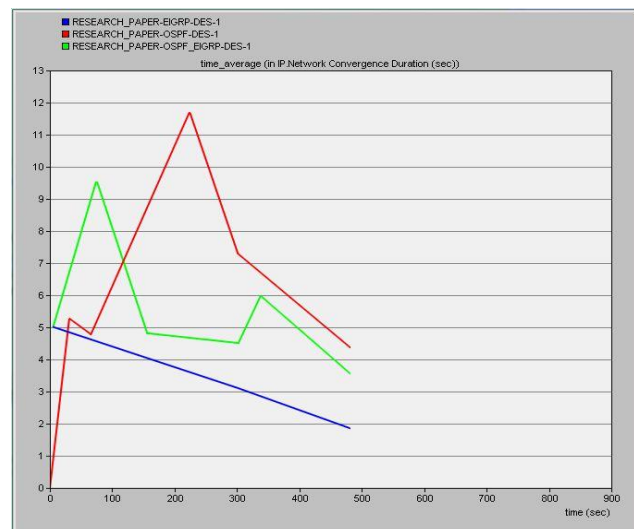


Fig. 2 Average Convergence Time

Table 1. Average Convergence time (secs)

Scenario No.	Scenario Name	Routing Protocol	Convergence Time(sec)
1 st	EIGRP	EIGRP	1.8600
2 nd	OSPF	OSPF	4.384
3 rd	OSPF_EIGRP	OSPF& EIGRP	3.572

In Fig. 2 and Table. 1, we observed that during convergence, the EIGRP network is faster than OSPF and OSPF_EIGRP networks. This was mainly because of the high rate at which EIGRP detected the rapid changes within the network and further communicated the changes with other neighboring routers until all the routers in the network were updated, this change was because of our failure recovery configuration where one of the links was

set to fail and recover at 300 and 480 seconds respectively. Between OSPF and OSPF&EIGRP combined networks, the latter network was slower than OSPF

4.2 End-to-End delay

End-to-end delay is the elapsed time for a packet to be passed from the sender through the network to the receiver and that the higher the delay between the sender and receiver, the more insensitive the feedback loop becomes, and therefore, the protocol becomes less sensitive to short term dynamic changes in the network.

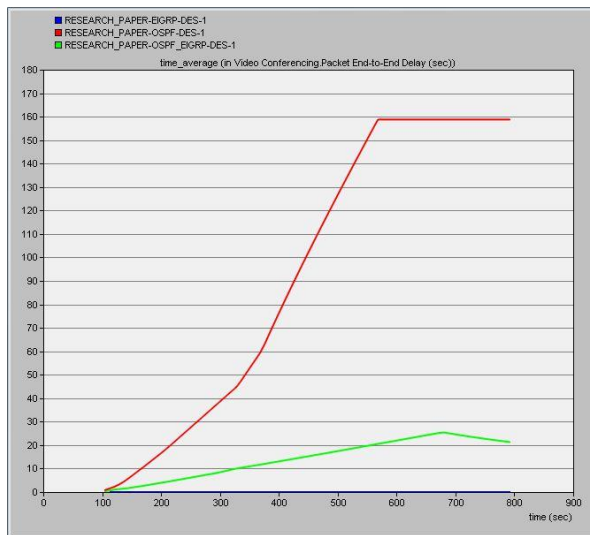


Fig. 3: Video Conferencing End-to-End Delay

Table 2. Video Conferencing End-to-End Delay

Scenario Name	End-to-End Delay (sec)
OSPF_EIGRP	21.304
OSPF	158.78
EIGRP	0.0022578

So, looking at Fig 3 along with Table 2, the OSPF_EIGRP network highest end-to-end delay compared to OSPF and EIGRP which had the lowest End-to-End and it was network congestion that brought about this result. Basically end-to-end delay mainly depends on the speed of the network and the degree of network congestion and traffic.

4.3 Packets Loss

Packet loss refers to the inability of transmitted packets to get to their desired destination node or network, in our

results the network scenario configured with only EIGRP dropped the least packets and the OSPF protocol configured network dropping the most packets. Figure 4 and Table 3 represent the Average Packet Loss of the networks.

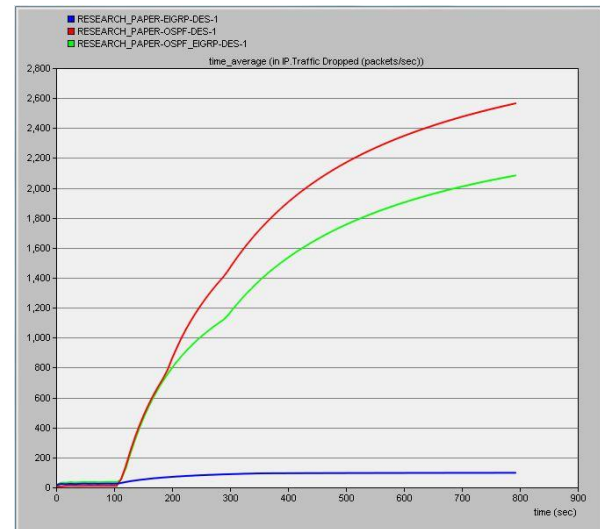


Fig. 4: Average Packet Loss (Secs)

Table 3. Average Packet Loss (Secs)

Scenario Name	Average Packet Loss (sec)
OSPF_EIGRP	2,083.3
OSPF	2,564.7
EIGRP	97.33

4.4 Video Conferencing Traffic Sent

As earlier on discussed in section 3, a video conferencing application was our main source of traffic for our simulation experiments. It generated the traffic throughout the various scenarios. The video resolution of this application was set at a high resolution with 15 frames/secs as the frame inter - arrival time. The frame size of 128x240 pixels was used and the Type of service set at Best effort. The users of the network in the various cities had video streaming servers from which they were accessing the video from. Figures 5 and 6 illustrate the total traffic sent and received from the video conference applications. OSPF and OSPF_EIGRP combined networks experienced a high degree of Packet loss, this further confirms the low rate at which EIGRP loses packets in the

network, Tables 3 and 4, present the details of Video Conferencing Traffic sent and traffic received.

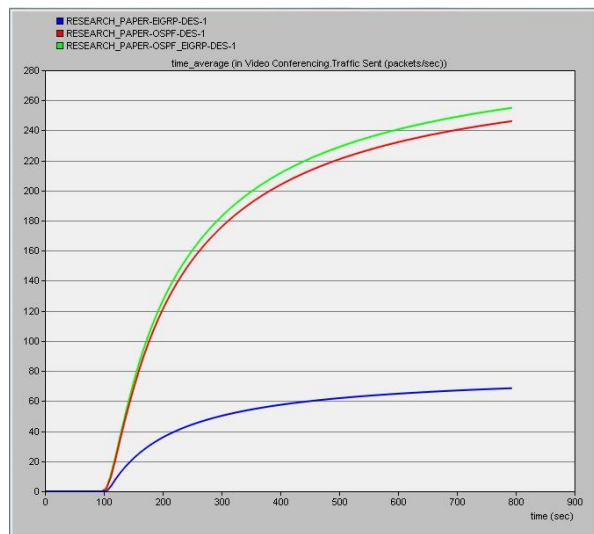


Fig. 5: Video Conferencing Traffic Sent (packets/sec)

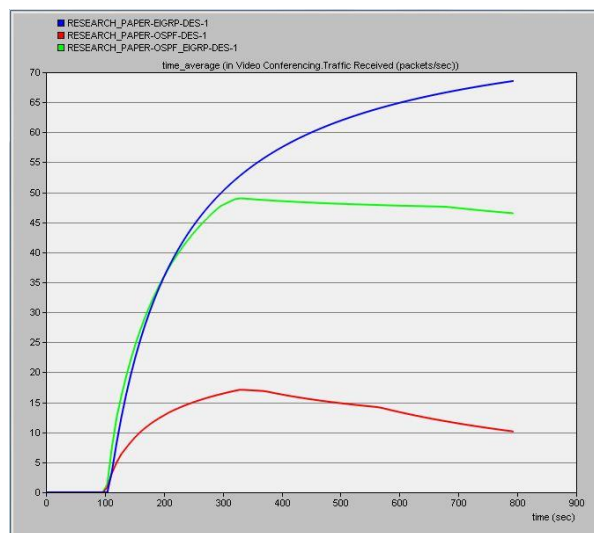


Fig. 6: Video Conferencing Traffic Received (packets/sec)

Table 4. Video Conferencing Traffic Sent/Received (Packet/sec)

Scenario Name	Sent(packets/sec)	Received (packets/sec)
OSPF_EIGRP	254.97	46.509
EIGRP	68.545	68.540
OSPF	246.12	10.149

4.5 Voice traffic sent

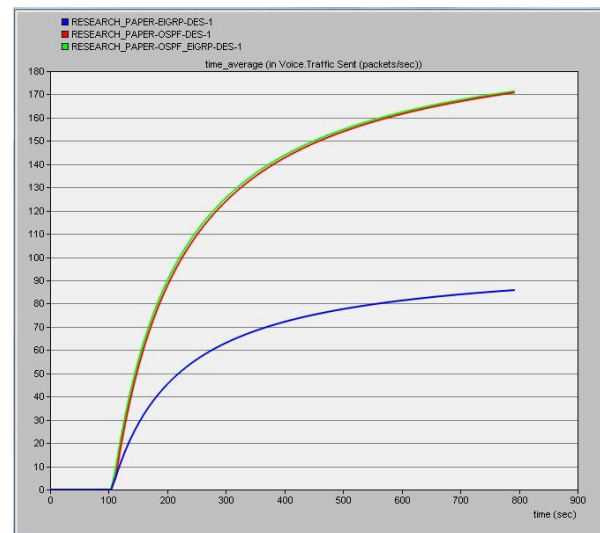


Fig. 7: Voice Traffic Sent (packets/sec)

4.6 Voice Traffic received

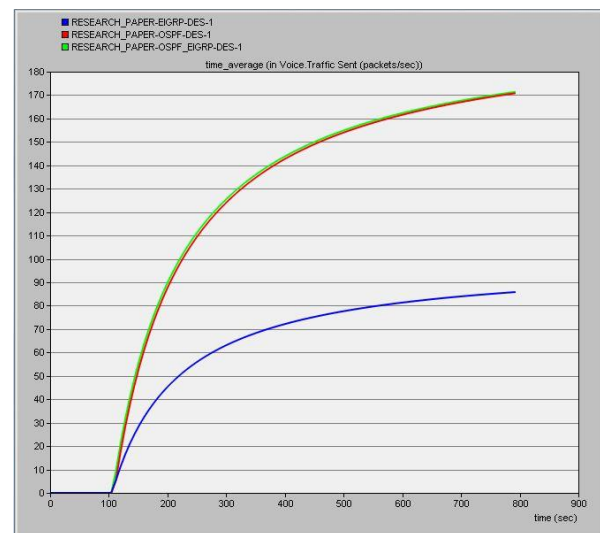


Fig. 8: Voice Traffic Received (packets/sec)

Table 5. Voice Traffic Sent and Traffic Received

Scenario Name	Voice Sent(packets/sec)	Voice Received (packets/sec)
OSPF_EIGRP	171.40	171.40
EIGRP	85.825	85.825
OSPF	170.83	170.83

5. Conclusion

In our study, it can be concluded that EIGRP has a much faster convergence time than OSPF and OSPF_EIRGP networks and this was because EIGRP is a fast protocol when it comes to accessing a network's topology information updates compared to the other protocols which tend to struggle.

The results generally confirmed that combining both EIGRP and OSPF together especially in heterogeneous networks is bound to assure QoS in these networks since they host a multitude of time sensitive applications, but if a network developer chooses to focus on a single real time application network, then EIGRP is better routing protocol than OSPF in terms of guaranteeing the desired QoS of a network this was further evidenced in figures 7,8 and Table 5 showing Voice traffic sent and Voice traffic received.

In future, we intend to explore and find out whether these protocols are bound to produce the same results in a different network environment such as a wireless network based mainly on IPv6 due to the fact that this study was based on a wired network with IPv4 environment.

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