Evaluation of Video Transmission Quality of Service over Next Generation Networks

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Abstract: The challenge in service quality for the performance of Next Generation Networks (NGNs) is a growing concern. The 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 handle the different types of applications in the network to improve the service quality in video and voice transmission, 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 their throughput, packet loss, convergence time, packet delay variation and end-to-end delay as our performance evaluation metrics. Our main source of network traffic was a typical video conferencing application. The results obtained during the simulation indicates 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.

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 delivering video 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).

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, in scenarios 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, 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.
2 Related work
In order to deliver multimedia across NGNs, there are various requirements that have to be met especially for specific real-time/interactive video service traffic delivered over networks in conjunction with other commercial traffic (e.g., voice calls, streaming multimedia, and Internet traffic). QoS mechanisms such as class-based traffic prioritization and effective IP routing protocols are necessary. The wide variety of multimedia and 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. In [3], [4], [5], [6], and [8] the authors categorize the importance of various QoS parameters for different interactive video services. Prioritization and resource allocation schemes for various types of video traffic delivered over various networks have been addressed in [3, 8]. Further studies have more specifically focused on evaluating support for the delivery of real-time video services over high-speed 3G/4G networks [9],[12, 10] and other types of broadband networks [11], with evaluation results showing generally reliable performance.

The research community has continued to carry out various investigations on new approaches, methods, strategies, techniques and tools for analyzing, designing, controlling and evaluation of future Next Generation Systems that support user interworking of multimedia applications and their mobility. Emphasis is now being given to QoS and its related aspects both in access and core networks in the presence of multimedia traffic, as [12] suggested. The real question above all is how can we evaluate QoS of service especially in an instance where it is impacted by an unexpected event? This may sound pretty an easy question to answer but that isn’t the case. We do not differ so much from [12], when he stated that “the only good reason to measure anything is to reduce uncertainty with respect to some course of action that must be decided”. Hardy [12] proposed and presented a method where he analyzed QoS using two processes, namely measurement and evaluation. In our study we focus more on evaluating the performance of two routing protocols based on selected metrics, we also evaluate these protocols but from a network routing point of view. Routing is part of network traffic engineering, the major objective of the latter is to ensure and improve network performance at the same time maintain the QoS requirements by optimizing network resources. One way this can be done is through the application of efficient IP routing protocols that will enhance the general performance of the network especially when it comes to QoS of the various multimedia applications.

3 Network Convergence versus Network Quality of Service
Due to the increase in the digital content on the Internet, more and more people, institutions and businesses are shifting their activities to the Internet through creating IP-based networks, applications and services. This is mainly because there has been an increase in access to high-speed broadband, availability of computing power and devices along with the communication media discussed later on in the work. All the above activities have been a major contribution to the utilization of the “convergence” term when discussing next generation networks.

Convergence is seen as a shift from traditional ‘vertical silos’ architecture i.e. a situation in which different services were provided through separate networks to a situation in which these services are accessed and used seamlessly across different networks and provided over platforms in an interactive way[4]. The different levels of convergence include Network convergence, Service convergence, Industry/market convergence, Legislative, institutional and regulatory convergence, Device convergence and Converged user experience.

Quality of Service as defined by the International Telecommunication Union Telecommunication Standardization Sector (ITU-T) Recommendation is the “collective effect of service performance which determines the degree of satisfaction of a user of the service”. As discussed earlier on, the next generation networks hosts a wide variety of service applications for multimedia transmission across the networks. All these applications and networks have their own desired level of service standards as far as the users of those applications and the service providers themselves are concerned. This therefore calls for a systematic way of realizing these objectives, and this is mainly achieved through putting in place mechanisms to control network traffic, allocate and manage network resources. Most researchers believe that QoS is strongly correlated to the main characteristic of a network – its heterogeneity [13].
4 Next Generation Network Challenges

Next generation networks are meant to support a wide variety of network traffic services and user mobility that is constraint free at the same time ensuring that there is a guaranteed Quality of Service for users at any given time and anywhere. However this doesn’t come cheap, there are quite a number of challenges that need to be addressed. The challenges are:

4.1 Application Traffic Types

The diverse nature of network traffic comes with a recognized challenge to any network engineer in their quest to guarantee quality of service to the users of that network in terms of Quality of Experience. In multimedia traffic, there are mainly two types of network traffic - real-time/interactive and non-real time. Examples of real time/interactive traffic are video and audio while file downloading is an example of non-real time traffic. Network traffic from real time/interactive applications are very sensitive to delay, whereas those applications that are non-real time may not have strict delay requirements, although they do not accommodate errors at the same level as the real time/interactive applications. One of the major aims of any network is to sufficiently meet the quality demands of all classes of applications and therefore guarantee Quality of Service in these applications. Therefore the major requirement of a next generation network is the ability to support a wide variety of services and applications with their respective traffic features, for instance diversity in delay and requirements in bandwidth.

With the evolving diversity in network traffic types, QoS provision within next generation networks is a real challenge, thus many researchers are now apportioning more resources to this cause and this is also one of the major aims of this thesis, though the slight difference here is that this thesis focuses on real time/interactive QoS video transmission over next generation networks.

4.2 Traffic Characterization

Traffic characterization is a way of dealing with traffic flow by determining its impact on the general performance of the network. The biggest concern for network engineers here is controlling data burstiness and the network characterization itself. Data burstiness is part and parcel of the different IP Networks’ patterns. Much as the area network traffic modeling and characterization continues to become a research area of concern, this thesis doesn’t proceed to venture into this area.

4.3 Protocol Specific

Next Generation Networks are IP based which means that they mainly rely on core network protocols. Much as Internet protocol may guarantee more scalability than Asynchronous Transfer Mode ATM, it does come with various challenges. One of the main problems is the QoS provision [16]. The Internet Engineering Task Force (IETF) working group is further evaluating and analyzing the various ways that QoS on IP can effectively be implemented.

4.4 Costing

One of the objectives of NGNs is to minimize on communication costs but there is always going to be a challenge of higher bandwidth costs and transmission rates. This is yet another challenge that various researchers are putting their heads together to strike equilibrium of service costing without affecting the QoS and the users at large.

4.5 Network Capacity

The evolution of telecommunication networks was fuelled by the demand for higher capacity. The continued upward trend in technology especially in telecommunications has further increased the rate at which multimedia applications are being developed. All these applications have a desired amount of bandwidth to ensure efficiency in service. Therefore, Next Generation Networks need to be able to meet the bandwidth capacity of the various multimedia applications. If all these factors are to be considered (which is always the case), then network designers and programmers are faced with many challenges especially when it comes to traffic engineering and network dimensioning.

4.6 Mobility Management

Heterogeneous networks have to promote mobility. The issues of global mobility in the core IP network are being solved by the introduction of the Mobile IP protocol [17] but Mobile Internet protocols also come with their own challenges for example triangular routing, and duplication of IP fields (“IP within IP”). Researches have emphasized that in mobility, the handovers from different cells should be properly handled to maintain the desired QoS [18].
4.7 Scalability
Scalability is the degree of adaptability to a particular force of demand, thus the challenging question network engineers have to ask themselves is - will the network be able to support an increase in the number of users at any given one time? This is entirely because the number of people using telecommunications has continued to increase thus the NGN should be able to meet the demand without affecting the Quality of Service.

4.8 Heterogeneous network Compatibility
A reliable next generation network should be compatible with different types of networks currently in use and must have provision for future developments. Diversity in a network comes with key issues that shouldn’t miss mentioning, and these include: Mobility management within applications of these networks, compatibility with other networks, secure interworking within access networks. End-to-end QoS of service guarantee to users of these networks. Various research groups are addressing the compatibility issues for the networks to work together [17].

5 Evaluation of Video Quality of Service
One way of guaranteeing effective video quality of service in next generation networks is through network traffic routing. Routing is a process where routers in a network specify paths that the various packets should follow during transmission across the Internet [18]. Routers and Routing protocols are at the center of this process; the latter are the various rules that specify how the routers are going to communicate with each other by disseminating data. There are quite a number of routing protocols widely in use in the telecommunication industry, this work concentrates mainly on Open Shortest Path First (OSPF) and Enhanced Interior Gateway Routing Protocol (EIGRP) as one of the ways of evaluating video transmission QoS in Next Generation Networks; this is mainly because they are the pre-eminently considered protocols for video transmission in a network.

5.1 Protocols under focus
The protocols presented in this paper are OSPF and EIGRP. These are the protocols we used in our experiments (section 8), and they are 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.

I. 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.

II. 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.

5.2 Experiment set up for Network Topology
Figure 1 is a depiction of the 4 subnets (Cape Town, Durban, Johannesburg and Mafikeng). This particular figure was duplicated thrice in order to implement the three network scenarios with different routing protocol configuration. The network topology was made up of different devices and different configuration utilities. These included:

I. The four subnets, each subnet with a video Ethernet server, LAN with Ethernet workstations and a CS_7000 Cisco router;
II. The routers are connected with the PPP_DS3 Duplex Links;
III. Ethernet 10 Base T Duplex Links that connect the various Ethernet workstations in the LANS together;
IV. Failure Recovery Configuration Utility;
V. Application Configuration Utility;
VI. QoS Attribute Configuration and a Profile Configuration Utility.
5.3 Evaluation and experiments results

During the experiment conducted, we set up and simulated 3 different network scenarios i.e. OSPF scenario, EIGRP scenario and in the third scenario both OSPF and EIGRP are combined. A video conferencing application was configured to generate the main traffic of the network. The sections that follow present the results of the experiment.

5.3.1 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.

<table>
<thead>
<tr>
<th>Scenario Name</th>
<th>End to End Delay (ms)</th>
</tr>
</thead>
<tbody>
<tr>
<td>OSPF_EIGRP</td>
<td>0.43171</td>
</tr>
<tr>
<td>OSPF</td>
<td>0.43194</td>
</tr>
<tr>
<td>EIGRP</td>
<td>0.63589</td>
</tr>
</tbody>
</table>

Table 2: Average of end to end delay

5.3.2 End to End delay (video conferencing)

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.

So, looking at figure 3 along with Table 2, the OSPF_EIGRP network had a lesser end-to-end delay compared to OSPF and EIGRP 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. Table 8-3 below shows the average values of end-to-end delay of the different networks.

Table 1: Average value of convergence time

<table>
<thead>
<tr>
<th>Scenario No.</th>
<th>Scenario Name</th>
<th>Routing Protocol</th>
<th>Convergence Time(SEC)</th>
</tr>
</thead>
<tbody>
<tr>
<td>1st</td>
<td>EIGRP</td>
<td>EIGRP</td>
<td>0.5473</td>
</tr>
<tr>
<td>2nd</td>
<td>OSPF</td>
<td>OSPF</td>
<td>3.7552</td>
</tr>
<tr>
<td>3rd</td>
<td>OSPF_EIGRP</td>
<td>OSPF_EIGRP</td>
<td>5.0001</td>
</tr>
</tbody>
</table>

Table 2: Average of end to end delay
5.3.3 Packet Delay variation (video conferencing)

Packet delay variation refers to the difference in end-to-end delay of the packets. This variation sometimes brings about an effect known as jitter. This variation is measured by taking the difference in delay of the packets. Figure 4 illustrates video conferencing packet delay variation.

Figure 4: Video Conferencing Packet Delay variations

Table 3 below shows the results of packet delay variations of the different networks in our simulations. It was observed that the OSPF_EIGRP network records a less packet delay variation than OSPF and EIGRP networks; that is why OSPF and EIGRP individually had a high packet delay variation.

<table>
<thead>
<tr>
<th>Scenario Name</th>
<th>Packet Delay variation (ms)</th>
</tr>
</thead>
<tbody>
<tr>
<td>OSPF_EIGRP</td>
<td>0.18749</td>
</tr>
<tr>
<td>OSPF</td>
<td>0.21466</td>
</tr>
<tr>
<td>EIGRP</td>
<td>0.22186</td>
</tr>
</tbody>
</table>

Table 3 Packet Delay Variations (ms)

5.3.4 Video Conferencing Traffic Sent

A video conferencing application was our main source of traffic in 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/sec 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 areas had video streaming servers from which they were accessing the video from. Figure 8-4 illustrates the total traffic sent from the video conference application.

Figure 5: Video Traffic Sent

The Figure also shows a total traffic of approximately 45688129 bytes per seconds through the network.

5.3.4 Video Conferencing Traffic Received

The figure above illustrates a small percentage drop of packets because of the network congestion. Table 4 shows that EIGRP experiences the least packet loss in comparison with OSPF_EIGRP and OSPF networks that experience almost the same amount of packet loss.

<table>
<thead>
<tr>
<th>Scenario Name</th>
<th>Sent(bytes/sec)</th>
<th>Received(bytes/sec)</th>
<th>Packet Loss</th>
</tr>
</thead>
<tbody>
<tr>
<td>EIGRP_OSPF</td>
<td>2333436</td>
<td>2333479</td>
<td>0.02%</td>
</tr>
<tr>
<td>EIGRP</td>
<td>2333436</td>
<td>233405</td>
<td>0.01%</td>
</tr>
<tr>
<td>OSPF</td>
<td>2333436</td>
<td>233265</td>
<td>0.07%</td>
</tr>
</tbody>
</table>

Table 4: Average value of sent and received (bytes/sec) for video.

6 Conclusion and Future work

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.

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 network based mainly on IPv6 due to the fact that this study was based on a network with IPv4 environment.

References


