Achieving Quality of Service in Medium Scale Network Design Using Differentiated Services

by

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Abstract

Quality of service (QoS) means packets are classified and sent to destination based on priority of the packet. Before the advent of this standard data packets were sent on a standard namely "Best effort". In this standard packets were sent on the policy of first come and first serve basis without providing reliability, bandwidth or latency. This often results congestion on the router due to the load of queue, packets were dropped due to congestion issues. The rise of multimedia application defines a need for a new standard which guarantee bandwidth with low delay and jitter. Multimedia applications like VoIP, Video conferencing are delay sensitive and cannot survive on the " Best effort" therefore we require some sort of differentiators that can detect these different types and appropriately prioritize and queue them for effective transmission and this transmission is achieved with a new standard known as Quality of Service (QoS). Quality of Service is achievable by different types namely RSVP, RSVP-TE, MPLS and differentiated services. The main objective of this project is to explain how a medium scale network can be redesigned to implement quality of service within the network. Real time simulations for multiple performance factors are obtained and implemented into a sample network to achieve the desired results.

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Acknowledgments

In the name of Allah, the Most Gracious and the Most Merciful

All praises belong to Allah the merciful for his guidance and blessings to enable me complete this project. I would like to thank:

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Dedication

To my father, **Shamim Khan** and my mother, **Sajida Shamim** for having a lifelong long dream to see me achieve my graduate qualification at a world class foreign institution. In difficult times, it proved as key motivating factor and enabled me to maintain focus.

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Abbreviations

Quality of Service (QoS)

Differentiated Services (Diff-serv)

Multi protocol Label Switching (MPLS)

Resource Reservation Protocol (RSVP)

Traffic conditioning (TC)

Type of Service (ToS)

Priority Queuing (PQ)

Weighted Fair Queuing (WFQ)

Weighted Random Early Detect (WRED)

Service Level Agreement (SLA)

Network Based Application Recognition (NBAR)

Per Hop Behavior (PHB)

Class of Service (CoS)

Chapter 1

Introduction

Quality of Service has become a vital need of any size of Corporate network in today's inter-networking world. QoS is defined as the proficiency of a network to offer quality service to dedicated network traffic over various technologies, like Frame Relay (FR), Asynchronous Transfer Mode (ATM), Ethernet and wireless (802.11) networks, SONET, and IP-routed networks. Quality-of-Service (QoS) can be achieved from any of those technologies which are mentioned above [1].The main purpose of Quality-of-Service (QoS) is to maintain allocated bandwidth, control the jitter, and manage the latency of the network. QoS also use to improve losses in the network. Quality-of-Service (QoS) has become one of the major factors in today's multimedia network due to which lots of research work is going on to make the network optimized as much as we can. Quality of Service (QoS) is used to classify and prioritize the delivery of different types of data packets (Voice, Video and data) based on different factors. It is important to make sure that providing priority for one or more flows does not mean that other packets will be failed. Quality-of-Service (QoS) was introduced to ameliorate the service provided by the traditional "Best Effort" data delivery [1]. As we know that Best Effort data delivery used to be acceptable when demand was not as high as it is today.

Quality of service enables us to establish an end-to-end traffic priority policy so that we can have prioritized data accordingly [2]. For example we can use Quality of service in managing the traffic from servers. We can control the traffic flow from dedicated VLANs. We can change the priority of traffic from different segments of our network depending on the requirements [1]. We can set priority policies in edge switches in our network to ensure the proper traffic handling according to our needs.

Chapter 2

Basic Quality of Service (QoS) Architecture

Quality of service (QoS) classifies packets on their type of service and defines priorities for each packet, it is achievable by different methods namely RSVP, RSVP-TE, MPLS, integrated services and differentiated services, among all of these we will achieve (QoS) by Differentiated Services which is one of the best practices that is mostly used by the most Internet service providers (ISP) to maintain end to end QoS. A brief comparison between the two most famous architectures integrated services and differentiated services is also provided in detail in later section that justifies the selection of Differentiated services over other architectures.There are three main enables for Quality of service (QoS) which is shown in Figure 2.1.

- QoS identification and marking
- Qos within a single network element or factors for assuring QoS (Queuing, Scheduling, and Traffic-shaping tools)
- QoS policy and management

2.1 QoS Identification and Marking

It refers to a method that deals with identification and marking of packets which enables QoS to coordinate from end to end between network elements. It is usually done through the classification of the packets. For example TCP and UDP packets should be identified

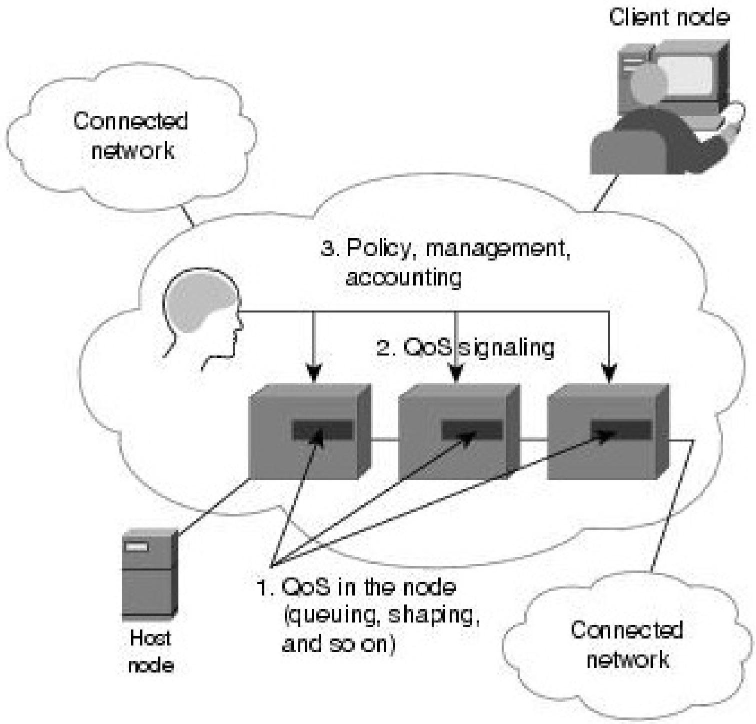

Figure 2.1: Basic QoS Architecture [5]

and marked then they should be treated according to their priority. For the preference of the particular data first of all the packet should be identified whether it is TCP or UDP. Second, the packet may or may not be marked. These two steps lead to classification. However if the packet is identified but not marked, classification is said to be on a per-hop basis [5] [13]. This is when the classification concerns only to the device that it is on, not passed to the next hop (router). This happens with priority queuing (PQ) and custom queuing (CQ). When packets are marked for network-wide use, IP precedence bits can be set as done in Differ-services. The methods that are used to identify flows include access control lists (ACLs), policy-based routing, committed access rate (CAR), and network-based application recognition (NBAR) [4] [5].

2.2 Factors for assuring QoS

QoS within a single network element depends on multiple factors i-e. congestion management, queuing management, link efficiency, traffic shaping and policing.

2.2.1 Congestion Management

During the transmission of different types of data packets of TCP and UDP usually the amount of traffic is increased than the capacity of the link. At this time router needs to decide that whether it should follow the FIFO process (first in first out) or adopt an alternative way. In this case congestion management acknowledges this issue by using the different tools like priority queuing (PQ), custom queuing (CQ), weighted fair queuing (WFQ), and class-based weighted fair queuing (CBWFQ).

2.2.2 Queue management

Traffic should be queued and scheduled according to the type of traffic this will help the router buffer to prioritize the delay sensitive data for efficient transmission on the network. Sometimes different buffers with appropriate routing mechanisms will need to be included in routers. Each buffer will pertain to a different class of service, and will forward packets through its queue according to the assigned queuing mechanism. For example Weighted random early detect (WRED).

2.2.3 Link Efficiency

Link Efficiency refers to the ratio of actual output received at the user end to the expected output considering the multiple loss factors. Link efficiency plays a vital role for maintaining QoS across the network. Link efficiency has a direct impact on serialization delay due to which QoS can be compromised. In other words link efficiency plays a vital role in order to minimize end to end packet delay [5].

2.3 QoS Management

QoS management plays a an important role in defining the QoS policies and goals. QoS management also ensures the monitoring of QoS across the network which helps in understanding the traffic flow. Following steps lead to QoS management in a network.

1. Baseline the network with devices such as RMON probes. This helps in determining the traffic characteristics of the network. Also, applications targeted for QoS should be base-lined (usually in terms of response time) [5] [13].

2. Deploy QoS techniques when the traffic characteristics have been obtained and an application has been targeted for increased QoS. Evaluate the results by testing the response of the targeted applications to see whether the QoS goals have been reached [5] [13].

2.4 Need of Quality of service in Networks

The traditional way for flowing of Internet traffic is "Best-effort" mechanism which does not assure to provide any guarantees that data is delivered properly or that a user would have a certain priority in data transmission or guaranteed quality level of service. In a best-effort network all users share the best-effort service which means that the date rate of each user will be different depending upon the current traffic load. If an important data needs to be pass on so there was no mechanism to prioritize that data packets than other packets during transmission. However, when the network congestion occurs, it is important to manage the traffic flow according to the priorities. So how to manage those priorities? The answer of this question is Quality of service .Without the adaption of Quality of Service (QoS) prioritization, the data which is least important will consume the bandwidth and halt the delivery of more important traffic [1] [2]. That is, without QoS, major part of the traffic received by the edge devices will forwarded with the same priority as it was upon entering at the device. So in this way we will not be able to have the most important data when it will be needed.

2.5 Advantages of Quality of Service (QoS)

Quality of Service (QoS) allows us to control the efficiency of complex networks, network applications and traffic types. QoS can be implemented to any size of network (small , medium and large corporates) for achieving efficiency within the available network resources [1] [3]

2.5.1 Control over resources

Using QoS one can have control over allocated resources like bandwidth, rate control, queuing and scheduling, congestion management, admission control and routing control traffic protection which is being used in the network.

2.5.2 Tailored services

Quality of Service (QoS) allows ISPs to distinguish the grades of services that are offered to their customers.

2.5.3 Foundation for a fully integrated network

Fully integrated network refers to a centrally managed network with all the required services with an addition of redundancy in case of devices failures. Implementing QoS technologies in your network will allow you to have the fully integrated multimedia network operational in future.

2.5.4 Traffic Shaping and Policing

Traffic shaping refers to the flow of traffic with the proper management of bandwidth utilization. Traffic Shaping decides the distribution of bandwidth among the shared resources (Virtual servers hosting multiple services). Policing is similar to shaping, but it differs in one very important way: Traffic that exceeds the configured rate is not buffered (and normally is discarded).

2.5.5 Service Level Agreement (SLA)

Service Level Agreement explicitly defines the assurance of Quality of Services in the network. SLA is a mutual consensus between users and suppliers that ensures the availability of services for example Internet connectivity between two different branches is managed by ISP and End-End QoS should me maintained during routing of packets across multiple hops.

Chapter 3

Identification of Problems and Requirements for Redesigning a Medium Scale Network

This chapter addresses the major problems of a medium scale network and how those problems can be removed by redesigning the same network within the limited resources. Small organizations are reluctant in introducing the QoS within their existing network because of the cost and complexity involved in order to accomplish those changes. The medium scale organization will highly prefer to make the changes if the changes are cost effective and easier to implement. For highlighting those problems Figure 3.1 shows a medium scale network which has set of core router and edge routers which are responsible for routing the traffic from the access layer to the distribution layer. Users are connected to the layer 2 switches and LAN server is connected to router R1 which manages the LAN network and file servers for the repositories. The network shown in Figure 3.1 is under utilized and has the capacity for introducing QoS infrastructure within the network. It has been seen often that medium scale companies usually use a separate telephonic network (ISDN) for making calls between different offices and sometimes they do not have any dedicated video conferencing facility in place due to the lack of budget. They usually utilize their Internet services just for sending the data packets across the network however they are really keen to utilize the resources efficiently. This project work provides a detail idea that how a medium scale network can host the multimedia services within the existing infrastructure without spending thousand of dollars. This chapter also covers some of the strict requirements from a medium scale organization and what performance factors are ideal for implementing voice and video services within the network. For this purpose, a sample network is designed using OPNET simulator to analyze performance of multimedia services. Multiple scenarios for the voice codecs and routing protocols are compared for analyzing the performance factors (Jitter , End to End delay and packet delay variation) which provided a great help in implementing multimedia services in the actual medium scale network [3] [4].

3.1 Problem identified in the medium scale network

Large scale organizations have already followed the trend of implementing Quality of service in their network but medium scale organizations still have some constraints and some common issues due to which they are usually reluctant for making any changes into the network. Some common problems for the medium scale network are discussed below.

1. The network diagram shows in Figure 3.1 shows a medium scale network which uses an MPLS network for the transmission of data packets within the local office and a branch office. Following are the problems exhibit by a medium scale network.

2. The company uses a separate telephony network for communication (telephony calls) in each office. The cost of maintenance of this telephony network is high as well is an additional burden on the company for a different network and a cost effective solution is needed for minimizing the cost.

3. The company also does not have a facility to accomplish the video conference among different branches. It does have temporary video conference through skype for business.

4. The company is relying on Best effort services if they temporarily use any voice or video conference solution for the communication across the enterprise and branch office. Every data packet is treated in a similar way

5. The company can not utilize the available bandwidth efficiently as it is only being used for the data packets

6. The company does not posses any central architecture for the management of Internet traffic

7. The company does not have any traffic policing as there is no central architecture to prioritize the network traffic
8. The company's communication is vulnerable to threats and attacks as there is no proper network security in place for securing the configuration

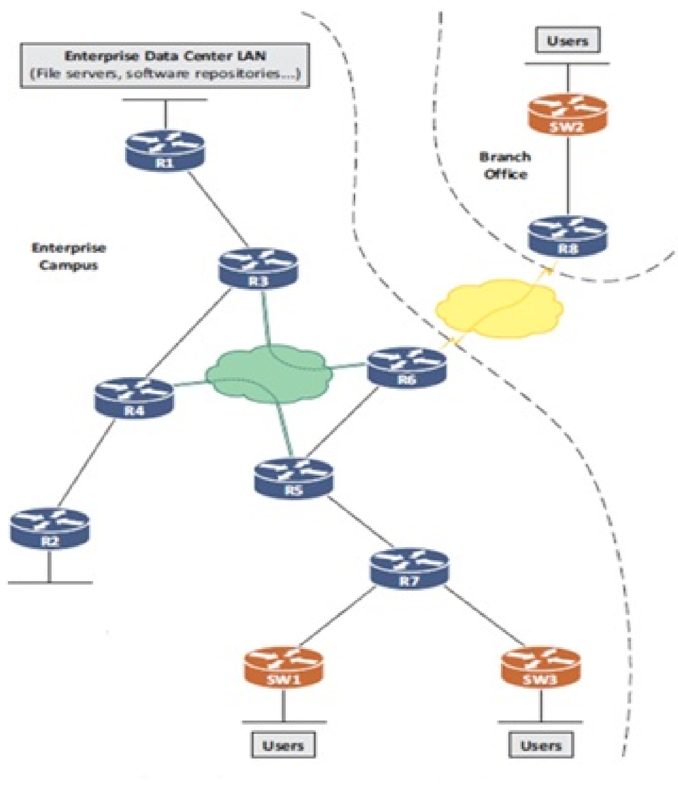

Figure 3.1: Medium Scale Network [2]

3.2 Requirements for Redesigning the Network

It can be easily perceived from the above discussion that the medium scale network shown in Figure 3.1 needs to be redesigned in order to achieve QoS across the network but keeping a fact in mind that every organization has particular requirements followed by the constraints when a change needs to occur within the infrastructure. Our medium scale network also has few requirements along with some constraints which should be considered during the redesigning of the the network. Following are the requirements that needs to be considered while the network is redesigned.

1. The company wants to fully utilize the existing network resources (routers, switches and servers) for introducing multimedia applications within the network
2. The company has a limited budget for introducing VoIP and video conferencing solution.
3. The company desires to have flawless voice and video communication between enterprise and branch office.
4. The company wants to utilize the available bandwidth efficiently.
5. The company wants to have central network management for monitoring and managing the network traffic
6. The company wants the route redundancy and load balancing for network traffic using existing network architecture (Routers)
7. Any solution proposed should be tested, verified and documented in order to avoid any sort of disruptions
8. The company wants to secure the communication across the network.

For best performance of multimedia applications every application should be implemented with the most suitable parameters. The parameters should not be chosen either by hit and trial or by referring to the research articles as most of them are written for specific scenarios . The best way to choose the parameters for a particular application is to perform a brief analysis of those particular parameters under strict requirements. For example if VoIP services will be introduced in the network the most important thing will be the choice of voice codec and the how that voice codec will utilize the available bandwidth in an efficient manner [4]. This approach is achieved by simulating a real time environment in OPNET and the performance metrics like jitter, queuing delay and end to end packet delay are observed for different number of VoIP calls between the enterprise and the branch office of the company. This real time analysis of simultaneous calls provided a significant idea that which voice codec will be the best fit for the redesigned network in order to achieve QoS. The company wishes to achieve following improvements in the network.

Chapter 4

Simulation Scenario and Results for the selection of parameters

In this chapter various simulation results are obtained in order to select the best performance factors for the redesigning the network (voice codec, routing protocols, video codec and methodology for QoS). A sample network shown in Figure 4.1 is designed using OPNET modeler 17.5 similar to the medium scale network Figure 3.1 with an addition of VoIP and video services into the network to analyze the behavior of voice codecs and routing protocols against different set of VoIP calls. Theses services (VoIP and Video) are implemented with the different voice codecs and the performance of these codecs is summarized by considering different factors (jitter, Packet Delay variation and Packet End-End Delay) in the real time environment.

4.1 Simulation Scenario and Methodology

The simulation scenario is designed through number of network devices that includes routers, switches and servers within Diff-serv domain. Figure 4.1 shows the overall configuration of network which includes four edge routers, four core routers , VoIP phones , Video conference bridges and Enterprise data center which has the facility of web hosting, file server and software repositories. A new facility of Enterprise media servers is introduced into the existing network which now hosts VoIP, Video conference and Data services altogether for the end users. A 1000-X Base Local Area Network (LAN) of 500 users is connected to edge Router 7 for accessing the multimedia services within the enterprise. Similarly we have a 1000-X Base Local Area Network of 100 users connected to Branch Edge router 4 which then connects to the enterprise core router 6 through an MPLS connection. The core routers are connected with each other through the backbone MPLS connection and they are provisioned for Weighted Fair Queuing (WFQ) queuing profile in which weights are assigned to individual queues on the basis of priority of services. The core routers are responsible for assuring the QoS for the different type of packets (VoIP, Video and Data) at inbound/outbound interfaces that is received through the edge routers. Core routers treat the packets on the basis of Per hop Behavior (PHB) and finally forward the packets as per the applied scheduling profile to the final destination. The edge routers are responsible for the DSCP based marking of the packets (VoIP, Video and Data) originated by the end users.

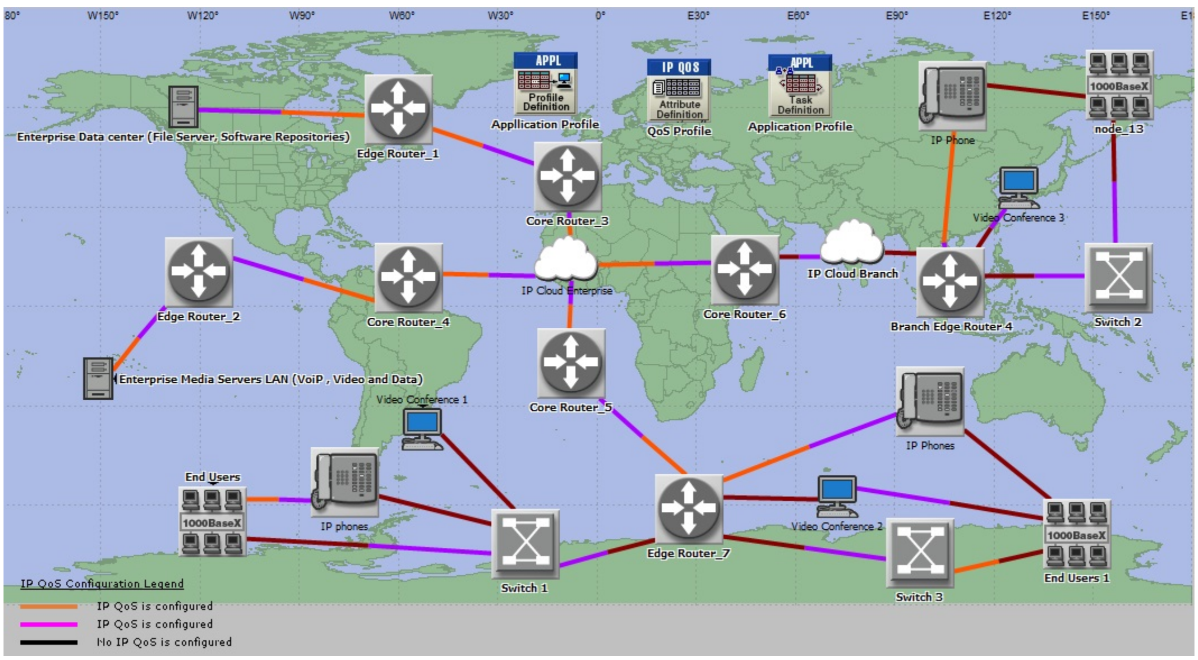

Figure 4.1: Sample Network Design for Simulations designed using OPNET

4.2 Methodology

Using the simulation environment shown in Figure 4.1 burst of multimedia traffic (VoIP, video and data) is initiated from the enterprise office to the branch office or vice versa to observe the behavior of network in terms of jitter , latency , Packet delay variation and End to End delay packet delay which eventually will help during the redesigning of the medium scale network Figure 3.1. Each router is configured to route the traffic under

Table 4.1: Assigned DSCP values to the Applications

Application	DSCP Name	DSCP Value
Voice over IP (VoIP)	EF	46
Video Conferencing	AF41	34
Database	AF21	18
Web	DF(CS0)	0

Weighted Fair Queuing (WFQ) and packets are forwarded to the destination hops on Per Hop Behavior (PHB). The edge routers are configured to mark to packets and forward those packets according to the DSCP marking shown in Table 4.1 to the core routers. Real time traffic (VoIP, video and data) is generated and multiple scenarios are tested for VoIP codecs, video codecs and routing protocols in order to select the appropriate parameters for maintaining QoS for the redesigned network. As voice and video are delay sensitive applications so they are given preferences to get routed during DSCP marking. The similar scenario will be applied to the actual network in later sections. Due to the MPLS connection the traffic routing is extremely fast than the traditional peer configuration. From Table 4.1 It can clearly be seen that highest priority is assigned to voice packets as it is configured with Expedited Forwarding (EF) for per hop. Video conferencing is configured with Assured forwarding AF41 and Database traffic is assigned AF21 while Web traffic is configured with Default forwarding (DF) . The network devices in sample network are configured with the Diff-serv QoS profile which provided a great help in redesigning the medium scale network. Simulations are fairly easy than the real networks because in real networks each node (routers) has to be configured manually by the network engineers and includes a fair amount of cost. This project work also helps in reducing the configuration cost because the actual network configuration is explained in later usually costs a fair amount of dollars.

4.3 Selection of voice codec for redesigned network

Voice is a delay sensitive application which demands high priority within the network. For maintaining the end to end QoS selection of appropriate voice codec is really important. This section discusses the number of performance factors like Jitter , End-End packet delay and packet delay variation in detail for different voice codecs i-e G.711 (64 Kbps), G.729 (8 Kbps) and G.723.1 (6.3 Kbps) within OPNET simulator. The simulation results provides us a broad idea that which is the most suitable voice codec for network hosting VoIP traffic [4]. In the simulation scenario simultaneous calls between enterprise and branch office are made for a certain period of time.

4.3.1 Jitter

Jitter is defined as a variation in the delay of received packets [9]. The simulation results shown in this section are obtained for jitter experienced by VoIP traffic (250, 750 and 1500 calls). VoIP traffic is originated from enterprise and branch office. The simulations are carried out for multiple voice codecs i-e G.711 (64 Kbps), G.729 (8 Kbps) and G.723.1 (6.3 Kbps) under Weighted Fair Queuing (WFQ) discipline. Each simulation is obtained for different set of VoIP calls across the network. A brief comparison of voice codecs is also provided in this section in order to summarize the performance.

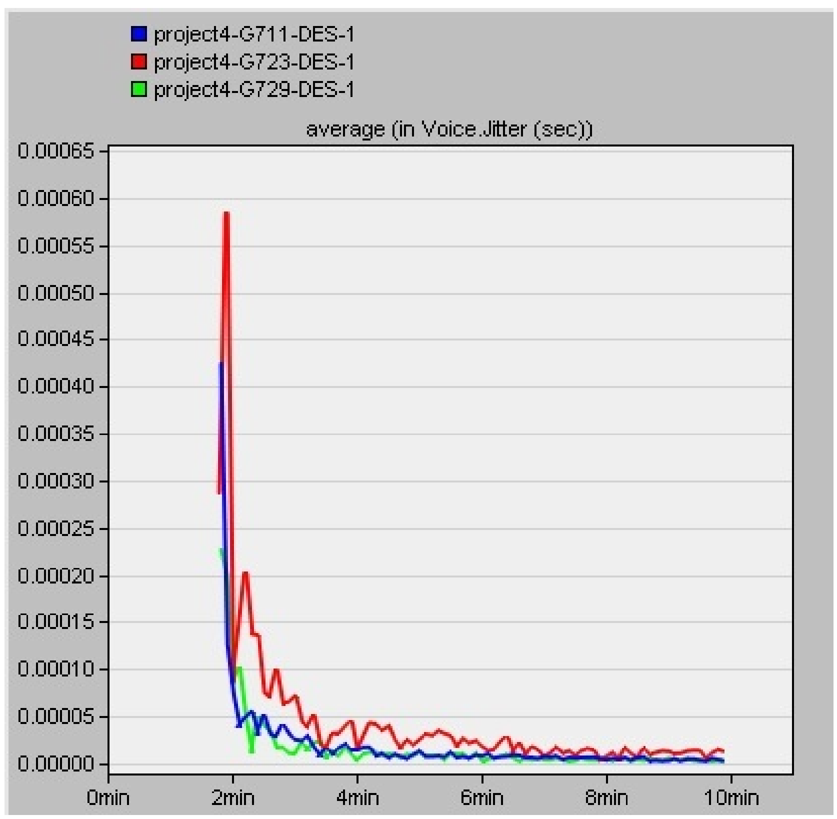

Figure 4.2: Jitter for = 250 VoIP calls

The simulation result in Figure 4.2 is obtained for average jitter (sec) vs simulation time (sec) for a set of 250 VoIP calls across the network. Figure 4.2 shows that the jitter value reaches to a maximum level because all the calls are considered to be connected at first due to which queue capacity also reaches to a threshold value and then after sometime the value of jitter started to decrease as the number of calls decreased. Figure 4.2 illustrates that G.729 exhibits the lowest jitter value however G.723 shows the highest jitter value for a given set of 250 VoIP calls between enterprise and branch office. The jitter experienced by the voice codecs is significantly high in the beginning of the simulation because all 250 sessions established in the beginning of the simulation which causes the buffer to fill immediately and then it started to decrease gradually with the end of the sessions. G.711 exhibits slightly higher jitter than G.729 in the beginning of the simulation but managed to achieve the same behavior similar as of G.729 with the time interval.

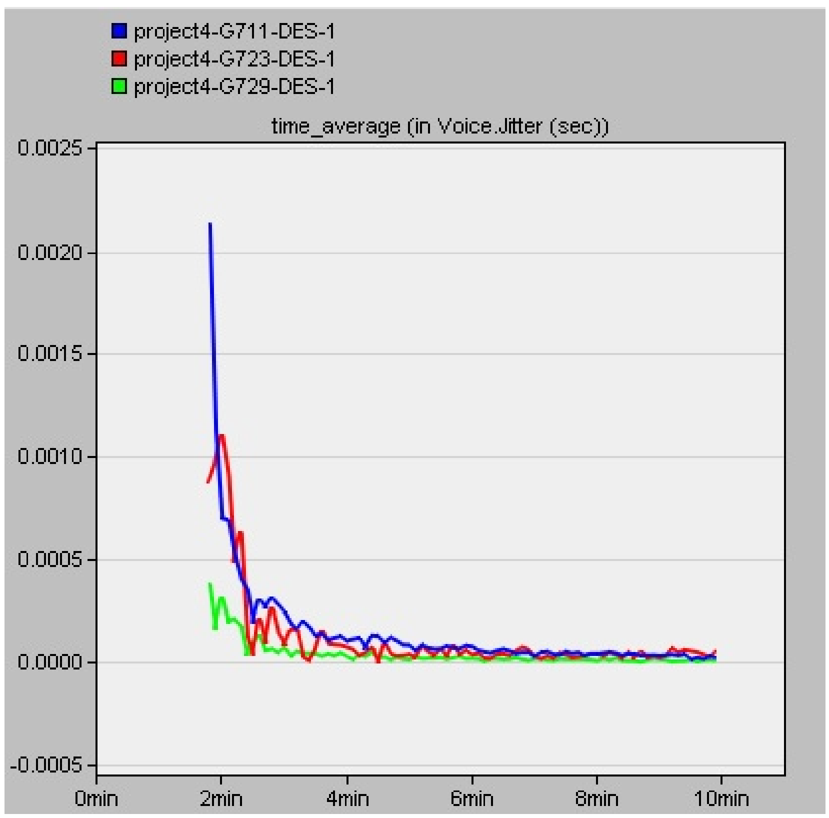

Figure 4.3: Jitter for = 750 VoIP calls

Figure 4.3 shows the result obtained for average jitter (sec) vs simulation time (sec) for a set of 750 VoIP calls across the network. Figure 4.3 shows that the jitter value reaches to a maximum level because all the calls are considered to be connected at first due to which queue capacity also reaches to a threshold value and then after sometime the value of jitter started to decrease as the number of calls decreased. Simulation result in Figure 4.3 illustrates that jitter experienced by the voice codecs is increased with the increase number of VoIP calls. G.729 still exhibits the lowest value of jitter however with an increase number of VoIP G.711 started to experience more jitter than G.723 under given set of VoIP calls. The jitter value for G.723 has slightly got better for higher number of VoIP calls.

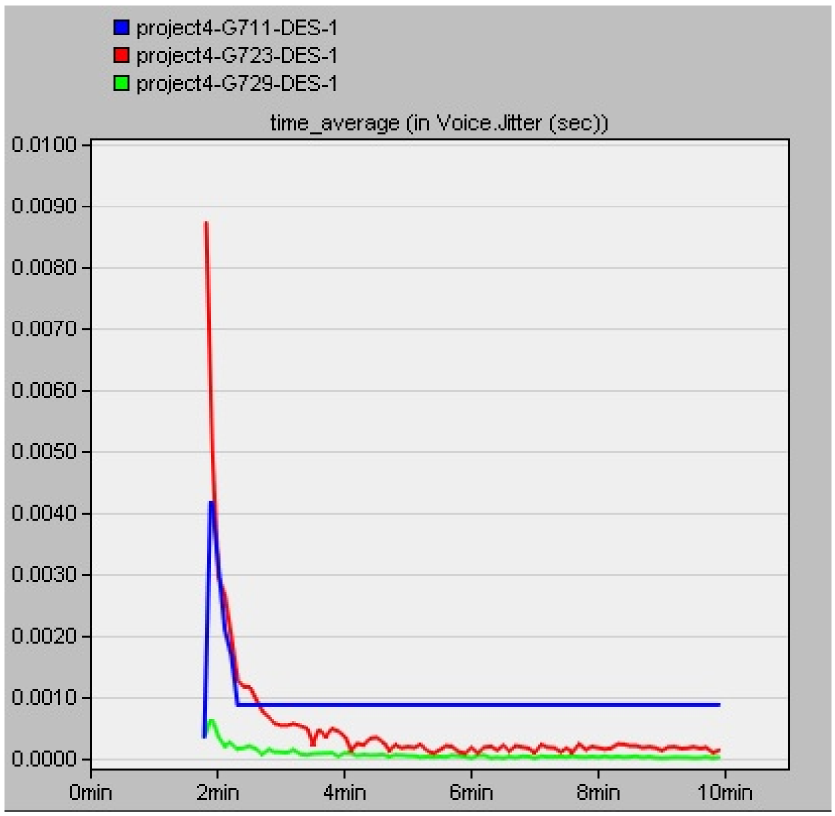

Figure 4.4: Jitter for = 1500 VoIP calls

The simulation result in Figure 4.4 is obtained for average jitter (sec) vs simulation time (sec) for a set of 1500 VoIP calls across the network. Figure 4.4 shows that the jitter value reaches to a maximum level because all the calls are considered to be connected at first due to which queue capacity also reaches to a threshold value and then after sometime the value of jitter started to decrease as the number of calls decreased. Figure 4.4 illustrates that G.729 exhibits the lowest jitter value however G.711 shows the highest value of jitter under the given configuration. G.723 shows more improved jitter value than previous simulations with an increase number of VoIP calls but the best results are shown by G.729 for all three simulations.

From the above simulation results of Figure 4.2, Figure 4.3 and Figure 4.4 it is observed that G.711 possess experienced higher jitter than other two voice codecs when the number of VoIP calls gets increased. Initially G.711 showed good results for the less number of VoIP calls but it increases significantly with the increase number of VoIP calls. Furthermore, G.729 experienced the lowest jitter value for all the given configurations. It can clearly be seen from these simulation results that how can we minimize the jitter in a network by selecting an appropriate voice codec which eventually will help in maintaining the end to end QoS across the network. The efficiency of VoIP traffic is also dependent on the size of datagram packets. Similar calculations are performed in the latter sections with the same scenario of the VoIP calls to observe the other metrics like packet delay variation and end to end packet delay.

4.3.2 End to End packet delay

End to End delay refers to the time taken by the packet to travel from source to destination. The simulation results in this section are obtained for the average end to end packet delay vs simulation time (sec) under same set of simultaneous calls (250, 750 and 1500) across the network. End to End packet delay is observed for different voice codecs under WFQ queuing discipline configured. This section also covers a detail comparison of end to end packet delay for different voice codecs i-e G.711 (64 Kbps), G.729 (8 Kbps) and G.723.1 (6.3 Kbps) for the given set of VoIP calls.

The simulation result in Figure 4.5 is obtained for the average End to End packet delay (sec) vs simulation time for a set of simultaneous 250 VoIP calls. Figure 4.5 illustrates that G.711 exhibits the lowest end to end packet delay while G.723.1 experiences highest end to end packet delay for the given set of VoIP calls. End to End delay increased slightly in the beginning and reached to a constant value with the time interval

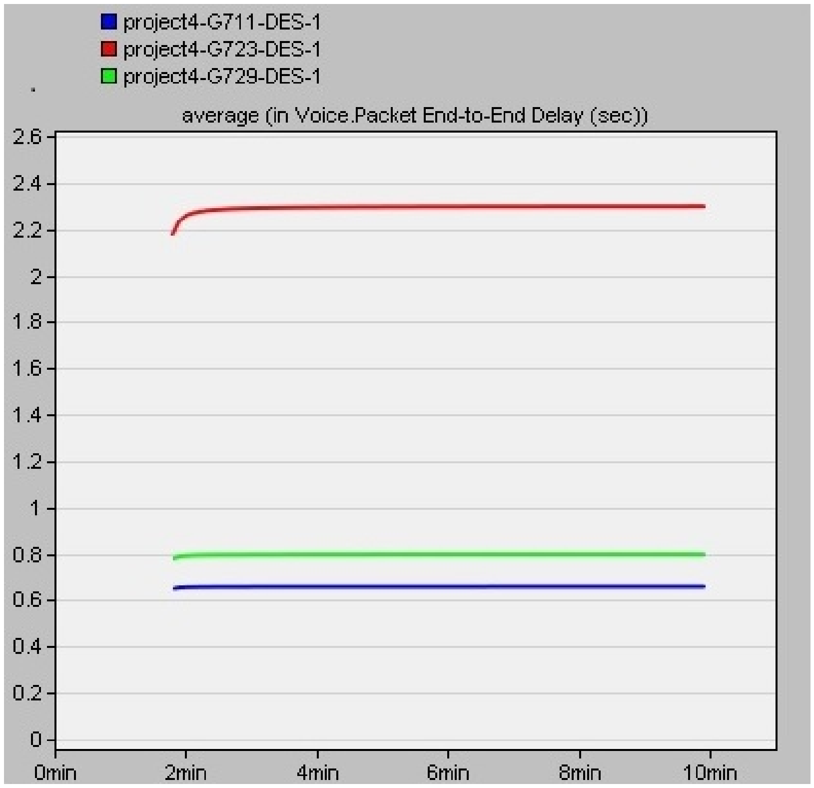

Figure 4.5: End-End packet delay for = 250 VoIP calls

for each voice codec.

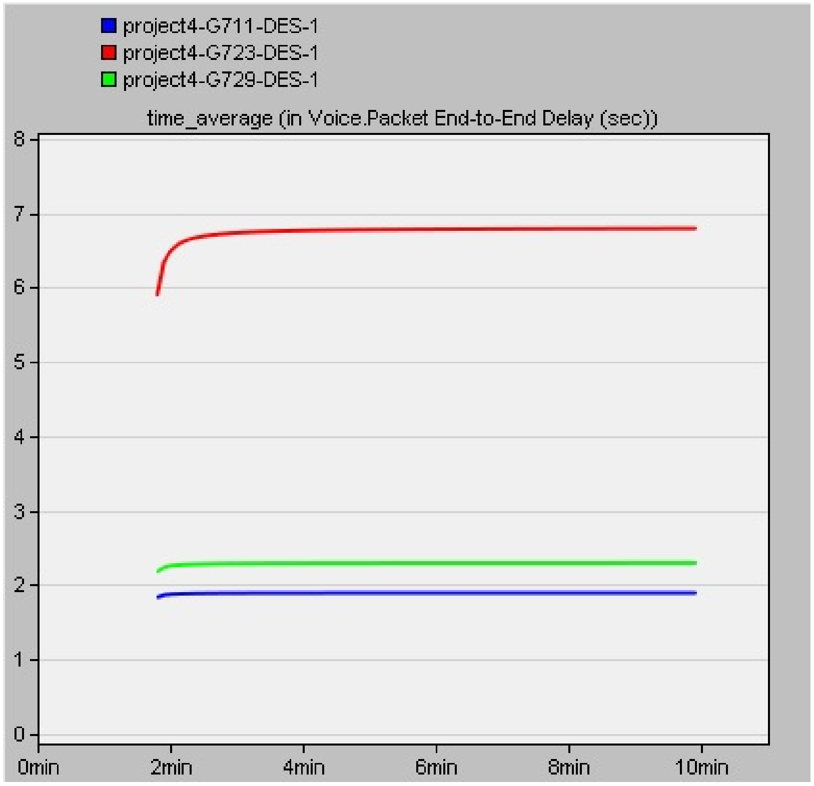

Figure 4.6: End-End packet delay for = 750 VoIP calls

The simulation result in Figure 4.6 is obtained for the average End to End packet delay (sec) vs simulation time for a set of simultaneous 750 VoIP calls. Figure 4.5 illustrates that G.711 still exhibits the lowest end to end packet delay for the given set of VoIP calls while G.723.1 shows the highest end to end packet delay and shows a continuous increase in End to End delay with the increase number of VoIP calls across the network. It can also be observed that End to End delay for G.729 started to decrease with the increase number of VoIP calls.

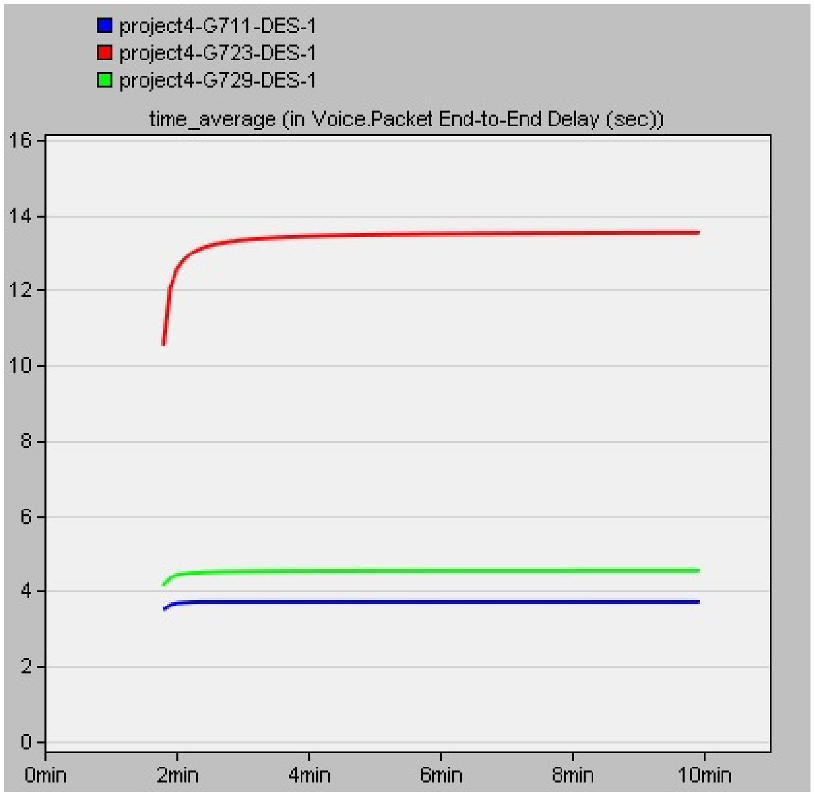

Figure 4.7: End-End packet delay for = 1500 VoIP calls

The simulation result in Figure 4.7 is obtained for the average End to End packet delay (sec) vs simulation time for a set of simultaneous 1500 VoIP calls. Figure 4.7 illustrates that G.711 still exhibits the lowest end to end packet delay for the given set of VoIP calls while G.723.1 shows significant increase in end to end packet delay with the increase number of VoIP calls across the network. End to End delay for G.729 showed significant improvement and will achieve similar results as that of G.711 with the increase number of VoIP calls.

From the simulation results shown in Figure 4.5, Figure 4.6 and Figure 4.7 it can be concluded that the end to end packet delay increases significantly for G.723.1 with the increase of VoIP traffic. In above simulation results G.711 possess the lowest end to end packet delay with the increase number of calls. However, G.729 showed continuous improvement in end to end packet delay than rest of the voice codecs for the increased number of VoIP calls. End-End packet delay is another very important factor in order to achieve QoS into the Network and the simulation results shows that G.729 will be more appropriate voice codec for the redesigning the medium scale network as it will show better results with the large number of VoIP calls.

4.3.3 Voice Packet Delay Variation

The simulation results in this section are obtained for the average packet delay variation vs simulation time (sec) under same set of simultaneous calls (250, 750 and 1500) across the network. Packet delay variation is observed for different voice codecs under WFQ queuing discipline configured. This section also covers a detail comparison of packet delay variation for different voice codecs i-e G.711 (64 Kbps), G.729 (8 Kbps) and G.723.1 (6.3 Kbps) for the given set of VoIP calls.

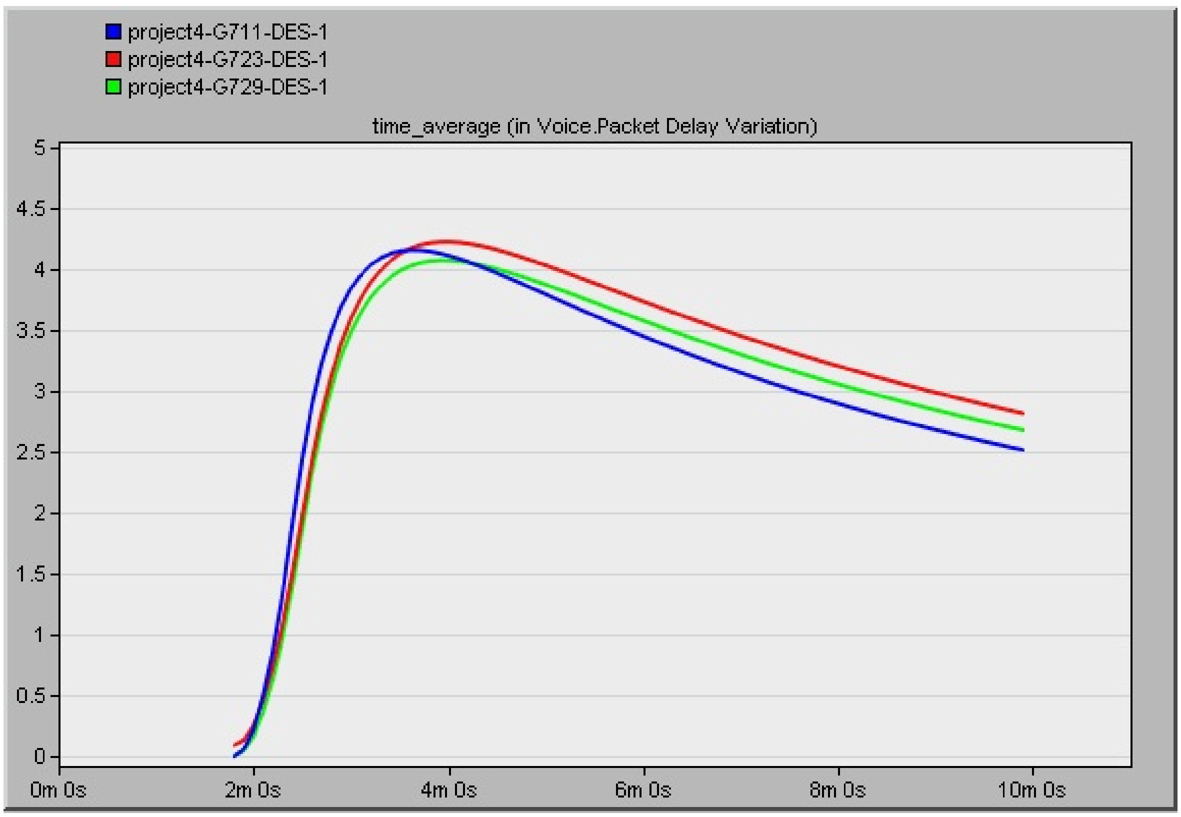

Figure 4.8: Voice Packet Delay Variation for = 250 VoIP calls

The simulation result in Figure 4.8 is obtained for the average packet delay variation vs simulation time for a set of simultaneous 250 VoIP calls. Figure 4.8 illustrates that G.711 still exhibits the lowest packet delay variation for the given configuration while G.723.1 shows the highest packet delay variation. G.729 possessed started with the lowest packet delay variation in the beginning but it increased with the time interval.

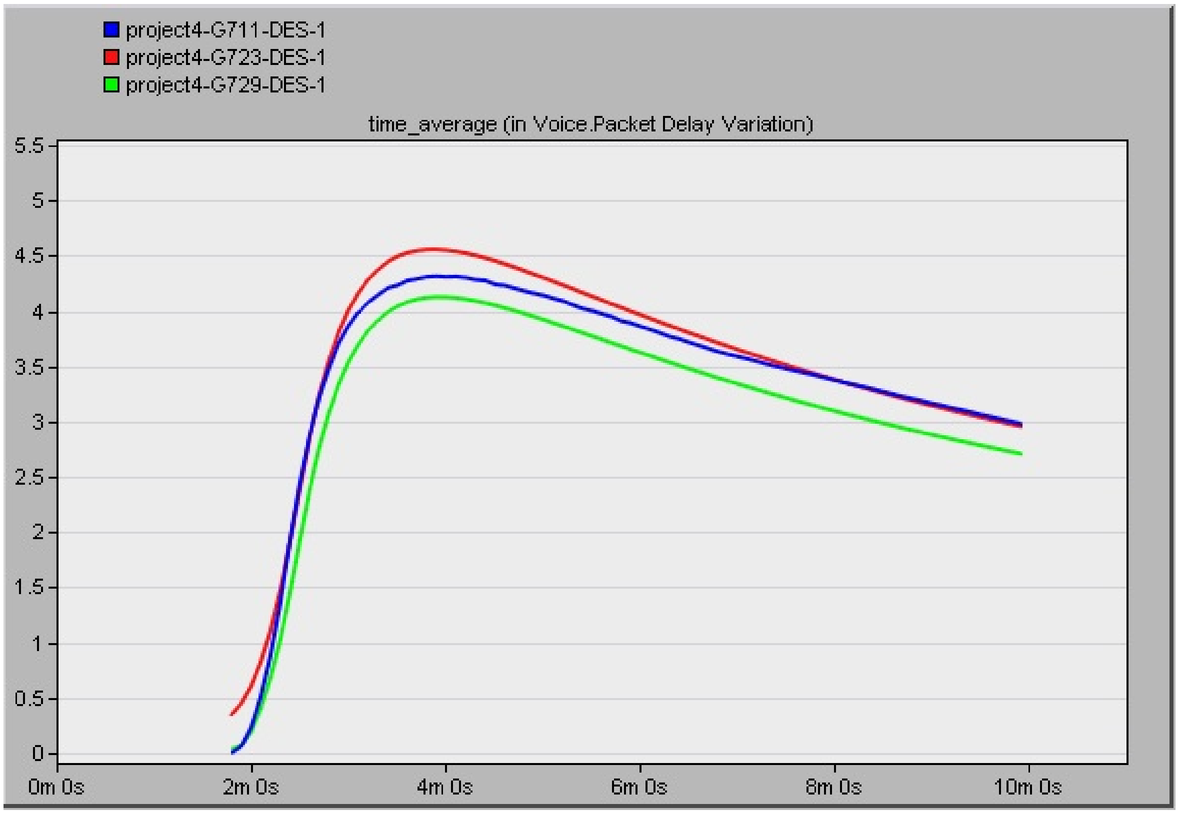

Figure 4.9: Voice Packet Delay Variation for = 750 VoIP calls

The simulation result in Figure 4.9 is obtained for the average packet delay variation vs simulation time for a set of simultaneous 750 VoIP calls. Figure 4.9 illustrates that G.729 exhibits the lowest packet delay variation with the increased number of VoIP calls for the given set of VoIP calls while G.723.1 shows significant improvement in packet delay variation with the increase number of VoIP calls as compare to G.711. Packet delay variation also increases with the increase in VoIP traffic and is dependent on the number of packets entering into the network.

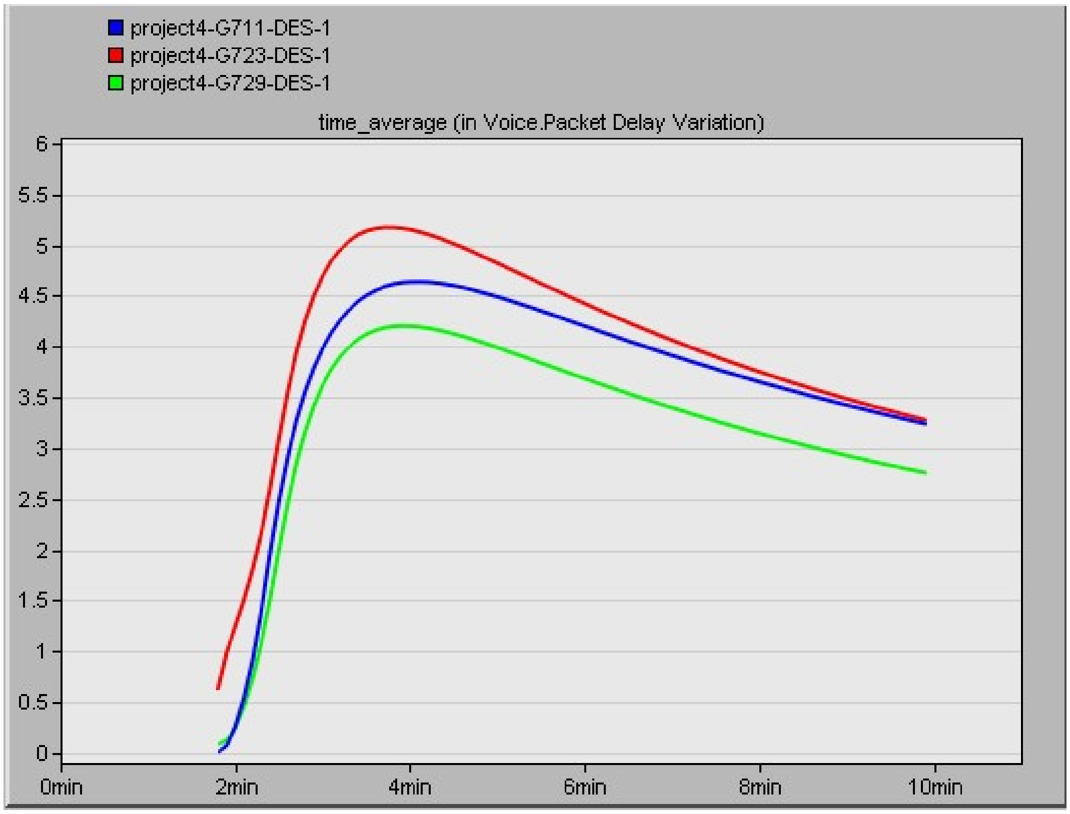

Figure 4.10: Voice Packet Delay Variation for = 1500 VoIP calls

The simulation result in Figure 4.10 is obtained for the average packet delay variation vs simulation time for a set of simultaneous 1500 VoIP calls. Figure 4.9 illustrates that G.729 exhibits the lowest packet delay variation for the given configuration and the value is continuously improving with the increase number of VoIP calls. G.723.1 started to experience high packet delay variation with the increase number of VoIP calls as compare to G.711.

From above simulation results of Figure 4.8, Figure 4.9 and Figure 4.10 It can be concluded that G.723 exhibits the highest voice packet delay variation for the increased VoIP traffic however G.729 experienced lowest packet delay variation for all three configurations. G.729 has shown best results for packet delay variation for all three configurations.

From the overall comparison of all the simulation results lead us to single point for selecting an appropriate voice codec which we will use during the redesigning of the medium scale network. It can be clearly seen that G.729 will be the most suitable voice codec for our r network. G.729 has shown significant and consistent results for jitter, end to end delay and packet delay variation than other voice codecst [6].

4.4 Selection of Routing Protocol

This sections highlights the need of an appropriate routing protocol in order to achieve end to end quality of service across the network after selecting a suitable voice codec for the redesigning the network shown in Figure 3.1. The selection of a suitable routing protocol is another most important factor as the routing of the packets can effect the QoS across the network. Currently the routing protocol which is configured within the existing medium scale network is EIGRP because the company's core architecture is comprised of Cisco devices and EIGRP has the fastest convergence among all the routing protocols but after simulating the VoIP and video traffic scenarios OSPF showed better performance results for Jitter and Traffic delay variation than EIGRP which can be clearly seen in the simulation results. For observing the performance of routing protocols different sets of VoIP calls are generated across the network. The set of VoIP calls (500. 2500 and 4000) are used for performance observation. The simulation time is set to 4000 sec for observing the definite results as simulation was found to be stuck for a shorter period of time

4.4.1 Jitter for OSPF and EIGRP

This section covers a detail comparison of simulation results obtained for the jitter (sec) vs the time at which the jitter of VoIP traffic for both OSPF and EIGRP started to increase under same set of VoIP calls/hr (500, 2500 and 4000) [7]. In addition to that minimum and maximum values of jitter are also calculated for each set of VoIP traffic in order to show the variation in Jitter.

The graph in 4.11 shows that EIGRP started to experience some jitter after 450 sec and it increased with the passage of time for 500 calls/hr however on the other hand the jitter for OSPF was negligible. The minimum and maximum jitter values for EIGRP were noted as 18.7 μs and 11 ms whereas minimum and maximum jitter values for OSPF were 79.21 ns and 0.96 ps.

The simulation result in Figure 4.12 illustrates that the jitter value for 2500 VoIP calls/ hr started to increase in the same way for EIGRP as of previous simulation while the jitter value for OSPF shows slight increase in between 480 sec to 560 sec and then it gets stable till the simulation time reaches to 3500 sec. The minimum and maximum jitter values for EIGRP were noted as 11.4 μs and 12 ms whereas minimum and maximum jitter values for OSPF were 7.23 μs and 0.12 ms.

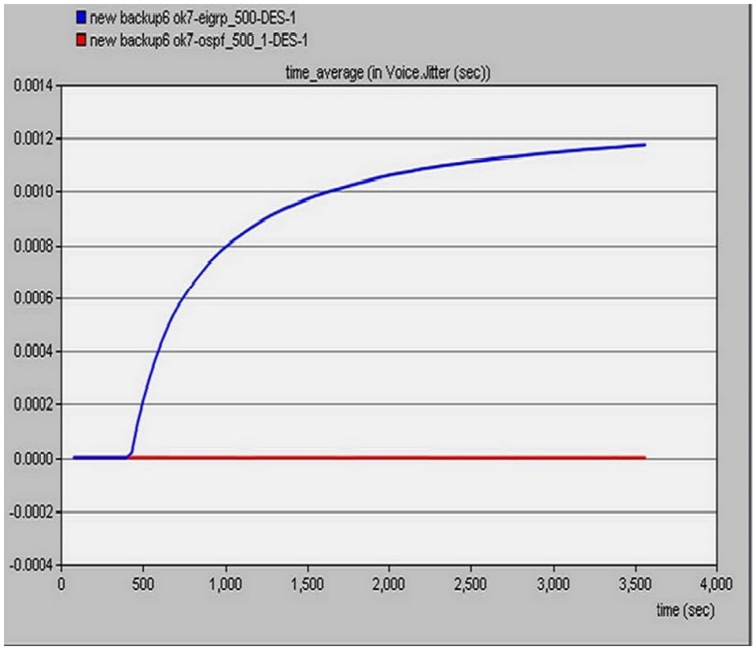

Figure 4.11: Jitter (sec) for = 500 VoIP calls

Table 4.2: Average Jitter value of OSPF and EIGRP

VoIP calls/hr	**OSPF Jitter**	**EIGRP Jitter**
500	0.39μs	0.59ms
2500	0.63μs	0.99ms
4000	0.20μs	0.64ms

The simulation result in Figure 4.13 illustrates that the jitter value for 4000 VoIP calls/ hr for EIGRP started to increase at 380 sec and becomes constant at 3650 sec while the jitter value for OSPF slightly increased in between 460 sec to 500 sec and then it gets stable till it reaches 3500 sec. The minimum and maximum jitter values for EIGRP were noted as 8.16 μs and 0.64 ms whereas minimum and maximum jitter values for OSPF were 2.32 μs and 1.92 μs.

Table.I shows the average Jitter value for both OSPF and EIGRP for different number of VoIP calls. Table.I illustrates that OSPF experiences less jitter than EIGRP for the increase number of VoIP calls across the network. .

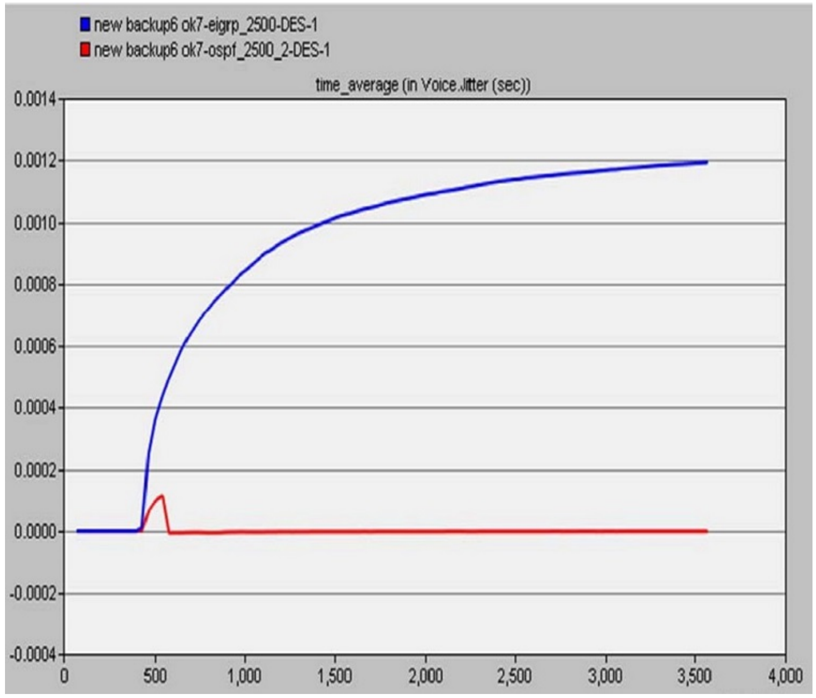

Figure 4.12: Jitter (sec) for = 2500 VoIP calls

4.4.2 Packet Delay Variation

The simulation results in this section shows a explicit comparison of Packet Delay Variation for both OSPF and EIGRP. The results are obtained for actual delay vs the actual time when the delay is started to increase for both routing protocols. It can be clearly seen from the simulation results that Packet Delay Variation posses by OSPF is much lesser than that of EIGRP.

The simulation result in Figure 4.14 shows that EIGRP started to experience some Packet Delay Variation after 860 sec for 500 calls/he and it increased exponentially with the passage of time however on the other hand OSPF has shown a lesser value of Packet Delay Variation. The minimum and maximum Packet Delay Variation values for EIGRP were noted as 1.39 ms and 480 s whereas minimum and maximum Packet Delay Variation values for OSPF were 11.9 ns and 192 μs.

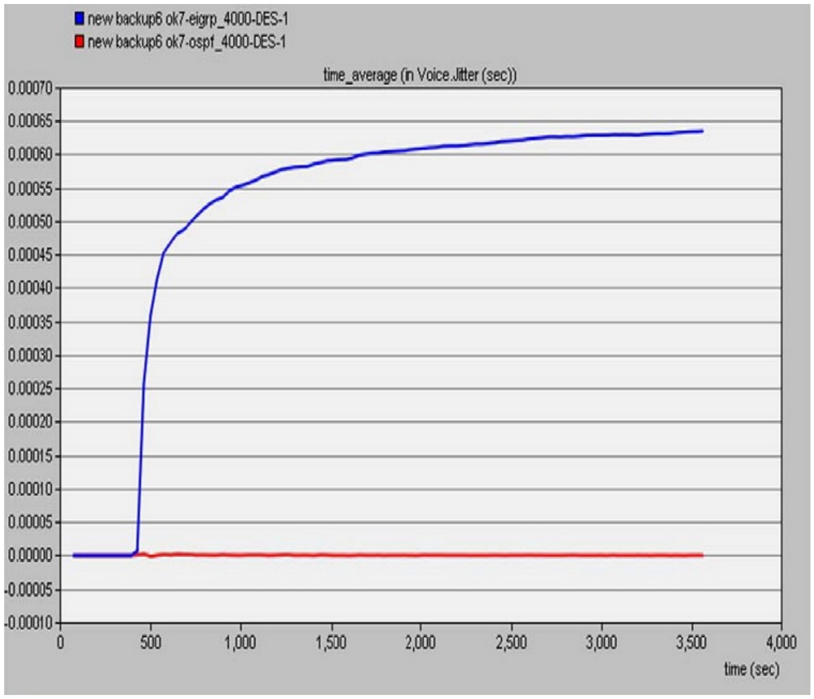

Figure 4.13: Jitter (sec) for = 4000 VoIP calls

The graph in Figure 4.15 illustrates that EIGRP experienced some Packet Delay Variation after 1000s for 2500 VoIP calls/ hr and gradually increased in the same way for EIGRP as of previous simulation while OSPF packet traffic delay is almost negligible. The minimum and maximum Packet Delay Variation values for EIGRP were noted as 0.52 μs and 42.5 s whereas minimum and maximum packet delay variation values for OSPF were 3.92ns and 0.48ms.

The graph in Figure 4.16 illustrates that EIGRP experienced some Packet Delay Variation after 980s for 4000 VoIP calls/ hr and gradually increased in the same way for EIGRP as of previous simulation while OSPF packet traffic delay variation is not noticeable. The minimum and maximum Packet Delay Variation values for EIGRP were noted as 2.2 ps and 3.78 s whereas minimum and maximum packet delay variation values for OSPF were 6.32 ns and 0.12 ms.

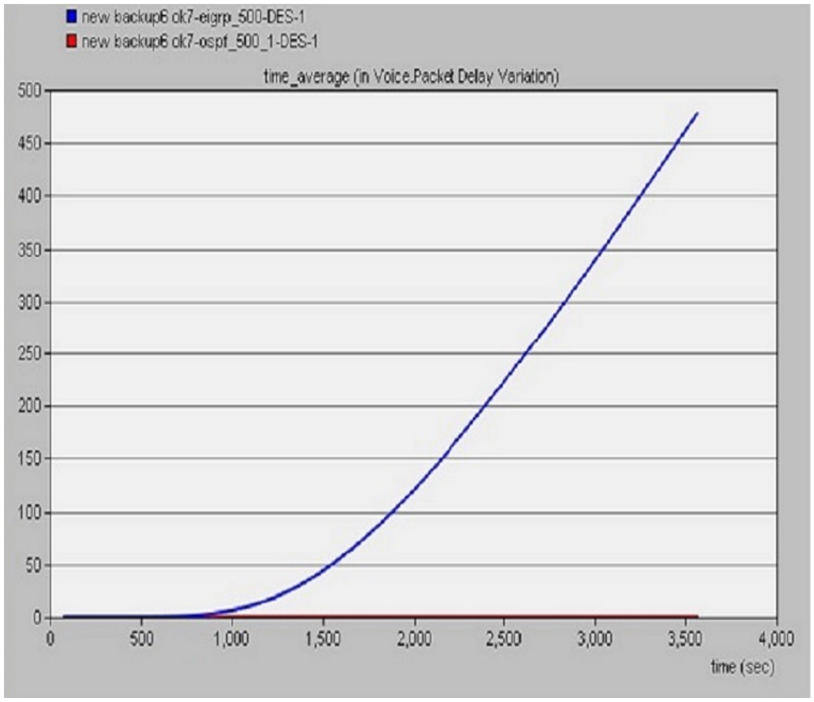

Figure 4.14: Packet Delay Variation (sec) for = 500 VoIP calls

From the simulation results shown in Figure 4.14, Figure 4.14 and Figure 4.14 it can clearly be seen that the OSPF has showed significant performance as compare to EIGRP for different metrics like jitter and packet delay variation which are very noticeable factors when we try to achieve QoS into a network. OSPF will be our new routing protocol instead of EIGRP when the redesigning of the network will take place. It is a fact that OSPF configuration is slightly difficult than EIGRP but the performance of OSPF is way better than EIGRP

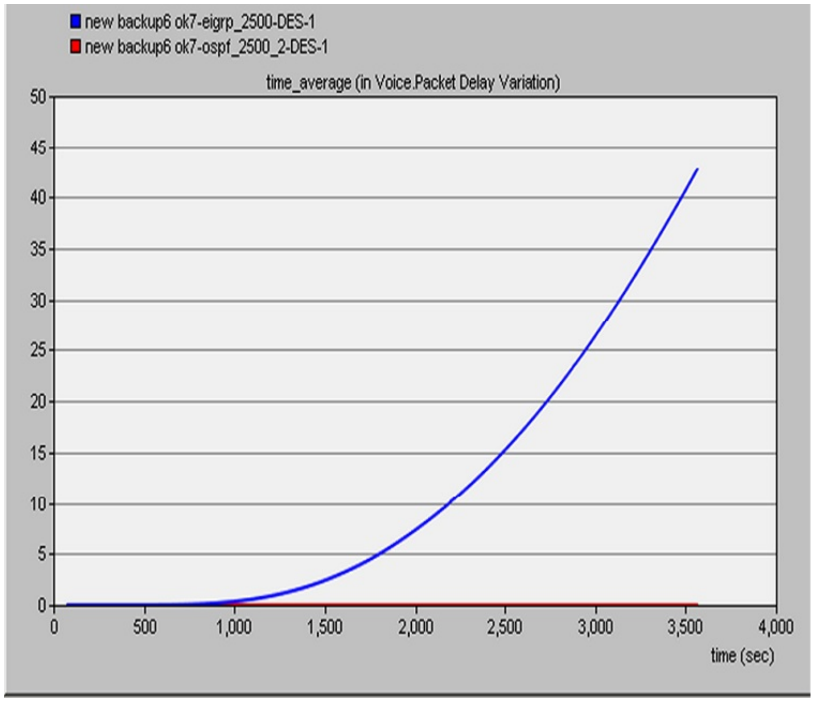

Figure 4.15: Packet Delay Variation (sec) for = 2500 VoIP calls

4.5 Selection of video codec for the redesigned network

As discussed in the previous sections that how important the selection of voice codec and routing protocol is in order to achieve the QoS across the network. Similarly another requirement for improving the network performance is to introduce a centrally managed video conference application in the network. So before introducing any video conference application into the network it is really necessary to pursue a justified research for selecting a video codec which is highly recommended for maintaining QoS. Number of research publications have shown that H.264 is highly recommended for the video applications [11]. There are some other codecs which have been introduced in recent years for example Theora and VP8 (owned by Google) but their performance is still slightly less as compare to H.264. Frame counts , PSNR over compression ratio for VP8 is closely comparable to H.264 [11] but H.264 compression techniques are still way better than other video codecs. So for fulfilling the requirements of video conference a compact solution of GoTo meeting is suggested for the redesigned network as it uses H.264 at the back end. GoTo meeting is preferred on other solutions because it is extremely cost effective

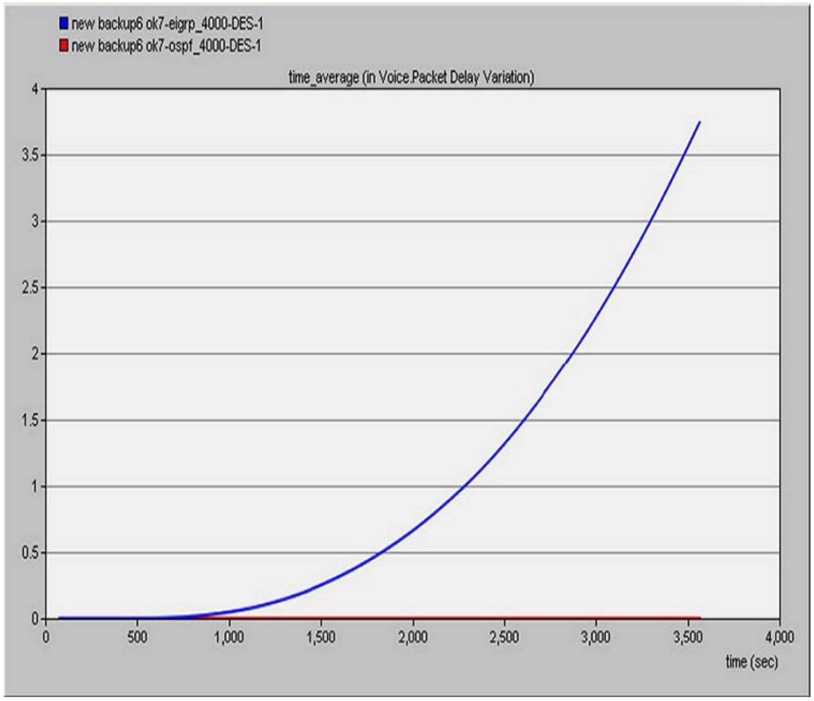

Figure 4.16: Packet Delay Variation (sec) for = 4000 VoIP calls

and easy to manage by IT across the network. H.264 is available with two different bandwidth levels so we can use H.264 level 3.2 (XP) in the enterprise office and H.264 level 1.2 (XP) for the branch office users as branch offices will not be hosting the same number of calls as the enterprise office so we can save some bandwidth for hosting other applications. The maximum bit rate required for H.264 Level 3.2 (XP) encoding is 20 Mb/s and the maximum bit rate required for H.264 Level 1.2 (XP) encoding is 384 kb/s. Furthermore, the redesigned network needs to a have DSCP video profile configured at each hop of the network in order to manage video sessions between enterprise and branch office.

4.6 Selection of appropriate QoS architecture

Quality of service (QoS) classifies packets on their type and defines priorities for each packet, it is achievable by different methods. These methods differ with each other in their level of QoS strictness, which describes how tightly the service can be bound by specific bandwidth, delay, jitter, and loss characteristics. The most popular methods to achieve end to end QoS across are discussed in detail in the below sections.

4.6.1 Integrated services (Guaranteed service)

It is also referred as hard QoS because of strict bandwidth reservations. It guarantees per-flow QoS. It needs signaling to accomplish path reservation (end-to-end) through a network for each application's packets similar to Asynchronous Transfer Mode (ATM). It does this using the Resource Reservation Protocol (RSVP), which dynamically maintains a path for each application's packets through a network, using the resources (Layer 3 switches) with the lightest load [7] [8]. This state is maintained as a flow, with an associated policy for admitting traffic to the network, and pre-determined packet handling characteristics at each hop. For proper results it must be configured on every router along with the path.

4.6.2 Differentiated service or Diff-serv(Soft QoS)

It refers to the method in which some traffic will be treated in better way than the rest (faster handling, more bandwidth on average, lower loss rate on average). This is a statistical preference, not a hard and fast guarantee. It scales well with large flows through aggregation. It is effective for traffic conditioning (TC) [8]. It works on per-hop behavior (PHB). Traffic conditioning is performed by the edge nodes. It allows core routers to do more important processing tasks. Differentiated services use the method of class of service (CoS) for categorizing the traffic into different classes and apply QoS parameters to those classes. Packets are first divided into classes by marking the Type of Service (ToS) byte in the IP header and then ToS byte is replaced with differentiated services filed for forwarding the traffic according to the priority pf the packets defined across the network. Figure 4.17 shows the replacement of IPV4 ToS field with differentiated services filed.

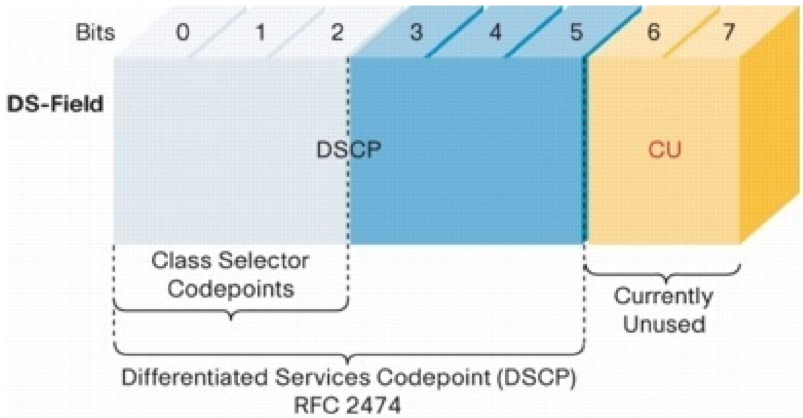

Figure 4.17: DiffServ Code point Field [12]

To deliver end to end QoS Differentiated services architecture has two major components.

- Packet marking

- Per Hop behavior

Packet Marking

The boundary node goes through the process of traffic classification as shown in Figure 4.18. Each component mentioned in Figure 4.18 also known as Differentiated services traffic block (TCB). In the above process the packets are marked in way to be treated accordingly within a DS domain the details of each component is as follows.

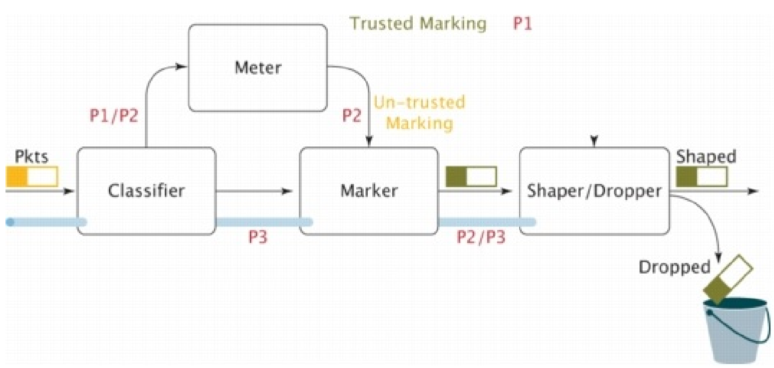

Figure 4.18: Differentiated services traffic conditioner block [12]

- Classifier– Categorizes the incoming packets into pre-defined aggregates [12].

- Meter – Measures packets to determine compliance to traffic parameters [12].

- Marker – According to the classification and meter, the packets are marked in their DCSP [12].

- Shaper/Dropper – Either buffers (if trying to achieve target flow rate) or drops a packet (if congestion) [12].

Per Hop Behavior

Packets that has same DSCP value and crossing in a particular direction is called Behavior Aggregate (BA) .Per hop behavior is referred to the packet scheduling, queuing, policing and shaping the behavior of a node on any packet belonging to a Behavior Aggregate .There are four standard of PHB which are available to construct a DiffServ enabled network to achieve end to end (CoS) and (QoS) [12]. The four standards are.

- Default PHB – It is also known as Best effort used for data traffic. The DSCP value used for this PHB is"000000".On the other hand routers will treat any packet that is not marked with DSCP value as default per hop behavior [12].

- Class-Selector PHB – In class selector DSCP has value from “xxx0000”, where x=0 or x=1. It was introduced for backward compatibility. The PHB associated with class-selector code point will retain to the same forwarding behavior as the node that implemented IP-precedence based classification and forwarding [12].

- Expedited Forwarding PHB – This service assures bandwidth with no loss and used in Areas where we need connectivity without loss and jitter .This service is best for voice traffic where we need a network with is free from delays and assures a dedicated bandwidth for these services [12].

- Assured Forwarding PHB – Assured forwarding has four classes assigned in three different levels of drop precedence Figure 4.19 shows a table for the DSCP values for each class and drop precedence [12].

DROP Precedence	Class #1	Class #2	Class #3	Class #4
Low Drop Precedence	(AF11) 001010	(AF21) 010010	(AF31) 011010	(AF41) 100010
Medium Drop Precedence	(AF12) 001100	(AF22) 010100	(AF32) 011100	(AF42) 100100
High Drop Precedence	(AF13) 001110	(AF23) 010110	(AF33) 011110	(AF43) 100110

Figure 4.19: DiffServ AF Code point Table [12]

4.6.3 Comparison between Integrated and Differentiated Services)

From the above explanation we can summarize that the methodology of Diff-serv will be a great choice for our redesigned network because we achieved all the parameters (Voice, Routing protocol and Video) under Diff-serv configuration. The above discussion strengthens our approach for using Diff-serv as a methodology to achieve end to end QoS across our redesigned network. Due to the soft QoS nature of Diff-serv the network traffic will be controlled through classifying the network traffic so priority will be given to the particular type of traffic which is delay sensitive i-e voice and video [8].

Chapter 5

Redesigning of Medium Scale Network

This chapter discusses the configuration and implementation of all the parameters (Voice codec, Video codec, Routing protocol and QoS architecture) that are selected for better performance for redesigning the medium scale network shown in Figure 3.1. This is the most critical part of the whole project because it contains step by step process for introducing QoS into the network. The existing infrastructure is redesigned using GNS3 because OPNET does not have the capability to configure each hop according to the design requirements. The network is redesigned in an effective manner that every medium scale organization can use the existing step by step approach and the device configurations to implement QoS within their medium scale network and can save a cost of design and device configurations which is usually very high these days. Router's configuration is not vendor specific so the Network administrator can use the configuration according to the hops in the network.

Figure 5.1 shows that OSPF is implemented across the whole network which is preferred after analyzing the above performance vectors for the complex routing. In Figure 5.1 design the enterprise office is connected to the branch office through MPLS connection. The network is divided into different areas according to the traffic routing requirements. The routers are classified as edge and core routers for different purposes. Edge routers will be responsible for marking packets as they enter into the network through connected users and will be forwarded according to the per hop behavior configured through core routers. The edge routers are R1, R2, R7, and R8, whereas R3, R4, and R5 are the core routers. Each area is designed with the particular bandwidth requirements

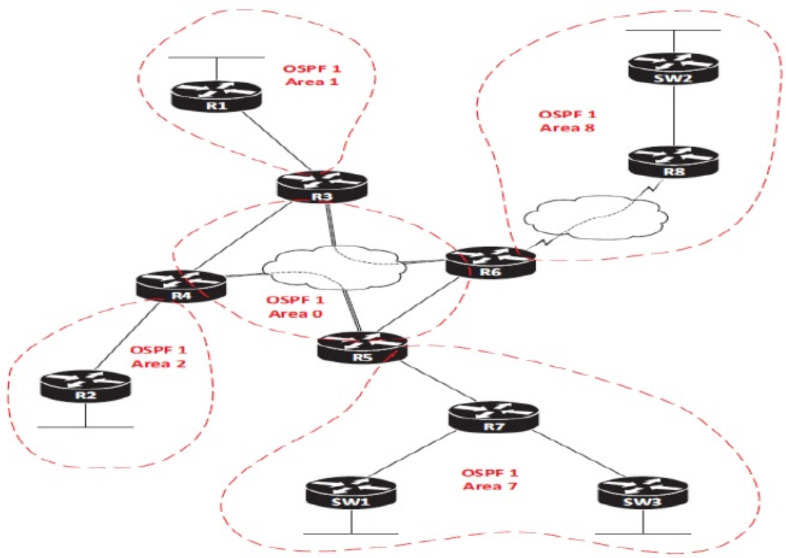

Figure 5.1: Redesigned Network

for routing the traffic smoothly. Each router is configured under a specific QoS profile which allows each hop to maintain QoS across the network. The network diagram shown in Figure 5.1 is redesigned in two phases i-e Phase I focuses on the calculation for the bandwidth requirements for supporting multimedia applications and the second phase is achieving QoS through Differv across the network.

5.1 Bandwidth calculation to support multimedia applications across the network

The first phase for redesigning our network is to calculate the bandwidth required to support multimedia applications (Voice , Video and Data). In Figure 5.1 OSPF is configures for each router. As OSPF works with the concept of Areas for routing the traffic across the entire network the first step is to calculate the bandwidth required in each area for supporting the multimedia traffic.

5.1.1 Bandwidth calculation for Area 1 (Enterprise Data Center)

For Area 1 there is no need of any complex bandwidth requirement as other areas as it only deals with traffic which is not delay sensitive. It deals with data traffic to send and receive on best effort because it deals with enterprise data center only.

5.1.2 Bandwidth calculation for Area 2 (Enterprise Media Server)

This Area has enterprise media server and is responsible for the implementation of VoIP and video conferencing functionalities. This server is responsible for managing the VoIP calls and video traffic between different branches. VoIP calls will be compressed and decompressed using the G.729 (8 kbps). For Area 7, 1000 simultaneous calls need to be managed by the Enterprise Media Server and 500 simultaneous calls for the users in Area 8. For video traffic the server uses the codec H.264 standard for video compression. H.264 comes in two different levels which are H.264 level 3.2 (XP) and H.264 level 1.2 (XP). H.264 level 3.2 (XP) operates on a bit rate of 20Mb/s on the other end H.264 1.2 (XP) follows a bit rate of 384Kb/s. For Area 2 video traffic from area 7 users will be H.264 level 3.2 (XP) because they are the high end users while H.264 Level 1.2 (XP) is configured for area 8 video sessions as they do not require high quality video streaming. Bandwidth calculation for multimedia sessions for area 2 is given below.

For Video Sessions

- H.264 Level 3.2 (XP) encoding = 20Mb/s
- H.264 Level 1.2 (XP) encoding = 384Kb/s

For VoIP Sessions

- Standardized bit rate for G729 is 8Kb/s
- For 1000 simultaneous phone calls coming from Area 7 to Area 2 and 500 simultaneous phone calls coming from Area 8 to Area 2((1000+500) * 8 kb/s) = 12Mb/s
- Full duplex (2 *12 Mb/s) = 24 Mb/s

For Data Sessions

- It is a general practice of cisco that 25 percent of the total link capacity should be reserved for data traffic.

- **Total bandwidth for Area 2 will be (20Mb/s + 384Kb/s + 24Mb/s) / 0.75 = 60 Mb/s**

5.1.3 Bandwidth calculation for Area 7 (Enterprise users)

Area 7 routes the multimedia traffic for Enterprise users which needs to be treated on the priority basis than the branch traffic so this area is designed with the high availability bandwidth requirements because enterprise users should not be effected at any cost.Sample bandwidth calculation for multimedia sessions for area 7 is given below.

For Video Sessions

- H.264 Level 3.2 (XP) encoding = 20Mb/s

For VoIP Sessions

- Standardized bit rates for G729 is 8Kb/s
- For 1000 simultaneous phone calls coming from Area 7 to Area 0 (1000 * 8 kb/s) = 8Mb/s
- Full duplex (2 * 8 Mb/s) = 16 Mb/s

For Data Sessions

- It is a general practice of cisco that 25 percent of the total link capacity should be reserved for data traffic.
- **Total bandwidth for Area 7 will be (20Mb/s + 16Mb/s) / 0.75 = 48 Mb/s**

5.1.4 Bandwidth calculation for Area 8 (Branch users)

Area 8 is responsible for the multimedia traffic of branch users which is then routed further areas through area 0. Branch users are the second priority users so that they do not require high availability of bandwidth for multimedia applications.

For Video Sessions

- H.264 Level 1.2 (XP) encoding = 384Kb/s

For VoIP Sessions

- Standardized bit rates for G729 is 8Kb/s
- For 500 simultaneous phone calls coming from Area 8 to Area 0 (500 * 8 kb/s) = 4Mb/s
- Full duplex (2 * 4 Mb/s) = 8 Mb/s

For Data Sessions

- It is a general practice of cisco that 25 percent of the total link capacity should be reserved for data traffic.
- **Total bandwidth for Area 8 will be (384Kb/s + 8Mb/s) / 0.75 = 12 Mb/s**

5.1.5 Bandwidth calculation for Area 0 (Backbone area)

All the traffic passes through this core network which lies in area 0 and it should have a capability to handle any type of data . Area 0 has the same bandwidth requirement than that of area 2.

- **Total bandwidth for Area 0 will be (20Mb/s + 384Kb/s + 24Mb/s) / 0.75 = 60 Mb/s**

Table 5.1: Recommended Bandwidth for each specific OSPF Area

Traffic Type	**Area 0 BW**	**Area 1 BW**	**Area 2 BW**	**Area 7 BW**	**Area 8 BW**
Video sessions	22.4 Mb/s		22.4 Mb/s	22 Mb/s	423 kb/s
VoIP calls	26.4 Mb/s		26.4 Mb/s	17.6 Mb/s	8.8 Mb/s
Data Traffic	16.28 Mb/s		16.28 Mb/s	9.9 Mb/s	2.31 Mb/s
Total	65.1 Mb/s	Best Effort	65.1 Mb/s	49.5 Mb/s	11.5 Mb/s

5.1.6 Recommended Bandwidth

For Enterprise office

- Video Traffic: 22.4 Mb/sec, the maximum video bandwidth among the areas.
- Voice Traffic: 26.4 Mb/s, to overcome peak time traffic. Extra Packets will be treated as best effort if EF traffic exceeds this rate.

For Branch office

- Video Traffic: 423 kb/sec, the maximum video bandwidth among the areas.
- Voice Traffic: 8.8 Mb/s, to overcome peak time traffic. Extra Packets will be treated as best effort if EF traffic exceeds this rate.

Table 5.1 summarizes the recommended bandwidth requirement for supporting multimedia sessions for each OSPF Area.

5.2 Implementation of Differentiated Services (Diffserv)

Differentiated services has been already been discussed in details in previous sections and now this section covers the implementation of Diffser to achieve Quality of service. Diffserv is implemented in to our network in two phases first we classified and marked the packets according to the application priority then each hop is configured with certain QoS profile in order to maintain end to end quality of service across the network [8]. As mentioned above the edge routers are responsible for DSCP marking and once the packets are classified and marked then they are routed to the core routers in multiple OSPF areas.

5.2.1 Edge Routers Configurations (Router 2)

This router marks the DSCP on frame of each of the following traffic packets came from the source.

- AF31 on video packets UDP (H264 port 1720)
- AF41 on video data packets UDP (RTP port 49152)
- AF31 on video signaling UDP (RTCP port 49153)
- EF on voice packets UDP (RTP port 16384)
- AF31 on voice signaling UDP (RTP port 16385)
- Default on any other packets.

5.2.2 Edge Routers Configurations (Router 7)

This router marks the DSCP on frame of each of the following traffic packets came from the source.

- AF41 on video packets UDP (H264 port 1720)
- AF31 on video signaling packets UDP (RTCP port 49153)
- EF on voice packets UDP(RTP port 16384)
- AF31 on voice signaling UDP(RTP port 16385)
- Default on any other packets.

5.2.3 Edge Routers Configurations (Router 8)

This router marks the DSCP on frame of each of the following traffic packets came from the source.

- AF41 on video packets UDP (H264 port 1720)
- AF31 on video signaling packets UDP (RTCP port 49153)
- EF on voice packets UDP(RTP port 16384)
- AF31 on voice signaling packets UDP(RTP port 16385)
- Default on any other packets.

5.2.4 Per Hop Behavior (PHB) configuration

The second phase of Diff-serv implementation is to configure each hop in an appropriate manner that they can route the traffic among different routers according to the DSCP marking and maintain end to end QoS. Per Hop Recommended bandwidth is usually defined in SLA (Service Level Agreement) provided to the ISP in order to maintain smooth routing of traffic. SLA guarantees to route and pass the traffic to/from company's routers without effecting internal design that support QoS.

For Enterprise office

- Video Traffic: Configure the Enterprise routers R2 and R7 to use the bandwidth of 22.4 Mb/s for video traffic and drop threshold limit of 150 kbps – 350 kbps
- Voice Traffic: Configure the Enterprise office routers R2 and R7 to treat the voice traffic as best effort when it goes beyond the limit of 26.4 Mb/s (Peak hours) by 450 kb/s, and will be dropped as it exceed the limit by 650 kb/s

For Enterprise office

- Video Traffic: Configure the branch router R8 to use the bandwidth of 423 kb/s for video traffic and drop threshold limit of 150 kbps – 350 kbps
- Voice Traffic: Configure the branch router R8 to treat the voice traffic as a best effort when it goes beyond the limit of 8.8 Mb/s by 450 kb/s, and will be dropped as it exceed the limit by 650 kb/s

The complete configuration of each router is given below which will maintain the end to end QoS across the network. The router configuration is tested on a free simulation tool VIRL from Cisco.

5.2.5 Configuration of Edge Routers

The Edge Routers (R2, R7 and R8) used first as a marker devices. Packets should be marked as DS value 46 (EF class), DS value 34 (AF41 class), DS value 26 (AF31 class) and DS value 0 (BE).

First using access-list commands for data permission

access-list 10 permit udp any any eq 16384 ** voice
access-list 20 permit udp any any eq 49152 **video
access-list 30 permit udp any any eq 16385 **audio signaling
access-list 40 permit udp any any eq 49153 **video signaling
access-list 50 permit ip any any **regular traffic

Now defining class map

class-map VoIP ** voice
match access-group 10
exit

class-map video **video
match access-group 20
exit

class-map audio signaling **audio signaling
match access-group 30
exit

class-map video signaling **video signaling
match access-group 40
exit

class-map regular **regular traffic
match access-group 50
exit

Defining policy map and decimal value of DSCP

policy-map Group 8

class VoIP **voice
set ip dscp 46
exit

class video **video
set ip dscp 34
exit

class audio signaling **audio signaling
set ip dscp 26
exit

class video signaling **video signaling
set ip dscp 26
exit

class regular **regular traffic
set ip dscp 0
exit

Deploying policy maps

policy-map UVIC

class VoIP **voice
UVIC 26400000 4500 6500
conform-action set-dscp-transmit 46

```
exceed-action set-dscp-transmit 0
violate-action drop
exit

class video **video
bandwidth 22400000
random-detect dscp 34 150 350
violate-action drop
exit

class audio signaling **audio signaling
bandwidth 600000
random-detect dscp 26 150 180
violate-action drop
exit

class video signaling **video signaling
bandwidth 600000
random-detect dscp 26 150 180
violate-action drop
exit

class regular **regular traffic
bingo 16200000 1750 1750
conform-action set-dscp-transmit 0
exceed-action drop
exit
```

5.2.6 Configuration of Edge Routers

The core routers (R3-R6) deal with already marked packets, so we create a traffic policy to treat and forward packets according to the proposed PHB.

Defining traffic policy

class-map match-all VoIP **voice
match ip dscp 46
exit

class-map match-all video **video
match ip dscp 34
exit

class-map match-all audio signaling **audio signaling
match ip dscp 26
exit

class-map match-all video signaling **video signaling
match ip dscp 26
exit

class-map match-all regular **regular traffic
match ip dscp 0
exit

Deploying policy maps

policy-map UVIC

class VoIP **voice
bandwidth 26400000
priority 500
exit

class video **video
bandwidth 22400000
random-detect dscp-based
random-detect dscp 34 150 350
exit

class audio signaling **audio signaling
bandwidth 600000
random-detect dscp-based
random-detect dscp 26 150 180
exit

class video signaling **video signaling
bandwidth 600000
random-detect dscp-based
random-detect dscp 26 150 180
exit

class regular **regular traffic
bingo 16200000 1750 1750
conform-action set-dscp-transmit 0
random-detect dscp-based
exit

Chapter 6

Conclusion

This draft provides us an explicit idea that how Quality of Service can be achieved in any size of network within the constrained resources. This document reflects all the approaches that are actually needed for any kind of network architecture. In today's world corporates are looking for viable solution within their constraint budget and this designed is a true depiction how we can achieve multiple functionalities by spending very less amount of money. Usually solution architects provide massive quotes for introducing multiple services into the Network which is not really appreciated by the medium size companies so with the above solution an IT engineer can easily introduce some vital changes into the network by referring to the approaches mentioned in this document. The simulations are carried out in the real time environment which are almost similar to what medium size corporates are expecting these days.

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