Performance Analysis of MultimediaTraffic in DiffServ Network

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***Abstract*—**Increased integrated traffic of voice, video and data on Internet/Multi Protocol Label Switch (IP/MPLS) core network demands end-to-end quality of service (QoS). Differentiated services (DiffServ) integrated with MPLS technologies in the multi service networks are envisioned to offer guaranteed QoS. Different traffic engineering (TE) techniques need to be exploited with DiffServ technologies to provide end-to-end performance. In this paper, we analyzed the integrated traffic end-to-end delivery performance on DiffServ IP Networks. Experiments are conducted using Omnet++ integrated INET by considering different data rates of video and voice traffic from multiple hosts.We also consider different capacities of bottleneck link through which all the traffic is traversed providing a congested link. Various experiments are conducted by allocating resources of a network to different forwarding classes.The parameters such as end-to-end delay, packet lost, queuing time and queuing length are analyzed for the voice and video packets. Results show significant improvement on QoS for such traffic.

Keywords–DiffServ; QoS;End to end delay, Bandwidth utilization; Schedulers.

I. Introduction

Networking companies are facing a big challenge to efficiently manage the spike in multimedia traffic which includes nearly 30 hours of video every minute alongwith voice, image and text data [1]. With such applications, demand of increased network performance is continuously growingand it also imposes stringent QoS requirements. DiffServ is scalablenetwork architecture [2] for classifying and managing network traffic to provide QoS for multi service networks. Critical network traffic such as voice and streaming media can beprovided with low latency and low endtoend delay while web traffic or simple file transfers can be provided with best effort services.

Although, QoS services can be guaranteed by offering high bandwidth or by over provisioning of bandwidth but results in an expensive solution for the service providers. DiffServ QoS approach utilizes the available resources in efficient manner to provide QoS. But it has several issues related to fair distribution of network capacity among all users. The issue is more stringent when a complex network topology contains a bottleneck link and also when the video and voice packets are continuously generated by all the hosts at a constant rate.

This paper presents such a network topology and analyzes its performance in different scenarios by varying the capacity of bottleneck link, datarate generated by all the hosts and keeping different message lengths of the packets. Extensive simulation is carried out to discuss the issues of fairness in allocating resources to the customers based on service level agreements (SLA) by changing assignments of forwarding classes (FC) to different hosts, by using different scheduling schemes and traffic conditioning mechanisms.

The rest of the paper is organized as follows. Section II presents DiffServ-based QoS model.In section III, the performance analysis is discussed. Simulation results are presented in section IV while section V concludes the paper.

II. DiffServ Based QoS Model

Differentiated services model is based on design paradigm which places complexity of traffic processing functions like multi field classification and traffic policing at the edge of the network. In DiffServ architecture, traffic is classified based on different classes. Each traffic class can be managed differently to ensure preferential treatment for higher priority traffic on the network. When a packet is received at the edge router, it is classified based on SLA [3] and checked for misbehaving traffic sources (ie. if sending rate > committed rate by service provider). Based on over-sending conditions, the packet is dropped or sent/delayed. After this the packet is marked as DSCP (Differentiated Services Code Point) in type of service (TOS) field of IP header to determine per hop behavior (PHB) if it is not dropped[3].

A PHB is implemented with buffer (queue) management and packet scheduling mechanisms. Thisqueue management and scheduling mechanism [4]are used by the router to provide service differentiation among different forwarding classes.After packets are marked with their forwarding classes at the edge of the network, the interior nodes of the network can use this information to differentiate the treatment of packets. The forwarding classes may indicate drop priority and resource priority. By marking the packets, we can also protect the domain from misbehaving traffic sources[5].

Fig. 1shows the architecture of DiffServ network [6]. The main elements of DiffServ networks are DS domains and the DS boundary nodes. The DS- domains may be private intranets. The DS boundary nodes exist at the edge of the DiffServ network as either ingress or egress nodes. The ingress node is more complex and performs traffic conditioning operation to forward the traffic into the network. Traffic conditioning rules are specified in traffic conditioning agreement (TCA).DiffServ performs traffic conditioning to ensure that the traffic entering the DS domain conforms to the rules specified in TCA. The DiffServ TCA consists of a classifier which classifies the incoming packet into pre-defined aggregates, a meter to segregate in-profile and out-of-profile packets, a marker to write the DSCP codes and shaper/dropper to delay the packets to achieve target flow rate/drops the packet in case of congestion[4].The DSCP codes determine PHBs which describe the type of services a certain packet receives.Most networks use the following commonly defined PHBs:

- Default PHB - The packets which receive this type of behavior are forwarded without any priority, equal to best effort forwarding. All packets that are not assigned to any standard PHB should be assigned to default PHB.

- EF (expedited forwarding) PHB - This standard PHB provides low delay and low loss policy. Applications like VoIP which requires low end to end delay and low loss can be achieved by assigning its packets as EF. The standard

TABLE I: STANDARD DSCP CODES

DSCP	Forwarding Classes
Expedited Forwarding	101110
Best Effort	000000
Network Control	110000

TABLE II DSCP FOR DIFFERENT AF CLASS OF SERVICE

Drop Precedence	Class 1 (AF1x)	Class 2 (AF2x)	Class 3 (AF3x)	Class 4 (AF4x)
Low Drop Precedence	001010 (AF11)	010010 (AF21)	011010 (AF31)	100010 (AF41)
MediumDrop Precedence	001100 (AF12)	010100 (AF22)	011100 (AF32)	100100 (AF42)
High Drop Precedence	001110 (AF13)	010110 (AF23)	011110 (AF33)	100110 (AF43)

DSCP codes for EF, best effort and network control are shown in Table I.

- AF (Assured Forwarding) PHB - AF PHB group provides forwarding in *k* independent AF classes. Within each class, IP packet gets one of *L* different levels of drop precedence assigned. AF implementation handles short term

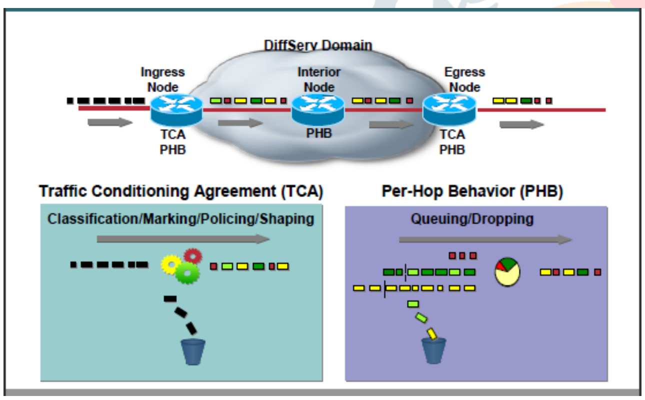

Fig.1: Differentiated Services Architecture

congestion by queuing packets and responds to long term congestion within each class by dropping packets. Currently four AF classes with each three levels of drop precedence are defined as shown in table II:

III. EXPERIMENT SET UP AND PERFORMANCE ANALYSIS

Anexhaustive simulation study is undertaken on Omnet++ version 4.2.2[6] using INET 2.1.1[7] by considering various experiments. A relatively complex topology as given in Fig.2 is considered. The topology consists of 16 hosts and 14 routers. Hosts are connected to the routers by 100MbpsEthernet links. The routers are also connected by point to point (PPP) links of comparatively lower data rates. A bottleneck link is created in such a way that all the connections traverse through that link.

Voice traffic is generated by user data packet (UDP) basic burst application which sends UDP packets in bursts to the destination. 172 bytes long packets (160 bytes data + 12 bytes of RTP header) are sent in every 20ms in the burst phase with mean burst duration of 0.352s and mean sleep duration of 0.650s[7]. Voice requirements are stern and for real time communication one way delay must be in the range of 100-150ms for satisfactory performance. The end-to-end delay of voice packets consists of the delay incurredby the voice signal from the instant when it is produced by the sender (speaker) till it gets heard by the listener at the destination. It includes packetization delay, transmission delay including store and forward delay and buffering delay. Video streaming traffic is generated by UDPBasicApp application and is modeled by 1Mbpsand 2Mbps constant bit rate (CBR) traffic from each host.

A number of experiments are performed to evaluate the designed network.

Comparison of Network with QoS and without QoS

Quality of service is to guarantee a certain level of performance to a data flow or to different services. A network that supports QoS may agree on a traffic contract between customer and service provider and reserve capacity in network. A best effort network does not use QoS.

In experiment 1, the aim is to observe the effects of different message length, different data rates of video and voice traffic and different link capacities of thebottleneck link on guaranteed services in with QoS (DiffServ) and without QoS (best effort) networks.DiffServ can guarantee voice requirements of low delay and high throughput as compared to the best effort network. Alsovoice traffic can be prioritized over video traffic which traverses through the same bottleneck linkeven though the link is made congested by increasing the video traffic rate to a high value.

The network is configured in such a way that video packets created at H1,H3,H5 and H7 are marked as AF11 while video traffic from H2, H4, H6 and H8 are marked as AF21, so the endtoend delay should be less for video packets initiated by H1, H3, H5 and H7. Also all the voice traffic is marked as EF traffic to provide guaranteed QoS. So the end-to-end delay of voice packets should be very less.

In this case, two set of readings are taken: i.) First the routers are configured to implement EF and AFxy forwarding classes. EF traffic (voice traffic) is allowed to pass through a priority scheduler and AFxy traffic (video traffic) is scheduled using a weighted round robin (WRR) scheduler. The different weights

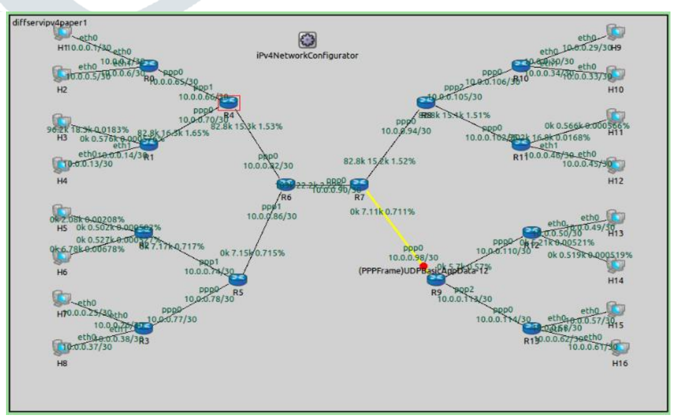

Fig. 2: Bottleneck link Network Topology

considered for WRR scheduler for AF11, AF21, AF31, AF41 and BE are (4,3,0,0,1) respectively. The variation in weights is considered for better results of differentiation in services;ii) Secondly, each PPP queue is considered to be a droptail queue in a network without QoS. Buffer space was taken equal in both Diffserv (with QoS) and best effort network (without QoS). Resources are kept constant in DiffServ and best effort network. Edgelinks are the links between R0-R4, R1-R4, R2-R5, R3-R5, R8-R10, R8-R11, R9-12 & R9-R13. Corelinks are the links between R4-R6, R5-R6, R7-R8, R7-R9. The capacity of edgelink and corelink are 10Mbps. Bottleneck link is the link between R6-R7 and in this case its value is 10Mbps. Drop tail queue size is 100, voice traffic datarate from each host is 80kbps peek rate and 28kbps average rate. Video traffic is sent at 1Mbps data rate from each host. Message length of video traffic is varied from 5000 bytes in 40ms to 500 bytes in 4ms. This large variation in the message length is aimed to offer better results in terms of throughput, packet end-to-end delay and queuing length.

IV. Results and Discussion

In this section, we present results of various experiments explained in section III and discuss the results of simulation.

Extensive simulations are carried out to analyze the results for experiments 1.1, 1.2. For each experiment graphs are plotted and effects of different message length, data rates and different capacities of bottleneck links are investigated.Different graphs for queuing time versus increase in traffic load with time and end-to-end delay for voice/video traffic versus increase in traffic load with time are plotted. Some of the important results are pictorially shown while consolidated results are included in the table.

A. *Comparison of Network with QoS and without QoS*

i) Effect of the message length

To examine the effect of message length , two values for video packets are taken, one 500 bytes of data packets in 4msproviding a data rate of 1Mbps from each host and other 5000 bytes of video packets in 40ms, providing same data rate. A PPP interface is used as bottleneck link so 5000 bytes are fragmented and forwarded through the network but message with length 500 bytes are forwarded without fragmentation. So end-to-end delay is an important parameter when message length changes. Table III discusses QoS parameters in details for different message length.Also in a QoS network if any forwarding class is given priority, the end-to-end delay for that has to be less and the queuing time of that forwarding class will also be less since packets need not wait in the queue and treated as important. So, queuing time is also an important parameter which can provide guaranteed end-to-end delay.

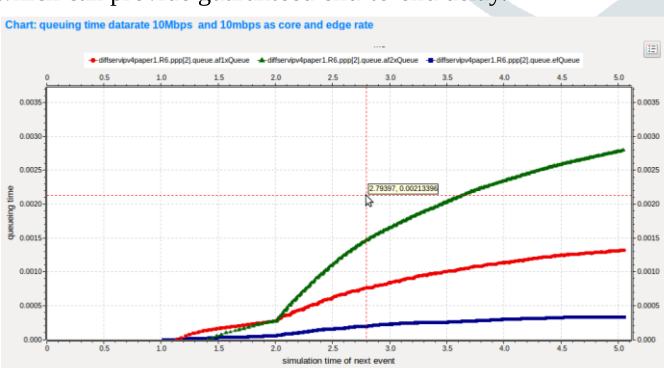

Fig.3: Queuing Time for Different Forwarding Classes in Exp.1.1

Fig.3shows the queuing time for different forwarding classes(EF,AF1x,AF2x) used in theExp.1.1 when video traffic is continuously generated at the rate of 1Mbps from all the 8 hosts and the message length is 5000 bytes. The x-axis shows the increase in traffic load with simulation time and y-axis is queuing time in sec. From the figure, it can be observed that the queuing time for EF (voice) traffic is 0.4ms or approximately negligible as compared to AF1x and AF2x (video) traffic. This is because EF traffic is passed through a priority scheduler and EF forwarding class is given precedence over other forwarding classes in DiffServ network. So the packets need not wait in queue. Similarly, AF1x is given more weight than AF2x, so queuing time for AF1x is 1.4ms and is less than AF2x which has 2.7ms as the queuing time.

Table III compares the performance of network for Exp.1.1 and Exp.1.2. For message length 5000 bytes, throughput is almost similar for both with QoS and without QoS networks but the end- to-end delay values are slightly more in Exp.1.2. But when the message length is 500 bytes, the throughput for both voice and video packets degrades in Exp 1.2and the end-to-end delay values are similar for both with QoS and without QoS networks.

TABLE III: Effect of Different Message Length

Exp No.	Message Length	Throughput=Successful packets/sent packets(%)			Packet End-to-end delay (ms)	
		Voice Traffic	Video Traffic	Total Traffic	Voice Traffic	Video Traffic)
Exp 1.1 (with QoS)	5000 bytes in 40ms (1Mbps)	99.78	99.27	99.48	12.49	23.81
	500 bytes in 4ms (1Mbps)	99.51	99.62	99.62	11.39	12.48
Exp 1.2 (without QoS)	5000 bytes in 40ms (1Mbps)	99.78	99.56	99.65	15.85	22.81
	500 bytes in 4ms (1Mbps)	93.50	93.63	93.63	11.67	12.51

ii) Effect of Data rate

Fig 4 shows end-to-end delay of voice and video packets in Exp.1.1 for 2Mbps data rate. The end-to-end delay for voice packets is very small as these packets are treated as EF class. The video packets from H1, H3, H5 and H7 has less end-to-end delay (red dark lines) as compared to video packets from H2, H4, H6 and H8(Blue and purple lines). This is because packets from H1, H3, H5 and H7 are treated as AF1x forwarding class and given slightly higher weight than AF2x class.

Similarly, in Fig.5 end-to-end delay of voice and video packets for the Exp.1.2 for 2Mbps data rate is shown. The end-to-end delay of video packets from different hosts is the same as shown by thick line.

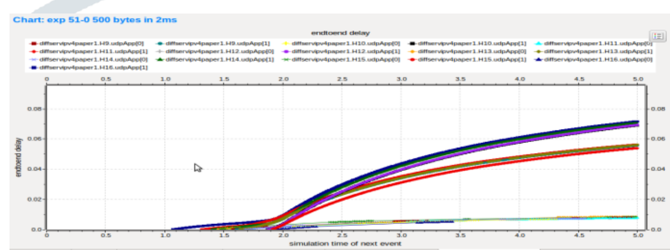

Fig 4: End to end delay of both voice and video packets in Exp 1.1 for 2Mbps data rate

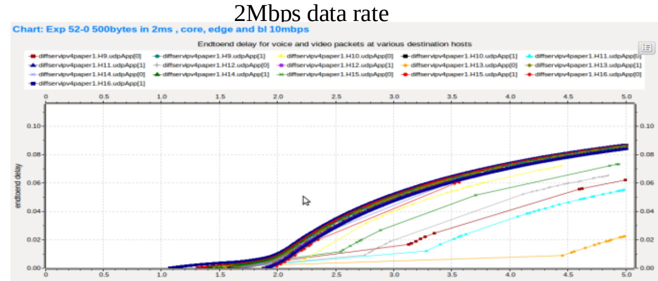

Fig 5: End to end delay of both voice and video packets in Exp 1.2 for 2Mbps data rate

TABLE IV: EFFECT OF DATA RATE

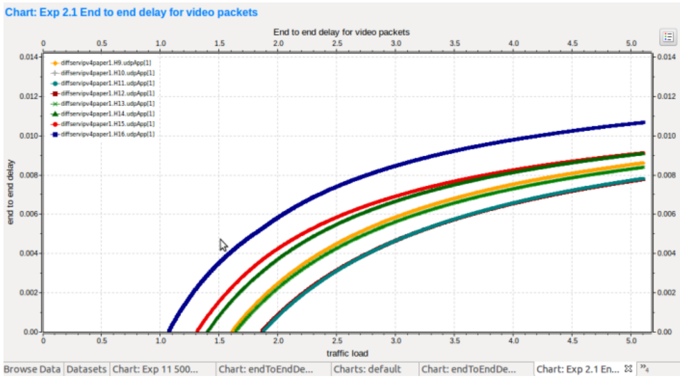

Fig 6: End to end delay of video packets in Exp. 1.1 for 1Mbps data rate (500 bytes in 4ms) when bottleneck link capacity is 10Mbps

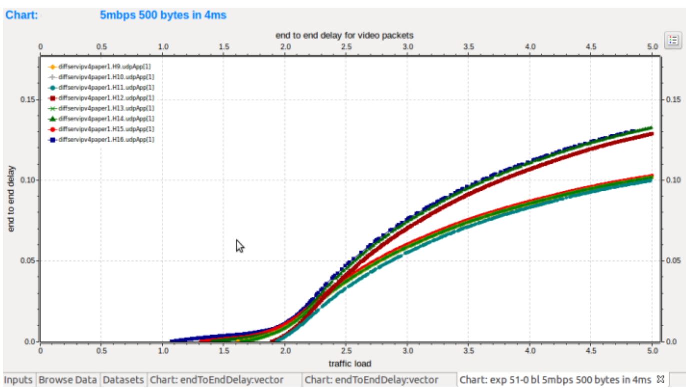

Fig 7: End to end delay of video packets in Exp.1.1 for 1Mbps data rate (500 bytes in 4ms) when bottleneck link capacity is 5Mbps

Exp No.	Data Rate	Throughput =Successful packets/sent packets(%)			Packet End to end delay (in ms)	
		Voice Traffic	Video Traffic	Total Traffic	Voice Traffic	Video Traffic)
Exp 1.1 (With QoS)	500 bytes in 2ms (2Mbps)	99.55	60.2	61.45	11.59	88.2
	500 bytes in 4ms (1Mbps)	99.51	99.62	99.62	11.39	12.48
Exp 1.2 (Without QoS)	500 bytes in 2ms (2Mbps)	47.13	60.85	60.42	95.74	108.2
	500 bytes in 4ms (1Mbps)	93.50	93.63	93.63	11.67	12.51

Table IV shows the comparison of Exp.1.1 and Exp.1.2 for different data rates. Voice data rate is kept constant and video data rate is increased to double from each host. The results show that in a network with QoS, the throughput for voice traffic is almost similar for 1Mbps and 2Mbps data rate but it degrades drastically in case of network without QoS. Also the end-to-end delay value for voice traffic is very less in Exp.1.1 as compared to Exp.1.2. The throughput for video traffic is similar in Exp.1.1 and Exp.1.2 but with more end-to-end delay in Exp.1.2

Fig. 6 and Fig.7 show the end-to-end delay of video packets in Exp. 1.1 for 1Mbps data rate with bottleneck capacities of 10Mbps and 5 Mbps respectively. Although the data rates and allocation of resources (queue size, precedence order of packets) are similar in both the cases but due to variation in the capacity of the bottleneck link, the deviation in result is significant for the end-to-end delay. The average end-to-end delay of video packets is very high with 5Mbps link as compared to 10Mbps link. Also, when bottleneck bandwidth is more, the end-to-end delay for video packets received at hosts H9, H11, H13, H15 is less and differentfrom the end-to-end delay with 5mbps link. The experimental details are given in Table V which shows that the throughput in both cases isthe same.Hence,for the stringent applications where delay and bandwidth are main requirement for customers, we can send voice and video traffic through abottleneck link of 5mbps, provided message length is kept small.

TABLE V: EFFECT OF CAPACITY OF BOTTLENECK LINK RATE

Exp No.	Capacity of Bottleneck Link (Mbps)	Message length (Voice Traffic 172 bytes in 20ms)	Throughput =Successful packets/sent packets (%)			Packet End to end delay (ms)	
		Video (1Mbps)	Voice Traffic	Video Traffic	Total Traffic	Voice Traffic	Video Traffic
Exp 1.1 (With QoS)	10	5000 bytes in 40ms	99.78	99.27	99.48	12.49	23.81
	5	5000bytes in 40ms	99.67	99.13	99.38	123.2	204.5
	10	500 bytes in 4ms	99.51	99.62	99.62	11.39	12.48
	5	500bytes in 4ms	99.50	59.18	61.42	12.24	24.50
Exp 1.2 (Without QoS)	10	5000 bytes in 40ms	99.78	99.56	99.65	15.85	22.81
	5	5000 bytes in 4ms	62.76	37.74	47.81	243.3	306.7
	10	500 bytes in 40ms	93.50	93.63	93.63	11.67	12.51
	5	500 bytes in 4ms	51.82	60.24	59.77	149.1	204.5

Table V provides detailed analysis of throughput and end-to-end delay of voice and video packets traversed through a bottleneck link when its capacity is reduced to half. In Exp.1.1 experiment, when the message length is taken as 5000 bytes with 1Mbps data rate, the throughput for both voice and video packets are approximately same with 10Mbps and 5Mbps capacities of bottleneck link but the end to end delay has remarkable difference and is comparatively very high for both voice and video packets in 5Mbps bottleneck link. But on the other hand when message length is 500 bytes with 1Mbps data rate of video packets from each host, the throughput for video packets is reduced drastically in Exp.1.1 but the end to end delay and throughput for voice packets is almost similar. Also if the capacity of bottleneck link is reduced to 5Mbps from 10Mbps in 1.2 (without QoS) experiment then the performance is degraded drastically in terms of both throughput and end to end delay as a single droptail queue is used.

V. CONCLUSION AND FUTURE SCOPE

The end-to-end delay and throughput performance of multi service network for voice and video traffic is analyzed and evaluated. Different experiments are performed under various conditions to understand the behavior of the network and to visualize the QoS support of such network. It is seen that message length affects the end-to-end delay in QoS supported network. Similarly, the resource allocation also affects the end-to-end delay as well as throughput.

We are examining performance in more details with more complexity. Now IPv6 is being taken into consideration. We will integrate MPLS and DiffServ for both the cases of IPv4 and IPv6 to evaluate a complex topology and optimize its performance for QoS support on data traffic like real time, video conferencing and so on.

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