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RESEARCH ARTICLE

RECONFIGURABLE DIFFSERV BASED QOS MODEL FOR REAL-TIME TRAFFIC.

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Abstract

The main architectures of the Internet adopts the best-effort model for traffic control. It is an unable to guarantee a timely delivery of data. To address this problem, a “differentiated services” (DiffServ) is proposed by the IEFT (Internet engineering task force) as a simple and scalable solution for providing the quality of service (QoS) for Internet applications with various levels of services. In DiffServ, scheduling disciplines play an important role in achieving the required level of service in terms of various criteria (e.g., delay, jitter, throughput, and packet loss).

In this paper, a Reconfigurable Scheduler Model (RSM) for real-time traffic through DiffServ router has been designed and implemented using OPNET Modular. The proposed scheduler applies the three most widely accepted queue schedulers in DiffServ routers; Priority Queuing (PQ), Class Based Weight Fair Queuing (CBWFQ) and Modified Deficit Round Robin (MDRR). The voice traffic can be scheduled using strict priority queue scheduler while the video and data traffic are scheduled by the CBWFQ and MDRR schedulers which can be interchanged according to the queue delay of the video traffic to provide better QoS. The performance evaluation shows an improvement in QoS parameters of voice and video traffics.

Introduction:-

Quality of Service (QoS) is a measure of the ability of network and computing systems to provide different levels of services to selected applications within a network. It is one of the necessary items, especially in real time applications. Different kinds of applications and users require different degrees quality of service (QoS). Voice traffic is time sensitive and intolerant of delay, jitter and packets loss. Video is also intolerant of delay and packets loss, with additional complication of being very bursty at times of arrivals.

The main mechanisms to support QoS are the flow-based and the class-based. In flow-based approach, a certain amount of end-to-end resources must be devoted for each traffic flow in the network according to its specific QoS requirements.

This approach has been used by the Internet Engineering Task Force (IETF) in the Integrated Service (IntServ)[1] architecture and in Multi-Protocol-Label-Switching with Traffic Engineering (MPLS-TE) [2] networks.

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In the DiffServ network, packets are classified into several classes according to their QoS requirements, such as delay, throughput and drop precedence. These classes are called Per Hop Behaviors, or PHBs. The three types of PHBs or forwarding techniques have been defined in DiffServ network as[3]; Expedited forwarding (EF), Assured Forwarding (AF), and Beset Effort (BE).

In DiffServ networks, congestion management is achieved through traffic scheduling and queueing. For example; a scheduling algorithm such as CBWFQ is used to provide guaranteed bandwidth to the different classes of traffic. In this paper, a reconfigurable technique is used to design a DiffServ scheduler to improve the level of QoS for real-time multimedia applications (such as voice and video) at low service cost and making use of existing widely deployed mechanisms. The proposed reconfigurable scheduler employ two of the widely used scheduling mechanisms, MDRR and CBWFQ, alternatively during the packet scheduling process. Multiple traffics sources which start at different selected times are used to generate different aggregate load levels applied to the queues. The reconfiguration is occur during the operation of the hardware resources, this is described as dynamically or run time reconfiguration[4]. By observing the queueing delay of video traffic, the reconfigurable scheduler algorithms can be interchanged between MDRR and CBWFQ to meet minimum queueing delay and to improve specific performance requirements. Both MDRR and CBWFQ scheduling schemes relay on the same packet classification, weight definition scheme and queue structure are expected to simplify the reconfiguration process.

The structure of this paper is as follows. Section 2 introduces an overview of previous related works. Section 3 reviews the congestion management mechanisms in DiffServ network. In section 4, a QoS requirements for Voice, Video, and Data are presented. Sections 5 discusses traffic classes and bandwidth assigning. A description to the proposed QoS model is provided in section 7. The simulation scenarios and results by using OPNET are discussed in section 8. Finally, the conclusions are drawn in the last section.

Literature Review:-

A lot of research work has been done and continuing to provide better QoS with real-time traffic. Many approaches have been proposed to realize the DiffServ model. Some QoS implementation models are presented in the literature as follows.

To manage multimedia traffic in a DiffServ network T. Ahmed at el. [5] use network monitoring , feedback and control. Bandwidth monitors are installed in each node of DiffServ domain to organize the interaction with a policy server. According to the implementation of the selected policies, the multimedia traffic entering the network is accepted, remarked, or dropped.

I. McDonald [6] presents a model to prioritize important data in video conferencing. In his model, the packets are reorder or discard by the transport layer to optimize the use of the network. This algorithm is implemented as interface to the Datagram Congestion Control Protocol (DCCP), and it gives better improvements to video conferencing using standard UDP and TCP. Combining Weighted Fair Queueing (WFQ) and LLQ (Low Latency Queue) (WFQ-LLQ) scheduling disciplines to ensure the Quality of Service (QoS) for high priority bursty Video conferencing, Voice and Data services at the same time has been proposed by B. Dekeris *et.al.* [7].

To enhance the QoS performance of delay sensitive applications, J. Wang et al.[8] presents a comprehensive system modeling and analysis approach for both predicting queuing delay and controlling average queuing delay of a single buffer to a required value in a multiple traffic source network environment. A discrete-time analytical model is used to analyze the relationship between the queuing threshold and average queuing delay. A control strategy with dynamic queue thresholds based on the analytical result is then used to control the average queuing delay to a required value within the buffer. S.G. Chaudhuri [9] proposes an adaptive scheduling model for real-time traffic in Diffserv network. It is based on a mechanism which yields low packets loss and delay for real time QoS traffic requirements.

Congestion Management Mechanisms in Diffserv Network:-

Congestion can be minimized by limiting the amount of transmitted traffic, or by managing the buffers of the router using scheduling or queuing techniques. The three of the most important software queuing techniques used in DiffServ routers are Priority Queuing (PQ)[10], class-based weighted fair queuing (CBWFQ)[11], and modified Deficit round robin (MDRR)[12].

QoS Requirements For Voice, Video, And Data:-

The characteristics of voice, video and data are defined in terms of delay, jitter, and packet loss. These have to be well understood in order to determine the appropriate treatment to give each application in the network. The QoS Requirements for Voice, Video, and Data are[13]:

QoS requirements for voice:-

For good quality of voice at the destination side, the following requirements must be satisfied:

- One-way latency from mouth to ear (per the ITU G.114 standard) ≤ 150 ms
- Delay Jitter ≤ 30 ms
- Packets loss percentage ≤ 1
- Guaranteed priority bandwidth per call from 17 to 106 kbps
- Guaranteed bandwidth for voice control traffic equal to 150K bps per phone

QoS requirements for video:-

The requirements for **streaming video**, such as IP multicast, executive broadcasts, and real-time training activities, are as follows:

- Latency allowable is equal to 5 seconds.
- Packets loss permissible is equal to 2%.
- Insensitive to Delay- and jitter.

The requirements for **videoconferencing** can be applied as either a one-to-one capability or a multipoint conference.

- One-way latency from mouth to ear (ITU G.114 standard) ≤ 150 ms.
- Jitter should be ≤ 30 ms.
- Packets loss should be ≤ 1%.

QoS requirements for data:-

Since there are thousands of data applications, the end application must have its own requirements.

Traffic Classes and Bandwidth Allocation:-

Traffic classification is the process of identifying traffic and categorizing it into different classes to identify packets with their QoS requirements. A separate queue must be provided to each traffic class and it is impossible to mix them in one queue due to the varying traffic characteristics of different applications such as Voice, Video and Data traffic. Cisco recommends a phased approach to media application class expansion, as illustrated in Figure 1 [14]. In this paper, we adopt an 8-class model (see Figure 1). According to this eight-class model:

4-Class Model	8-Class Model	Weight(%)	12-Class Model
Realtime EF/CS4/CS5	Voice-EF	10	Voice-EF
	Interactive Video-CS4/CS5	23	Broadcast Video-CS5
			Real-time Interactive-CS4
Signaling / Control-CS2/CS3/CS6	Network Control and Management – CS2/CS6	5	Network Control-CS6
			Network Management –CS2
	Call Signaling-CS3	2	Call Signaling-CS3
Critical Data AF1/AF2/AF3/AF4	Streaming Video-AF3	10	Multimedia Streaming-AF3
	Critical Data-AF1/AF2/AF4	24	Multimedia Conferencing-AF4
			Transactional Data-AF2
			Bulk Data-AF1
Best Effort	Best Effort	25	Best Effort
Scavenger	Scavenger	1	Scavenger- CS1

Fig. 1:- WAN QoS Class Models.

1. A static bandwidth allocation is used.
2. The voice traffic is marked as EF (high priority), it guaranteed small delay and jitter. It is assigned a weight of 10%.
3. A policy is needed to regulate EF traffic to prevent the service starvation of other low priority service classes. If EF traffic exceeds certain rate limit, it will be dropped before accessing the network. Other traffic classes share the “rest” of bandwidth that is not used by EF class as shown in Figure 1.
4. The Interactive-Video traffic (traffic cless_6) includes both Broadcast Video (marked CS5) and Realtime Interactive (marked CS4) and is assigned a weight of 23%.

5. A minimal amount of bandwidth, such as 1%[15] is assigned whenever traffic cless_0, scavenger, is enabled. Figure 1 shows an example of how application class bandwidth allocations may be sub-divided into more granular QoS application-class models while retaining consistent bandwidth allocations.

A Proposed Reconfigurable QoS Model:-
In this section, first we will give the assumptions that are use through the design of the proposed RSM and then the details of the design are followed.

Model Assumptions:-
1. Three widely used schedulers (PQ, CBWFQ, and MDRR) are chosen to support realtime allocations.
2. The voice traffic (EF traffic) is handled by a strict priority queue scheduler.
3. It assumed that a policer is used to regulate EF traffic, the policer ensures that the traffic entering a queue does not exceed assigned limit.
4. The interactive video traffic (traffic class_6) is handled using a reconfigurable scheduler to give it preferential treatment.
5. The configuration is switching between the CBWFQ and MDRR schedulers based on the value of the queueing delay of the video traffic.
6. The two scheduling disciplines MDRR and CBWFQ are used alternatively to serve the other seven traffic classes (class_0 to 6).
7. The reconfiguration parameter (that is used to reconfigured the RSM, in runtime, either as a MDRR scheduler or as a CBWFQ scheduler) is the value of queueing delay of video traffic.
8. We implement the "single bottleneck" topology to compare the performance of different schedulers on a single core router. This topology is widely used in research about scheduling and queue management algorithms and differentiated service [16,17].
9. There are 8 source nodes generating traffic to the server. The link between the scheduler and server are the only bottleneck link. All Other links have enough bandwidth.
10. In our simulations all the packets have the same size (500 bits). Hence the bottleneck link has a speed of sending 2000 packet per second. Since the objective of the simulations is to compare the performances of different schedulers, the absolute values of link speed or packet size or traffic sources rates will not affect our results as long as they are set to the same for all schedulers[18].

Figure 2 shows the block diagram of the RSM which is used for the packet scheduling at the DiffServ routers.

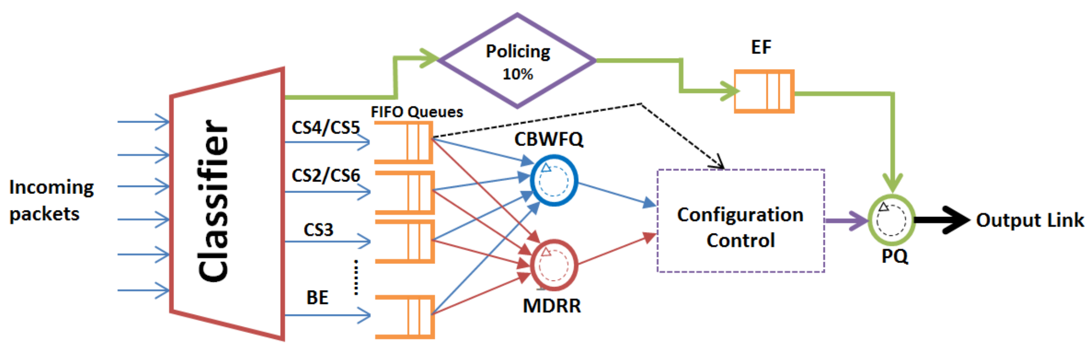

Fig. 2:- Reconfigurable QoS router model

Reconfiguration process:-
To reconfigure the RSM scheduler, a new procedure to estimate the value of video traffic queue delay is suggested. Figure 3 shows a flowchart of the proposed procedure.

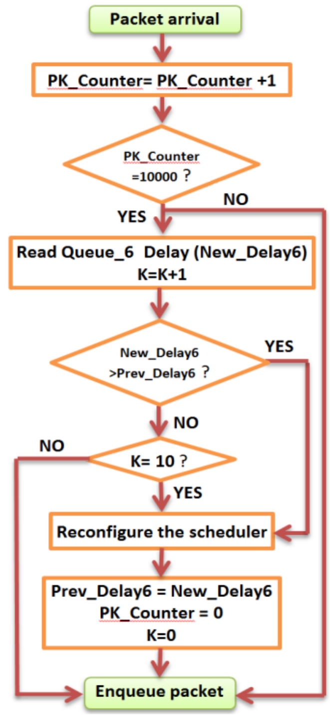

Fig. 3:- Flowchart of Scheduler Reconfiguration process.

Process model design:-

The state transition diagram of the RSM scheduler consists of eight states as shown in Figure 4:

- *init_1*, *init_2*, used to initialize the constants, variables and statistics and to read subqueues attributes.
- *enqueue*, and *dequeue*, used to enqueue and dequeue the packet to and from appropriate queue.
- *Sched_CBWFQ*, *Sched_MDRR*, *and Sched_PQ* implement the scheduling algorithms CBWFQ, MDRR and PQ respectively.
- and *idle state*. All states are forced states except the *idle* state which is an unforced state. In the simulation the delay, packet loss, and queueing size of each subqueue (using built-in statistic functions in OPNET) are measured.

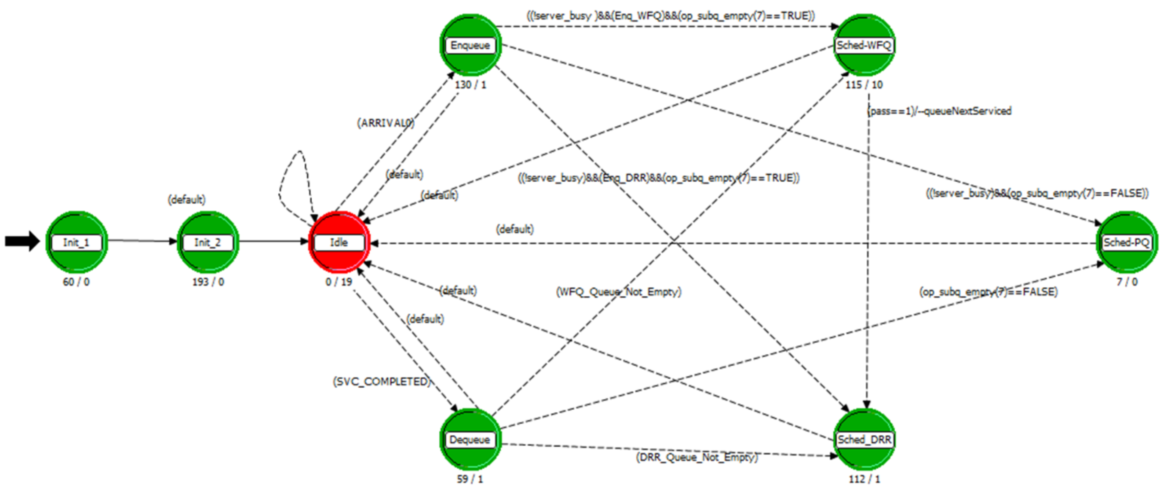

Fig. 4:- Finite State Machine of RSM process Mode

Simulations:-

OPNET software is used to simulate the RSM model based on the topology shown in Figure 5. There are 8 traffic classes each represented by four traffic sources generating traffic to a single server represented by the sink node that receives all packet sent from the traffic sources. A flow control mechanism at the sources is used to limit the amount of traffic sent by the traffic sources at different simulation times. The link between the scheduler and server is only the bottleneck link with service rate of 1M bit/sec. All Other links have enough bandwidth.

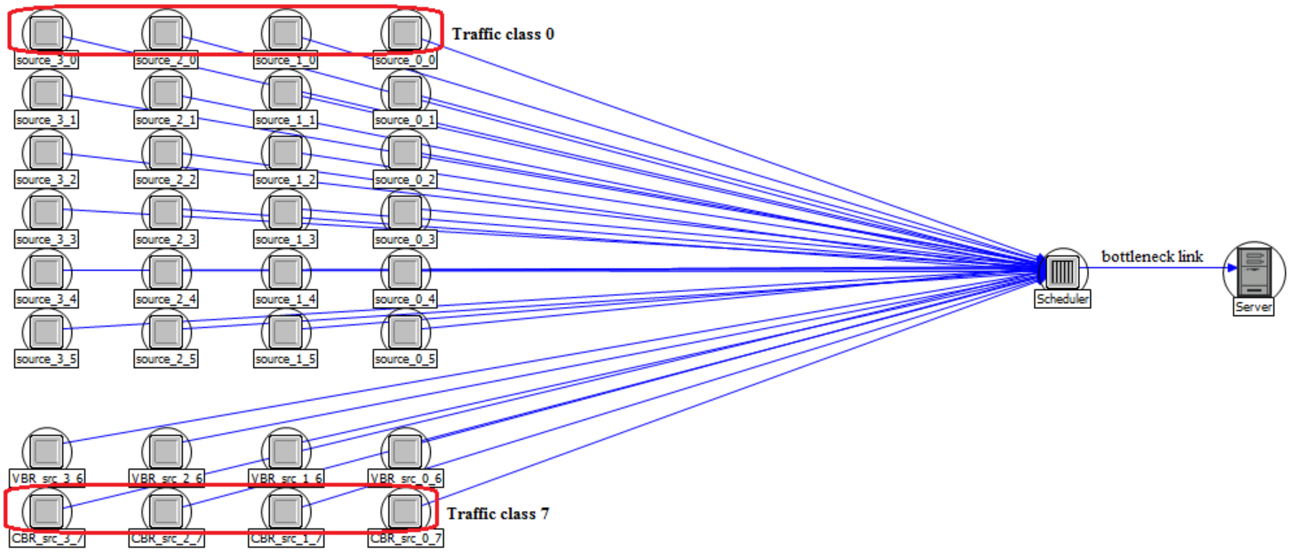

Fig. 5:- Network topology

The offered traffic load that goes to the bottleneck link of the network is equal to the sum of all aggregated traffic at the queues. The reconfiguration parameter is applied to video traffic class (Queue_6).

RSM scheduler model assumptions:-

The performance of RSM scheduler will be compared with the performance of Cisco MDRR and CBWFQ. The assumptions to be used in the scenarios during simulation of all schedulers are as follows:

1. The voice traffic is represented by four Constant Bit Rate (CBR) traffic sources with constant packet size of 500bit.
2. The video traffic is represented by four Variable Bit Rate (VBR) traffic sources. The video frame is segmented into equal size packets with a size of 500 bits.
3. Other traffic sources generate exponential distribution traffic for traffic classes 0-5.
4. The packets size of all other traffic classes are equal to 500 bit.
5. The bottleneck link has a speed of sending 2000 packet per second (1Mb/s). Each traffic class participates in portion of the load in proportional to the weight assign to it as shown in Table 1.

Table 1:- Participation of every traffic class in the total link bandwidth

	Class 0	Class 1	Class 2	Class 3	Class 4	Class 5	Class 6	Class 7
Class weight	1%	25%	24%	10%	2%	5%	23%	10%
Full load (Kbps) of the traffic class	10	250	240	100	20	50	230	100

6. The buffer space of the voice traffic (class 7) is set to 5 packets in order to maintain a small delay to EF packets.
7. In all other traffic classes, the buffer size is fixed to 500 packets in all scenarios.
8. The distribution of inter-arrival time in all traffic sources are exponentially distributed with different mean values.
9. All the queues, under the three schedulers, use Tail-drop queue management scheme.
10. The simulation time in every scenario is 30 minutes.
11. In order to study the performance of the proposed reconfigurable scheduler at different traffic conditions, the traffic sources are activated at different times (in four steps) under three scenarios as shown in Table 2.

Table 2:- Load assign to each traffic class in each step (as a percentage of the full load)

	Start time	Stop time	Scenario 1		Scenario 2			Scenario 3		
			Class 0-6	Class 7	Class 0-5	Class 6	Class 7	Class 0-5	Class 6	Class 7
Step 1	0.167min	30min	75%	25%	75%	100%	25%	100%	75%	50%
Step 2	4min	30min	100%	50%	100%	100%	50%	100%	100%	100%
Step 3	8min	30min	125%	75%	125%	100%	75%	100%	125%	100%
Step 4	20min	30min	150%	100%	150%	100%	100%	100%	150%	100%

12. As illustrated in Table 2 , the voice traffic (class 7) is policed to produce traffic that does not exceed the 10% of the output link speed in all steps of all scenarios.
13. The four steps of the traffic conditions (except class 6) shown in Table 2, are corresponding to 75%, 100%, 125%, and 150% of the full load of output link. The last two steps represent heavily loaded network (congestion condition).
14. As shown in Table 2, for voice traffic (class 7) in scenario 1 and 2, each source (from 0_7 to 3_7) generates traffic at rate of 25% of the full load (25 kbps), in different simulation time, which (approximately) emulates, for example, three voice connection using low bit-rate codecs such as G.729(8kbps). These three voice connection contributes a mean load of 50 packet/sec which, with 4 vice sources (12 voice connections), gives a total mean load of 10% of service capacity (2000 packet/sec). In scenario 3, source 1 and 2 (0_7 and 1_7) activated at the same time (in step 1) to generates packets at rate of 50% of the full load, which (approximately) emulates six voice connection using G.729 codec. The same things for source 3 and 4 (2_7 and 3_7) which activated at the same time (in step 2) to generates packets at rate of 50% of the full load. For the video traffic (class 6), in scenario 1 and 3, the traffic source (0_6) generate video traffic in step 1 at rate 75% of the full load (345kbps) which (approximately) emulates a video connection using MPEG4 compression standard (the bandwidth typically between 5Kbps and more than 1Gbps). Other three traffic sources from (1_6) to (3-6) generate video traffic in different steps each at rate 25% of the full load (115kbps).

Results:-

Figure 6 shows the queuing delay of video traffic as a function of the simulation time in minutes (m) under CBWFQ, MDRR, and RSM schedulers in the three scenarios. It shows that due to use the RSM scheduler, the queuing delay of the class_6 traffic under RSM follows the minimal paths as compared to the queuing delay under the other two schedulers.

Figure 7 shows the packets drop of video traffic (class_6) as a function of time under the three schedulers and scenarios. The packet drop of video traffic under the RSM reconfigurable scheduler shows an improvement for the three scenarios as compared to the other schedulers.

It should be notice that there is no packets drop of class 7 traffic (voice traffic), this is due to the policy that is applied to the voice traffic to prevent it from exceeding the assigned weight of the traffic. The average queuing sizes under the three schedulers and scenarios are shown in Figure 8. It is clear that RSM scheduler provides the minimum queuing size path. Also scenario_3 shows better results as compared to the other scenarios.

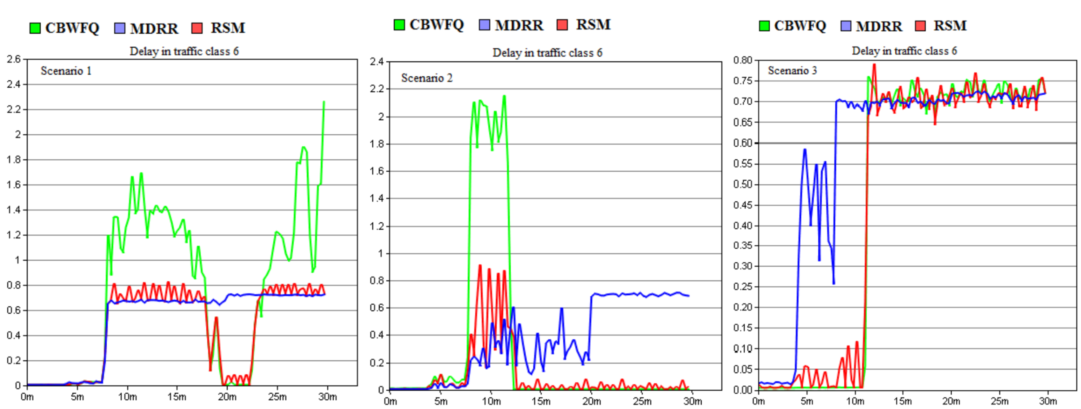

Fig. 6:- The queuing delay of class_6 traffic (video traffic) under CBWFQ, MDRR, and RSM schedulers for the scenarios 1, 2 and 3

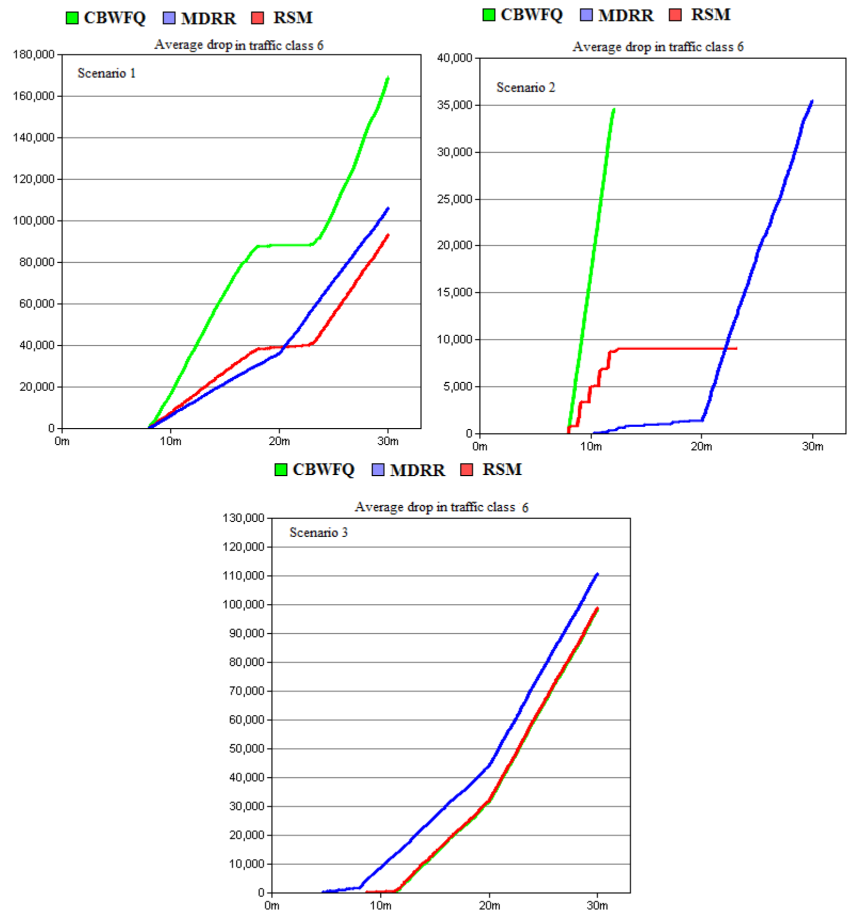

Fig. 7:- The packet drop of class_6 traffic (video traffic) under CBWFQ, MDRR, and RSM schedulers in scenario_1, 2, and 3.

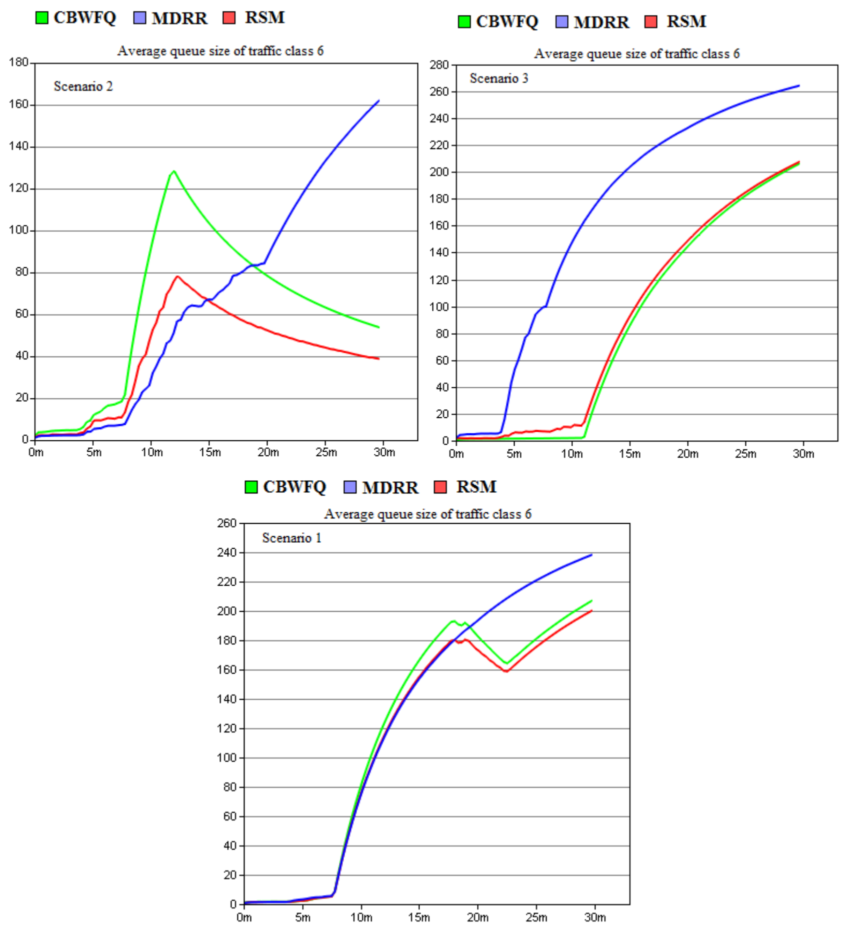

Fig. 8:- The queuing size of class_6 traffic (video traffic) under CBWFQ, MDRR, and RSM schedulers for the scenario_1, 2, and 3

The queueing delay of the voice traffic (class_7 traffic) is the same (almost zero) under the three schedulers and scenarios as shown in Figure 9. The packets drop in voice traffic is almost zero under the three schedulers and in the three scenarios. These results can be justified , since class_7 traffic uses strict priority queue that gives priority to the voice traffic over other traffic classes in addition to that the voice traffic is policed and does not exceed the 100% of the full load. It can be concluded from these results, that using the reconfigurable technique have no effect on the queuing delay or packet drop of the voice traffic. The dropped and sent packets are also similar under the three schedulers and not affected by the reconfiguration scheduler. The effect of using the reconfigurable scheduler to the average queueing delay of classes 0 to 5 is small. However, the other traffic classes are assumed delay insensitive.

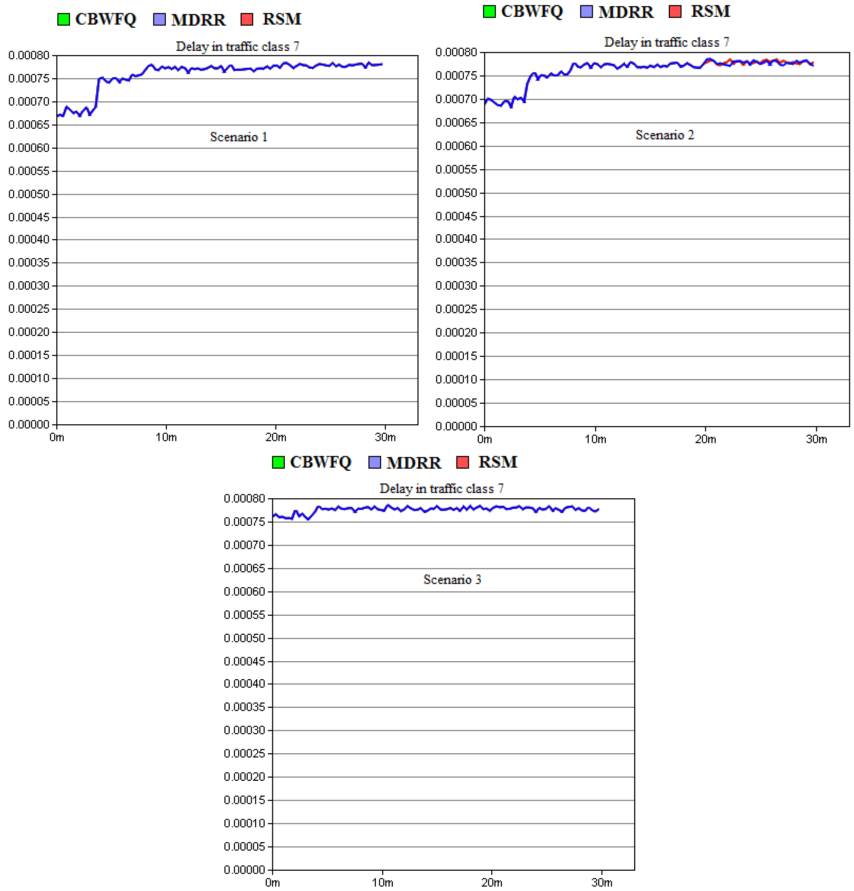

Fig. 9:- The queuing delay of class_7 traffic (voice traffic) under CBWFQ, MDRR, and RSM schedulers in scenario 1, 2 and 3

Also, the packets drop of classes 0-5 under the reconfigurable scheduler, stay within the range of packets drop under the other two schedulers and not exceed them. For example, Figure 10 and 11 , illustrate the queuing delay and packet drop, respectively, of traffic classes from 0 to 5 under the three schedulers in scenario 1. From the simulation result it is clear that with this reconfigurable QoS scheduling model, the packets loss and packets delay of real-time voice and video traffic are remaining at minimum value though the total offered load of the link is increasing beyond the available bottleneck bandwidth.

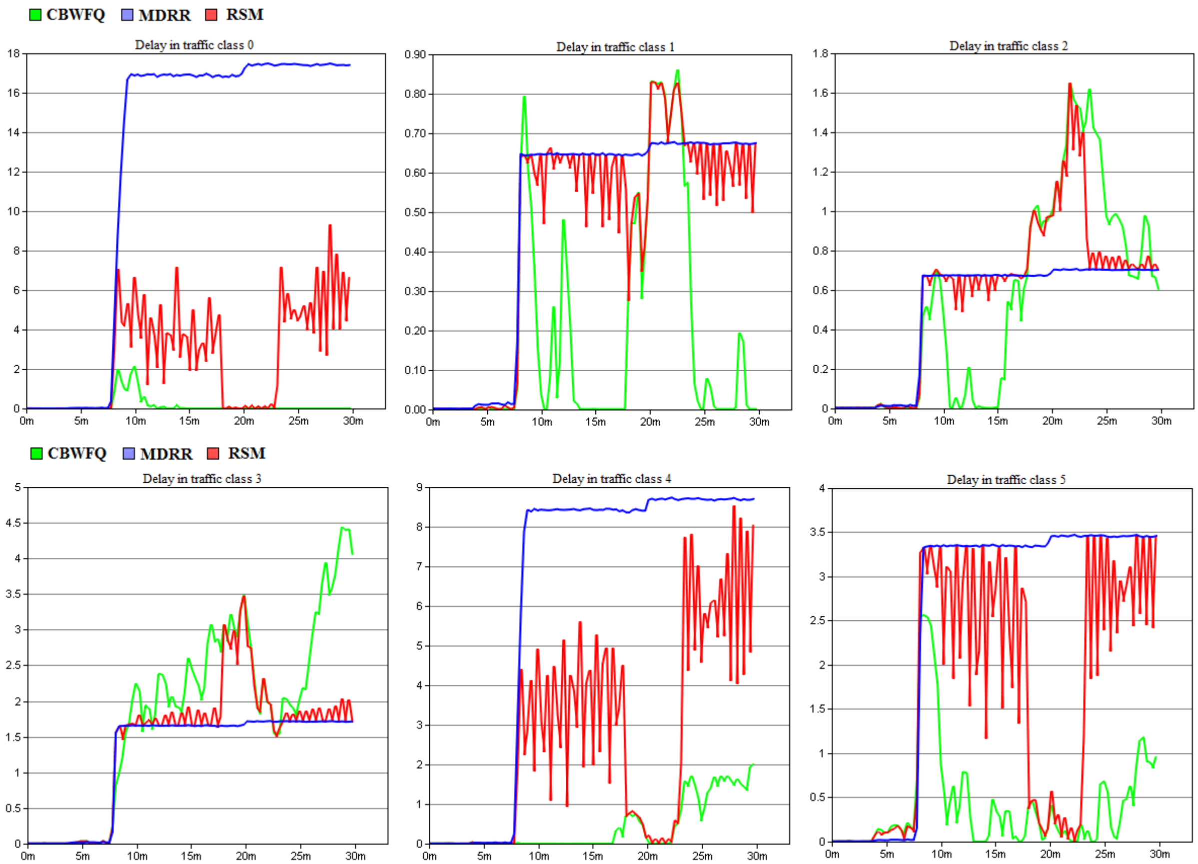

Fig. 10:- The queuing delay of classes (0-5) traffics under CBWFQ, MDRR, and RSM schedulers in scenario 1

Conclusion:-

Globally, real-time traffic (video and audio traffic) has dominated the internet data consumption for some years now[19]. However, how to improve the QoS of real time traffic is still remains an open issue, and this has motivated our research work.

In this paper, the proposed scheduler, that use the reconfigurable technique, improves the QoS (queue delay and packet loss) of real-time application in DiffServ networks. Eight queuing models classify the incoming traffic to eight classes, (each class is pleased in one queue). Queue_7 is considered as a high priority queue and assigned to the voice traffic(EF class). The real-time video traffic is handled using reconfigurable scheduler which switches their functionality between two well-known schedulers CBWFQ and MDRR. The simulation result show that by using reconfigurable QoS scheduling model, the packet loss and packet delay of video traffic are improved though the total offered load of the link is increased beyond the available bottleneck bandwidth. Also the simulation result show that these performance improvements by reconfigurable scheduler can work when the load is very high, thus providing a good solution to quality degradation of real-time traffic under congested network conditions. The simulation result also shows that in the three scenarios under RSM scheduler the queue delay and packet loss of voice traffic (queue 7) are almost zero due to the usage of priority queuing for voice traffic, and the queuing delay and packet loss of other traffic classes (classes 0-5) remain within the range of queuing delay of one of the other two schedulers (CBWFQ or MDRR).

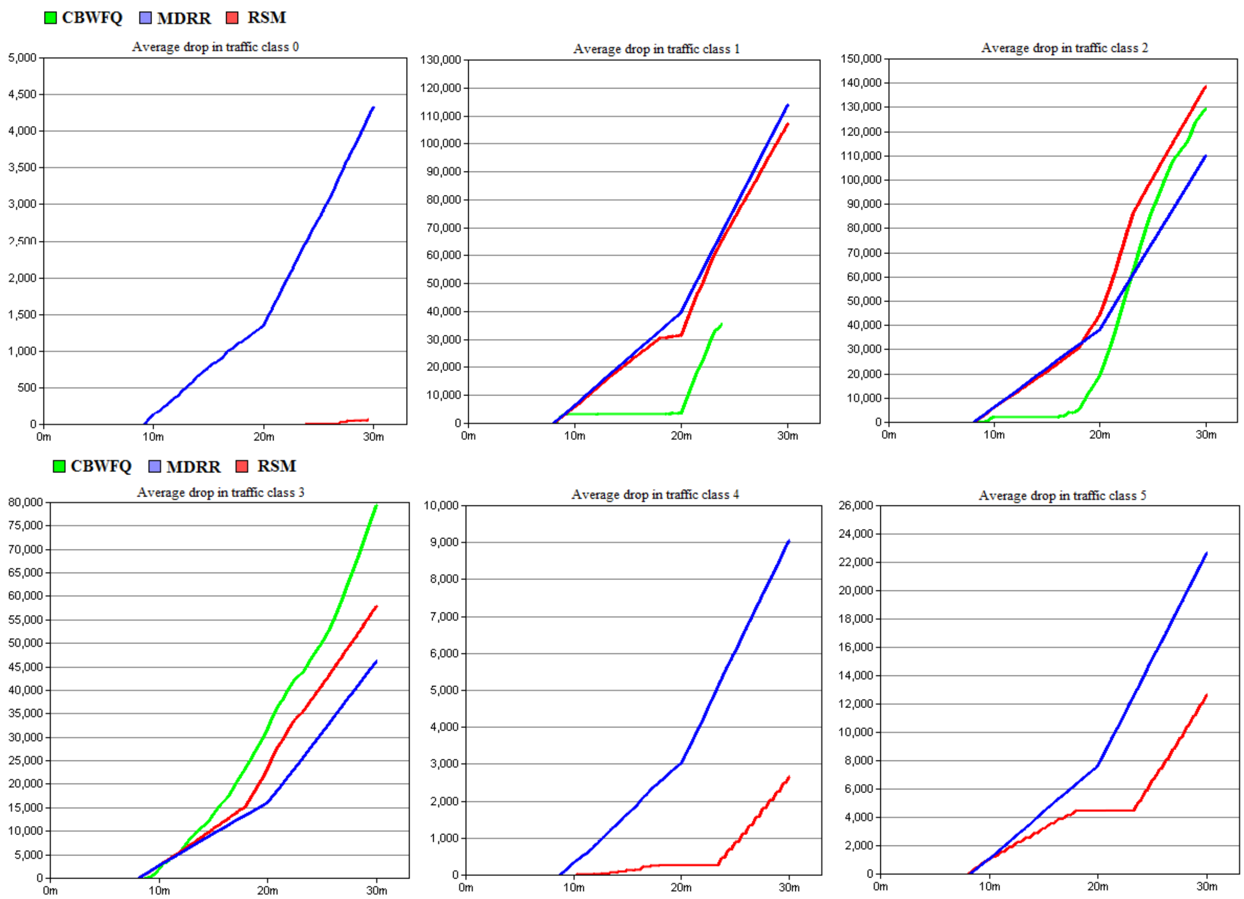

Fig. 11:- The packet drop of classes (0-5) traffics under the three schedulers in scenario 1

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