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Simevents/ Stateflow base Reconfigurable Scheduler in IP Internet Router

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Abstract: Quality of Service (QoS) is becoming increasingly important with the rise of multimedia applications (such as voice over IP (VoIP), video conferencing, and so on) in public data networks. These applications demand real-time service, while the networks should still be able to transfer also other traffic reliably (e.g., data, email, and WWW). In this paper, we concern how to improve the QoS of multimedia applications (Voice and video traffic) in DiffServ internet router by using the reconfigurable technique. Therefore we design an QoS model called Reconfigurable Scheduling Model (RSM) for real-time traffic in Diffserv router. In this model, we use the strict priority technique for the voice traffic to reduce the delay and then improve the QoS. While for video traffic, a hardware reconfigurable technique is used to provide the flexibility that improves QoS levels when compared to other techniques such as software-only implementations. This permits the run-time reconfiguration of the router resources, i.e., to change their functionality (from one scheduling algorithm to another), to adapt the fluctuation in traffic parameter.
We choose two of the most widely used schedulers (CBWFQ, and MDRR) to support real-time allocations and to improve the QoS of video traffic. The reconfiguration or the switching between those two schedulers is based on queuing delay of the video traffic. To evaluate the proposed approach, the queuing model was built and simulated with MATLAB Simevents and Stateflow components. The result of simulation shows an improvement in QoS of voice and video traffics.

Keywords: Reconfigurable scheduler, QoS, DiffServ

1. Introduction

Delivery of services of appropriate quality has always been one of the key goals of telecom operators and Service Providers[1]. Different classes of traffic (e.g. voice, video, data, etc.) have different bandwidth and delay requirements and then different QoS requirements[2]. In the internet network, two QoS approaches are used: Integrated Service (IntServ) and Differentiated Service (DiffServ). IntServ providing per-flow QoS guarantees to the individual application. However, it suffers from scalability problems. DiffServ is more scalable than IntServ and easy deployable for service differentiation in IP networks [3].

The basic principle of the DiffServ model is to separate the network traffic into several classes with the specific weight based on varying QoS requirements. Packets from applications with similar QoS requirements are assigned the same service class at the edge of the DiffServ network and aggregated in the core network. DiffServ model meets the QoS requirements of different service classes by providing Per Hop Behaviors (PHB).

Three types of PHBs or forwarding techniques have been defined in DiffServ network as follows[4]:
1. Expedited forwarding (EF), also known as *premium service*, it can be used to build assured end-to-end bandwidth with low loss, low jitter, and low latency.
2. Assured Forwarding (AF) assign a different level of drop precedence for each of the three levels in its four classes. It does not provide end-to-end service.
3. Best Effort (BE) provides the same service as that in the current Internet.

These three PHBs, EF, AF, and BE, are handled in a descending priority order. In this paper, we use the reconfigurable computing technique which is a rapidly emerging computing paradigm, not only in research but yet also in real applications due to its superior performance and lower power consumption compared to general purpose computing using microprocessors[5]. Reconfigurable router improves the throughput by changing their packet processing decision to reflect changes in the traffic flow rate.

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The traffic inside the proposed scheduler is divided, globally, into three types:

1. Voice traffic which needs to be kept separate because it is especially sensitive to delay.
2. Video traffic is also delay-sensitive and is often so bandwidth-intensive that care needs to be taken to make sure that it doesn't overwhelm low-bandwidth WAN links.
3. Other traffics are identified and marked in a reliable way to make sure that it is given the correct classification and QoS treatment within the network.

The voice traffic is put into the highest priority queue and has strict priority over the other two types in a non-preemptive way. The video traffic is handled by the two of the most widely accepted queue schedulers in DiffServ routers; Class-Based Weight Fair Queuing (CBWFQ) and Modified Weight Round Robin (MWRR). These two schedulers are used in a reconfigurable way by switching between them based on lower video traffic queue delay. Other traffics are scheduled by the active scheduler (CBWFQ or MDRR) which determined by video traffic queue delay.

This paper is organized as follows: Section 2 gives an overview of the previous related works. A description of the proposed RSM model is provided in section 3. Section 4 reviews the simulation results using MATLAB. Finally, the conclusions are drawn in section 5.

2. Literature Review

Several studies develop different methods and techniques for improving the QoS of the multimedia service in a variety of communication media and devices.

Prabesh et al. [6] provide a theoretical model for minimum buffer size as a means of achieving the desired QoS for video streaming application. The Research provides a general optimal video smoothing algorithm based on the concept of dynamically controlled Coefficient of Variance (CV), which is the ratio of standard deviation of the end-to-end delay and the expected value of the delay for each ensemble of packets being transmitted through the network. The paper discusses how the size of the "receive buffer" is affected by the allocated bandwidth for each source-pair end users for supporting video streaming applications without any gaps.

A scheme to improves the QoS of multimedia transmissions is proposed by Kim et al.[7] using an adaptive algorithm that switches between UDP (User Datagram Protocol) and TCP (Transmission Control Protocol) based on the size of the data. Also, an important multimedia data segments are selected for retransmission using the Forward Error Correction (FEC) methodology to prevent the loss of packets during the transfer and to improve transmission efficiency.

In [8] an adaptable network architecture, called ADNET is proposed. The architecture provides mechanisms to allow the application adapt to the resource constraints to achieve improved QoS. The design aims to unify different QoS control mechanisms (e.g. integrated services, differentiated services, and active networks) together to provide a wide range of network services to all users to meet their specific needs. Also, the author proposes a fragmentation scheme with low overhead to transfer large-size multimedia data and use this fragmentation scheme to integrate a new transport protocol, called ACtive Transport Protocol (ACTP) is integrated with the design.

In order to improve the quality of service of multimedia communication and experience, the author of the article [9] propose a crowd service cooperation control protocol based on the opportunistic wavelet model. The wavelet multimedia mobile analysis model is founded after the combination of the video frame resolution analysis and the wavelet scale space. T. Pliakas [10] reviews solutions which address the problem of end-to-end QoS for multimedia streaming applications over heterogeneous networks, including wireless and wired network domains. He combines the QoS multimedia streaming techniques and coding methods and the work on QoS support in the network layer, considering wired and wireless networks, and then he proposes techniques for mapping application QoS related semantics with the appropriate network low-level description themes in the context of state-of-the-art proposed QoS architectural frameworks.

To reflect the dynamic characteristics of the QoS and network performance , a QoS evaluation system has been proposed by Al-Sbou[11] based on using the nonlinear autoregressive with exogenous input model (NARX). His proposed approach was developed in three phases: firstly, data collection and pre-processing; secondly, NARX modeling; and finally, the analysis and validation of the proposed model performance in comparison with other prediction models. The QoS parameters(delay, jitter, and losses), and previous QoS values were used as inputs to the developed NARX model to the estimated QoS.

The End-to-End QoS control for multimedia content delivery over heterogeneous networks has been investigated by B. *Shao et al.* [12] from multimedia terminal perspective. they propose a new end user terminal architecture including MPEG-21 Digital Item Browser, usage environment characteristics provider, end user's QoS monitor and Scalable Video Coding (SVC) audio-visual player coordinated under a terminal middleware. Several adaptation methods have been proposed and the system architecture also has been optimized. Such a terminal facilitates content adaptation solution to maximize user's fruitful experiences of video quality. The architecture design also exploits and supports the MPEG-21 and H.264 extension SVC codec standards.

In [13], a genetic algorithm (GA) and neural networks (RNN) are applied to optimize the QoS of multimedia traffic over LTE network. This work introduces an integration framework between genetic algorithm (GA) and random neural networks (RNN) applied to QoE-aware optimization of video stream downlink scheduling. To meet varying Quality of Service (QoS) needs of applications, [14] proposes an SDN (Software Defined Network)-based Application-Driven Network (ADN) that adopts customized data paths on an application basis. This allows using network resources in a more efficient way and providing tailored network services to applications.

3. Scheduling Algorithms

Packet scheduling mechanism is one of the important items for QoS control. Packet scheduling specifies the service policy of a queue within a node (e.g. an IP router, an ATM switch)[15]. In practice, scheduling decides the order that is used to pick the packets out of the queue and to transmit them over the channel. Each queue scheduling mechanism attempts to find the right balance between control, complexity, and fairness. Also, different algorithms aim to optimize different aspects of packet QoS and network behavior, such as delay, delay jitter, throughput, prioritization, and fairness. The simplest algorithm is FIFO, where packets are temporarily stored in the form of first come first serve (FCFS) order until network resources become available[16]. In the proposed system, we chose two of the most widely used packet scheduling algorithms in Diffserv router: *Modified deficit round robin* (MDRR) and *Class-based weight fair queuing (CBWFQ)*.

These algorithms are preferred since they exhibit the following desirable characteristics[17]:
(a) ease of implementation and verification.
(b) good exploitation of the available network bandwidth.
(c) limited processing power and memory requirements.
(d) network administrators find them easy to understand and configure.

A. Modified Deficit Round Robin (MDRR)

With MDRR, non-empty queues are served one after the other, in a round-robin fashion. Each time a queue is served, a fixed amount of data is dequeued. The algorithm then services the next queue. Every queue within MDRR is defined by two variables: ***byte count*** and ***quantum.*** In each new round, the ***byte count***, which specifies the number of bytes to be transmitted from the queue upon its turn, is increased by its ***quantum*** value. If the length of the packet at the top of the queue is less than byte_count, the packet scheduler can dequeue the packet and then subtracts the packet length from byte_count. Otherwise, it accesses the next queue[18].

B. Class-Based Weight Fair Queuing (CBWFQ)

Class-based weighted fair queueing (CBWFQ) extends the standard WFQ functionality to provide support for user-defined traffic classes. It is widely used in DiffServ routers[19]. CBWFQ queuing algorithm has the ability to guarantee bandwidth and dynamically ensure fairness to other flows within a class of traffic. CBWFQ allows you to specify the exact amount of bandwidth to be allocated for a specific class of traffic. This will ensure that certain traffic such as voice will not suffer latency. The bandwidth assigned to a class is the minimum bandwidth delivered to the class during congestion. Also, to characterize a class, you specify the queue limit for that class, which is the maximum number of packets allowed to accumulate in its queue. Packets belonging to a class are subject to the bandwidth and queue limits that characterize the class.

4. RSM Model

As mentioned previously, The RSM scheduler has been implemented using three queue schedulers in DiffServ routers; Priority Queuing (PQ), Class-Based Weight Fair Queuing (CBWFQ) and Modified Deficit Round Robin (MDRR). These queuing methods offer differentiated service to network traffic flows, optimizing performance based on administrative configurations. The voice traffic (*EF traffic)* is put into the highest priority queue and has strict priority over the other classes. To improve the QoS of the video traffic, we use the reconfigurable technique by switching to one of the two schedulers (MDRR or CBWFQ) that shows a smallest queuing delay of the video traffic .

Fig. 1 shows a block diagram of the RSM scheduler[20]. Packets are first divided into classes by marking the Type of Service (ToS) byte in the IP header. A 6-bit bit-pattern (called the Differentiated Services Code Point [DSCP]) in the IPv4 ToS Octet or the IPv6 Traffic Class Octet is used for this purpose. Although RFC 4594 outlines a 12-class model, Cisco estimates that not all users are ready to implement a complex QoS design. Cisco recommends a phased approach to media application class expansion, as illustrated in Table 1 [21].

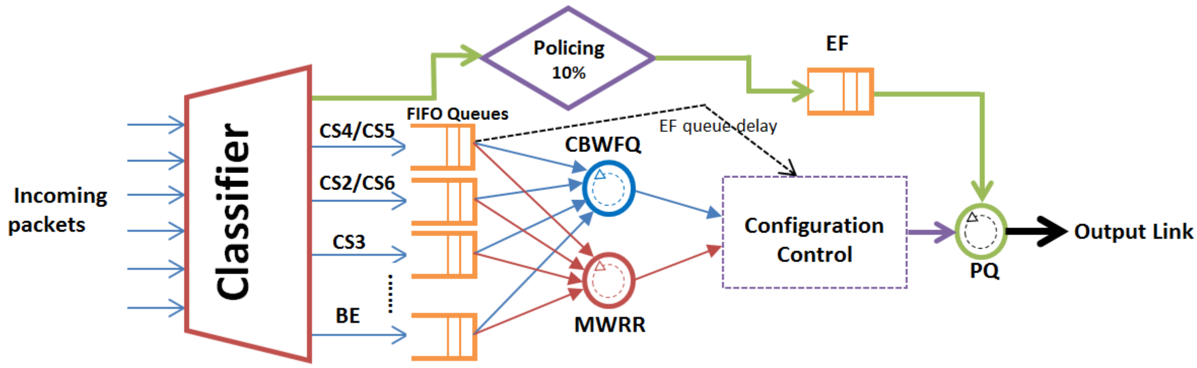


Figure 1. Reconfigurable QoS router model

This 8-class model was adopted in this design. The traffic is classified into eight classes, each class put in a single queue with a specified weight according to Table 1. The scheduler uses a priority queue (PQ) to serve the voice traffic queue as long as this queue is not empty (premium service), otherwise, the traffic classes(0-6) are served. However, a policer (at the ingress router of the DiffServ domain), is needed to regulate the voice traffic to ensure that the traffic entering a queue does not exceed a certain limit and to prevent the service starvation of other low priority service classes. If the voice traffic exceeds the allowable limit, it will be dropped before accessing the network. To serve the traffic classes(0-6), the queue delay of traffic class-6 (video traffic) is used by the configuration control circuit of the RSM scheduler to configure the scheduler during the run-time to work as CBWFQ or MDRR.

5. Reconfiguration Process

To reconfigure the RSM scheduler, we suggest a procedure to test, periodically, the value of video traffic queue delay under CBWFQ and MDRR schedulers. Based on these tested values, the configuration control circuit of the RSM scheduler switching to the scheduler that shows the smallest queue delay of video traffic. Fig. 2 shows a flowchart of the proposed procedure.

6. Matlab RSM Model Components Design

MATLAB Simevents/Stateflow tool 2014a was used to develop the RSM model. Simulink is a simulation and model-based design environment for dynamic and embedded systems, integrated with the MATLAB. It is basically a graphical block diagramming tool with a customizable set of block libraries. There are several other add-on products provided by MathWorks and third-party hardware and software products that are available for use with Simulink. One of them is the **Stateflow** which allows developing state machines and flow charts [22]. Fig. 3 illustrate a MATLAB block diagram of the proposed RSM model. It consists of the following main subsystems:
8-class packet generator, Enqueue unit, Dequeue unit, Sink unit, Reconfigurable scheduler and Scheduler control. In the following are brief descriptions for these subsystems.

TABLE 1. WAN QoS CLASS MODELS

Weight(%) / Class	4-Class Model	8-Class Model	12-Class Model
10 (class-7)	Realtime EF/CS4/CS5	Voice-EF	Voice-EF
23 (class-6)		Interactive Video-CS4/CS5	Broadcast Video-CS5
			Real-time Interactive-CS4
5 (class-5)	Signaling / Control-CS2/CS3/CS6	Network Control and Management –CS2/CS6	Network Control-CS6
			Network Management –CS2
2 (class-4)		Call Signaling-CS3	Call Signaling-CS3
10 (class-3)	Critical Data AF1/AF2/AF3/AF4	Streaming Video-AF3	Multimedia Streaming-AF3
24 (class-2)		Critical Data-AF1/AF2/AF4	Multimedia Conferencing-AF4
			Transactional Data-AF2
			Bulk Data-AF1
25 (class-1)	Best Effort	Best Effort	Best Effort
1 (class-0)	Scavenger	Scavenger	Scavenger

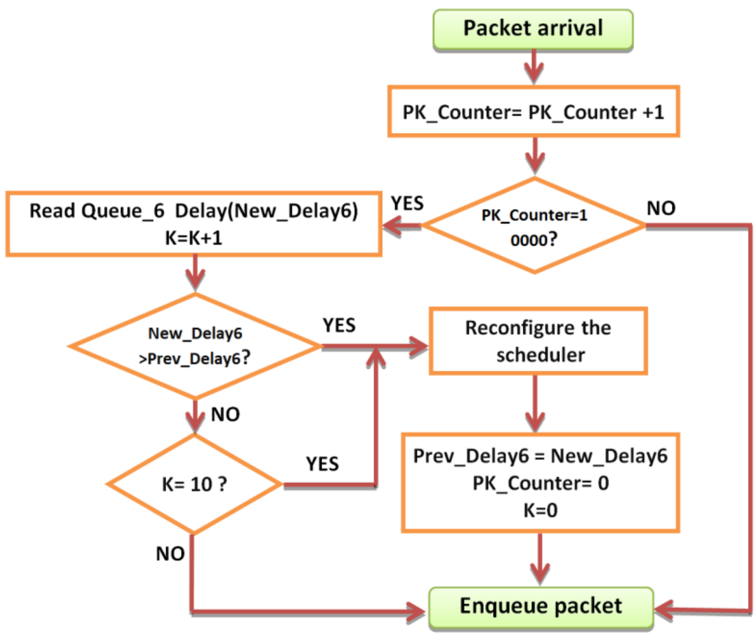


Figure 2. Flowchart of Scheduler Reconfiguration process

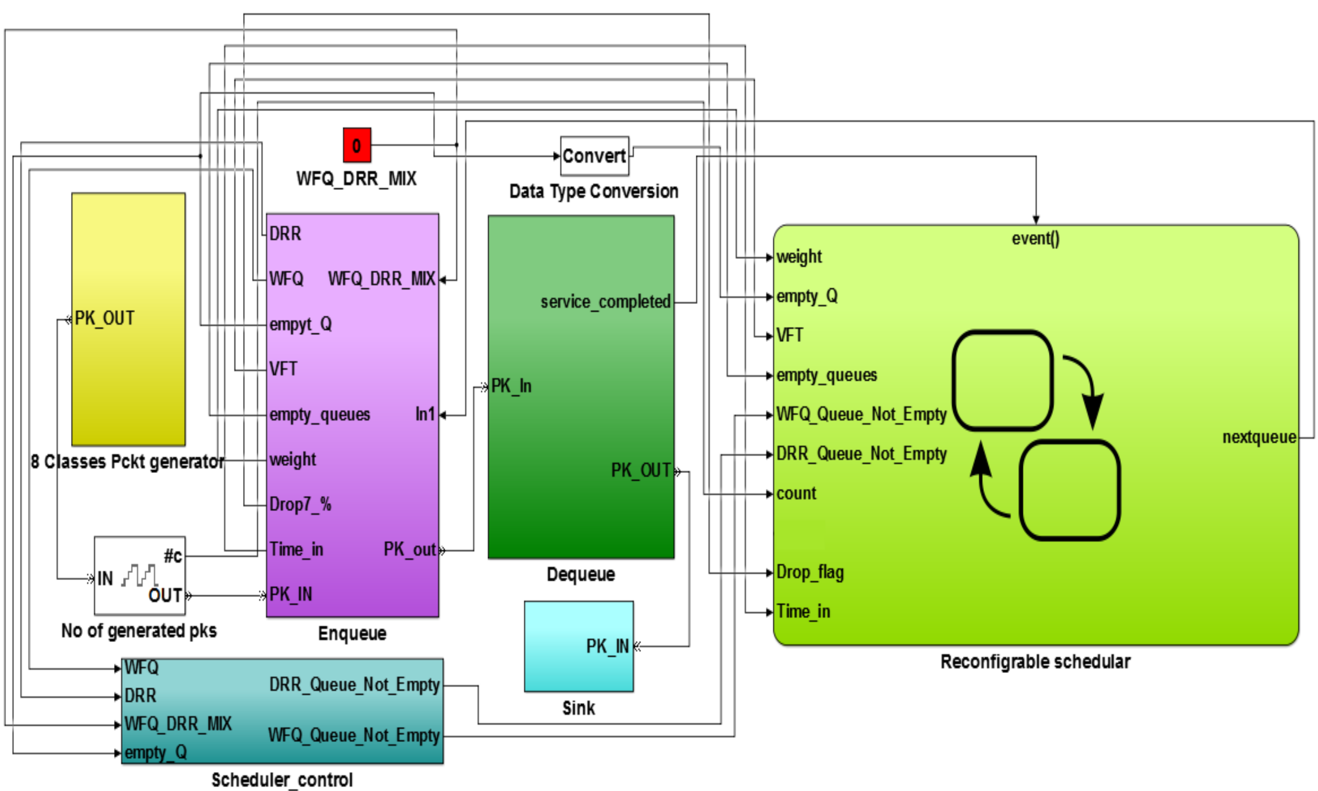


Figure 3. Block diagram of the RSM model using Simevents/Stateflow tools

A. 8-class packet generator

Fig. 4 shows the 8-class packet generator. This unit generates eight traffic classes. It consists of 32 traffic generators divided into eight groups. Each group responsible for generating one traffic class. Each of the four traffic generators of every group starts and stop at a specific time, as will be shown later. Finally, each traffic generator (class 0-5) has the following parameters:

1- Distribution: Exponential with initial seed equal to 12345, and mean equal to the inverse of (service_rate*weight/2).

2- Packet size : 500 bytes.

B. Enqueue unit

After a packet is classified to one of the eight classes, the router will enqueue the packet in the corresponding queue if the queue is not full (tail-drop within each class). Fig. 5 shows the block diagram to the enqueue unit. It consists of the following subsystems:

1) VFT Calculation subsystem: Upon arrival of every new packet, the VFT Calculation subsystem (a MATLAB function) calculate the Virtual Finish Time (VFT) for the packet (i.e. estimates the time of finishing serving the packet) according to Eq.(1)[23]. This finish time then will be used to determine which packets to serve first. The calculated VFT time is tagged with the packet by the Set Attribute subunit.

finish_time=arrival_time+pk_total_size/service_rate (1)

2) Switching Function subsystem: This subsystem (a MATLAB function) read the delay value of video traffic queue and compare it with the previous value to determine if the scheduler should be switched to the alternate scheduler. This unit implements the flowchart of Fig. 2.

3) 8-FIFO Queues subsystem: It consists of eight 500 packet size FIFO the queues, one for each traffic class. The router will enqueue the packet at the tail of queue if the queue limit has not been reached, otherwise the packet is dropped.

4) Output Switch and Replicators subsystem: This unit contains one output switch to route the incoming packets, according to its class number, to eight replicators. The replicators produce two paths for every arrived packet. One path to the PK_Drop subsystem, while another path to subsystem 3 to enqueue the packet at the tail of the queue.

5) PK_Drop subsystem: The PK_Drop subsystem checks the queue capacity after each arrived packed if the queue capacity is full, the packet is dropped. Otherwise, it produces enable signal to subsystem 3 to enqueue the packet at the tail of the queue.

6) Subsystems 1, 2 and 3: Subsystem 1 is a collection of logic gates blocks used to check the capacity of the FIFO queues to produces ***empty_queues*** (all queues are empties) and ***empity_Q*** (eight empty queues signals to reflect the status of each queue). Subsystem 2 contains ***Get_attribute*** units to extract the attribute (as ***weight, Time_in,*** and ***VFT***) from the packets. Finally, subsystem 3, has eight enable gate blocks controlled by eight enable signals, coming from PK_Drop subsystem. These gates enable or prevent the packets to go to the 8-FIFO Queues subsystem.

7) Stateflow chart1 and chart2: These charts are used to set the limit of the values that control the path combiner and output switch in Output Switch and Replicators subsystem.

C. Dequeue unit

As shown in Fig. 6, the dequeue unit receive the removed packet from the subqueue, in the enqueue unit, and pass it to the single server. The service time of the single server is 1/2000 sec. After completion serving the packet, the entity departure function-call generator generates a service_completed signal which is used as a trigger signal to the reconfigurable scheduler to start to produce the number of the next queue (nextqueue signal) to be served.

D. Sink unit

After the packet has been serviced, it goes to sink unit and update the statistics.

E. Reconfigurable scheduler

The reconfigurable scheduler unit implemented using Stateflow and MATLAB functions as shown in Fig. 7. Five states (*idle, ini, Sched_PQ, Sched_WFQ,* and *Sched_DRR*) and two MATLAB functions (*drr* and *wfq*) are used in the implementation. Following is a brief description of these states and functions:

- ***Idle****:* At this state, start and end all the transitions to other states. Also, it initializes some variables.
- ***ini****:* Initialize the variables.
- ***Sched_PQ****:* The transition to this state occurs if queue 8 (voice traffic queue) is not empty. Implement strict priority queue.
- ***Sched_CBWFQ:*** The transition to this state call *wfq* MATLAB function to implement CBWFQ algorithm, and this occurs if queue 8 (voice traffic queue) is empty and any of other queues (except queue 8) is not empty.
- ***Sched_DRR:*** Same as *Sched_WFQ* but it calls *drr* MATLAB function to implement MDRR algorithm.

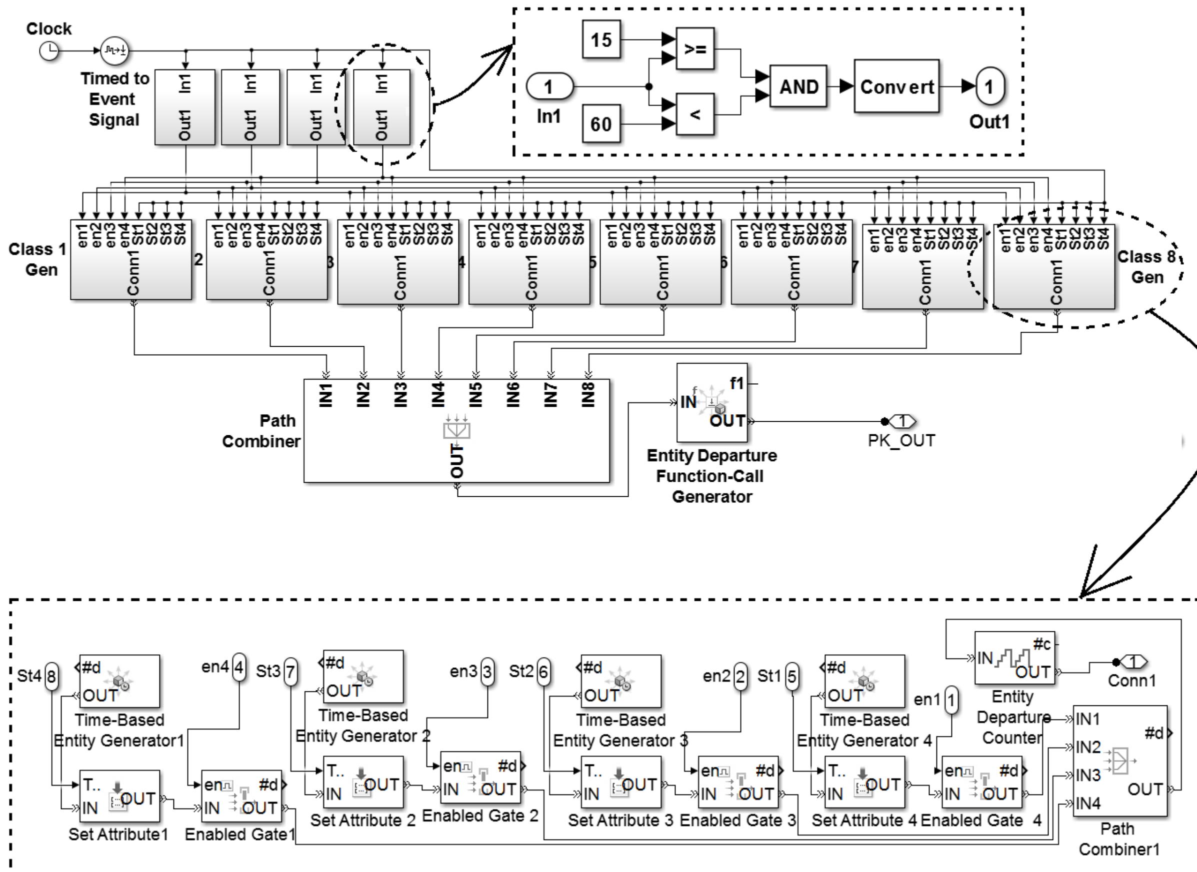


Figure 4. Block diagram of 8-class traffic generator

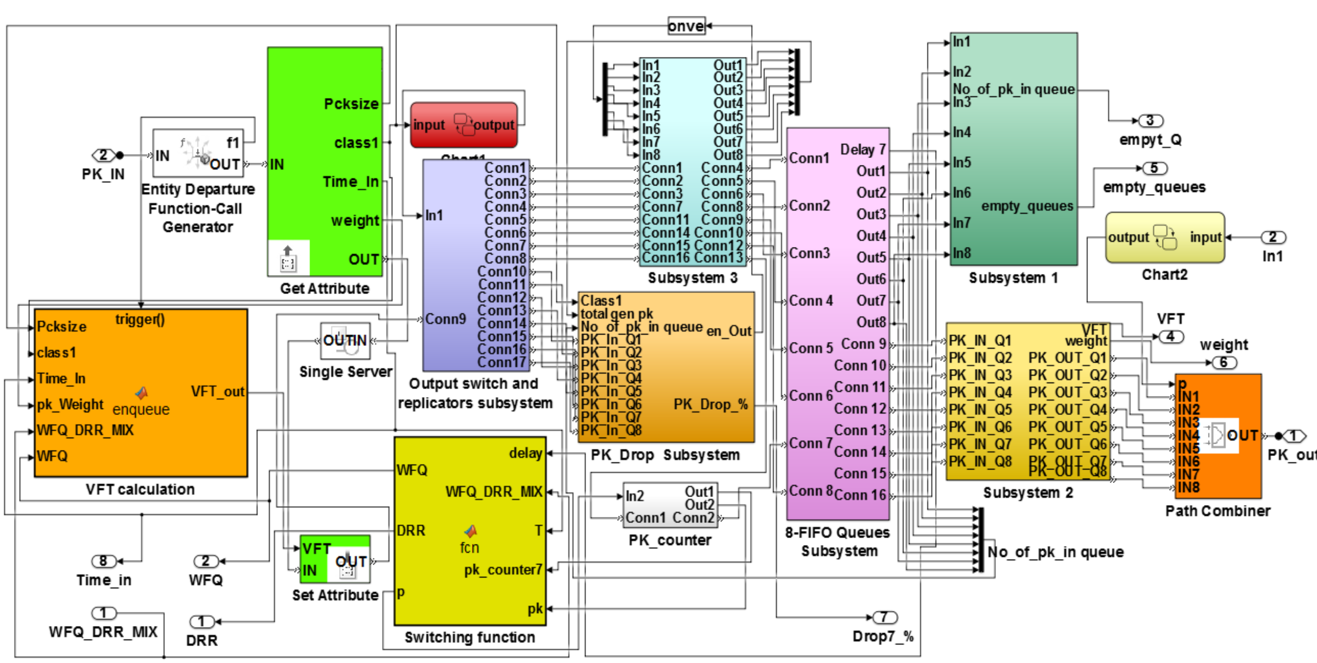


Figure 5. Simevents/Stateflow block diagram of the Enqueue subsystem

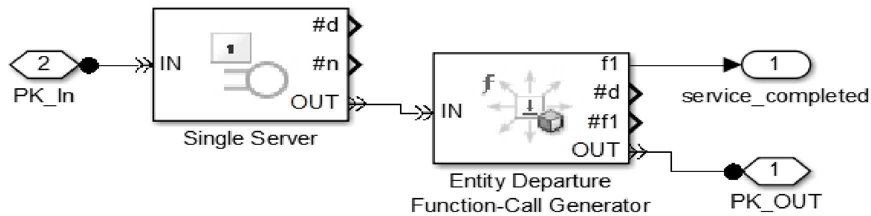


Figure 6. Block diagram of the Dequeue unit

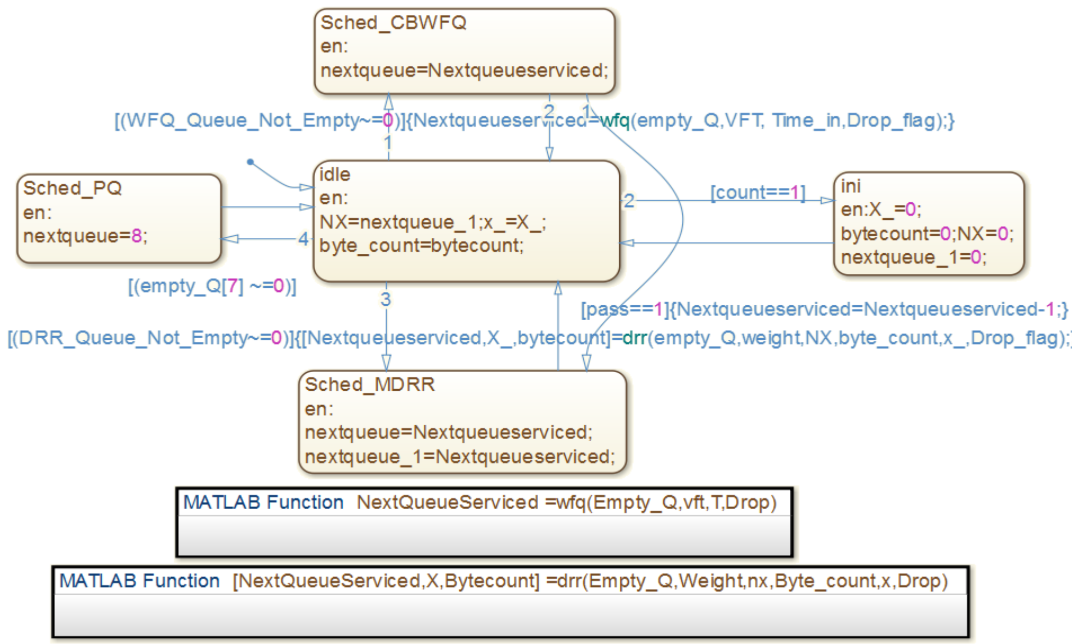


Figure 7. Stateflow diagram of the reconfigurable scheduler

F. Scheduler control

It is a collection of logic gates blocks that produce a signals (***WFQ_Queue_Not_Empty*** and ***DRR_Queue_Not_Empty***) applied to reconfigurable scheduler unit to enable the transition to the ***Sched_CBWFQ*** or ***Sched_MDRR*** state according to the condition described in the previous paragraph.

7. SIMULATION RESULT

In the following section, QoS under RSM scheduler is evaluated by simulation and compared to QoS under MDRR and CBWFQ schedulers. We implement the "single bottleneck" topology to compare the performance of different schedulers on a single core router as shown in Fig. 8.

A. RSM scheduler model assumptions

Two simulation scenarios are implemented. The assumptions to be used in the two scenarios during simulation of all schedulers are as follows:

1) The packets size of all traffic classes is equal to 500 bit.
2) The bottleneck link has a speed of sending 2000 packet per second (1Mb/s).
3) The buffer space of all traffic classes is fixed to 500 packets in all scenarios.

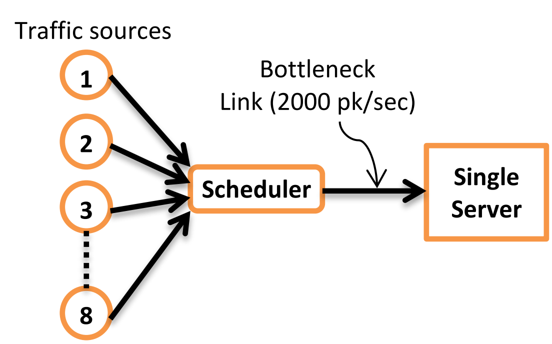


Figure 8. Single bottleneck topology

4) The voice and video traffic are represented by a constant bit rate traffic sources.
5) The distribution of interarrival time of traffic sources (0-5) is exponentially distributed with different mean values.
6) All the queues, under the three schedulers, use Tail-drop queue management scheme.
7) The simulation time in every scenario is 120 seconds.
8) 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) in the two scenarios. Table 2 shows the start and stop times of the traffic sources of classes 0-7 in the simulated scenarios. As mentioned earlier, each traffic class consists of four traffic sources. Both implemented scenarios use the same timing table.

TABLE 2. PARAMETERS SETTING FOR TRAFFIC CLASSES

Scenario No.	**1, 2**			
Traffic Class	**0 - 7**			
Step No.	1	2	3	4
Active Traffic source	1	2	3	4
Start Time(sec)	2	10	15	35
End Time(sec)	120	120	120	120
Traffic percentage (of the full load)	50%	100%	150%	200%

9) In each step, the traffic of classes (0 to 7) is increased by 50% of the full load according to the weight assigned to that class.
10) Traffic class 7 (voice traffic) is policed to produce traffic that does not exceed the 10% of the output link speed in all steps of the two scenarios
11) The four steps of the traffic conditions of the classes in the two scenarios are: step 1 is under provisioning, step 2 is provisioning and step 3 and 4 are overprovisioning. These steps corresponding to 50%, 100%, 150%, and 200% of a full load of output link respectively(see Table 3).

B. Results discussion

We evaluate the performance of proposed system implementations in terms of average queue delay, average queue size, and the packet loss. In both implemented scenarios, the traffic conditions are the same. However, to show the reconfiguration process in RSM scheduler; we give traffic class 6 (video traffic) in CBWFQ scheduler in scenario 2, a priority for a small period at the beginning of simulation time. By this method, we produce a cross section between the video traffic delay curves of CBWFQ and MDRR schedulers.

Fig. 9 shows the queuing delay of video traffic as a function of the sink traffic under CBWFQ, MDRR, and RSM schedulers in the two scenarios. It shows that due to the reconfiguration, the queuing delay of the class_6 traffic (video traffic) under RSM follows the minimal paths as compared to the queuing delay under the other two schedulers. Fig. 10 shows the packets drop of video traffic (class_6) as a function of sink traffic under the three schedulers and in the two scenarios. The packet drop of video traffic under the reconfigurable scheduler (RSM) follows the minimum path or near the minimum path of the two schedulers in both scenarios.

The average queuing sizes under the three schedulers, in the two scenarios, are shown in Fig. 11. It is clear that RSM scheduler provides the minimum queuing size path. The queueing delay and packets drop of the voice traffic (class_7 traffic) is the same (almost zero) under the three schedulers and two 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 10% of the full load.

It can be concluded from these results, that using the reconfigurable scheduler have no effect on the queuing delay or packet drop of the voice traffic. The effect of using the reconfigurable scheduler to the average queueing delay of other traffic classes (0 to 5) is also showing some improvement. In sometimes, it follows the minimum path of queue delay or at least (in other times) do not exceed the delay path of one of the two schedulers (CBWFQ or MDRR). However, the other traffic classes are assumed delay insensitive. Also, the packets drop of classes 0-5 under the reconfigurable scheduler (RSM), stay within the range of packets drop under the other two schedulers and not exceed them. For example, Fig. 12 illustrates the average queue delay in traffic classes from 0 to 5 under the three schedulers and in scenario 1. From the simulation result it is clear that with this reconfigurable scheduling model, the packets loss and packets delay of real-time voice and video traffic are remaining at a minimum value though the total offered load of the link is increasing beyond the available bottleneck bandwidth.

8. CONCLUSION

Multimedia traffic over Internet network is one of the highest percentages of traffic and it has been growing rapidly in recent years[24]. However, the effectiveness of many today's Multimedia applications, such as Voice over IP (VOIP), videoconferencing, Video on Demand (VOD), and others similar rely heavily on the quality of the network communication systems[25].

TABLE 3. TRAFFIC CONDITIONS(PK/SEC) FOR CLASSES 0-7 UNDER THE TWO SCENARIOS

Traffic condition (in packets)	Step No.	Class 0 1%	Class 1 25%	Class 2 24%	Class 3 10%	Class 4 2%	Class 5 5%	Class 6 23%	Class 7 10%
Under-Provisioning	1	10	250	240	100	20	50	230	100
provisioning	2	20	500	480	200	40	100	460	200
Over-provisioning	3	30	750	720	300	60	150	690	300
	4	40	1000	960	400	80	200	920	400

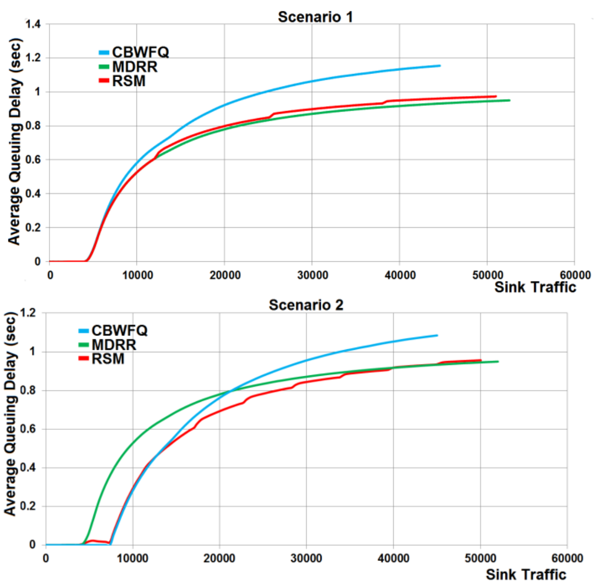


Figure 9: Video traffic delay under three schedulers in scenarios 1 and 2

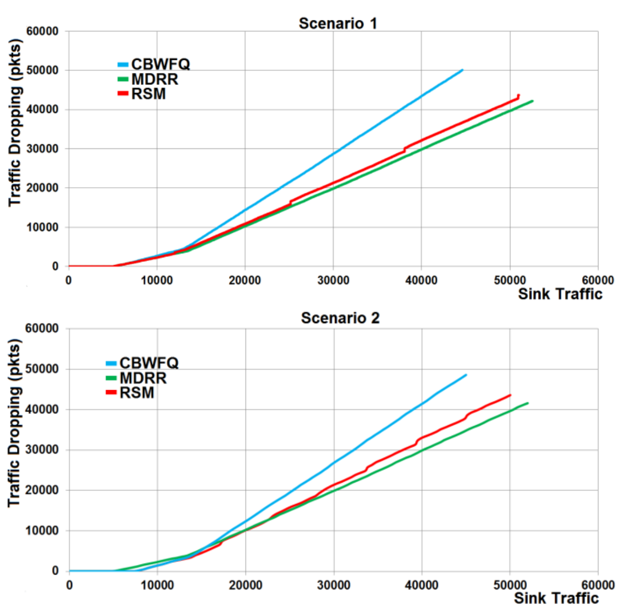


Figure 10. Drop in video traffic under three schedulers in scenarios 1 and 2

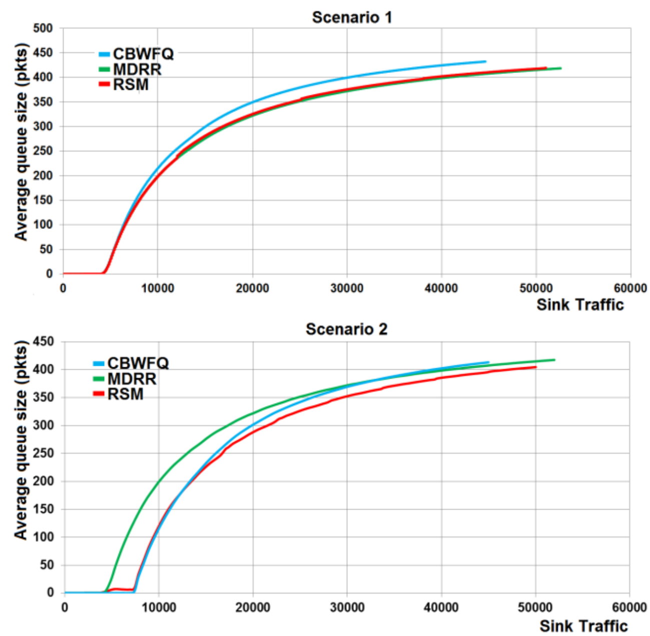
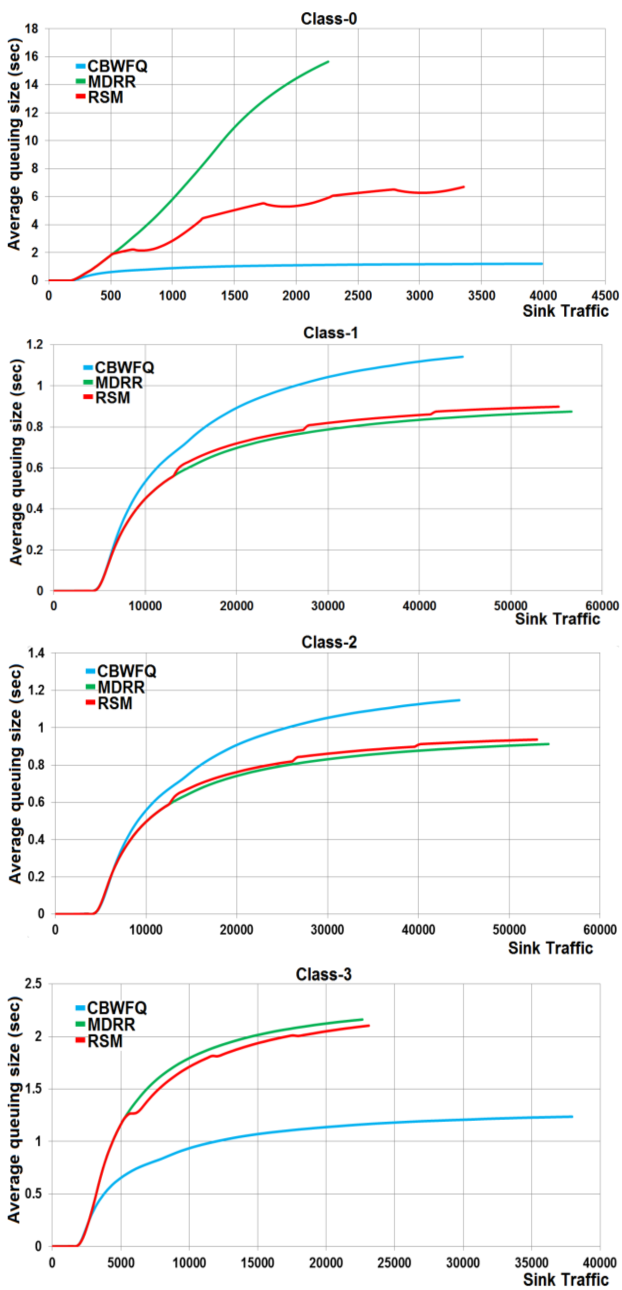


Figure 11. Average queue size of video traffic under three schedulers in scenarios 1 and 2

In this paper, we focus on the development a quality of service (QoS) model in DiffServ router using the reconfigurable technique for scheduling multimedia traffic. The Simevents/Stateflow software is used to show how the reconfigurable technique can improve the QoS (queue delay and packet loss) of multimedia application in DiffServ networks. SimEvents blocks implement the main system units as the traffic generators, enqueue, dequeue, and sink units. Stateflow block implement mainly the reconfiguration process in addition to some other small control process. 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 shows 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. The simulation result also shows that 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).



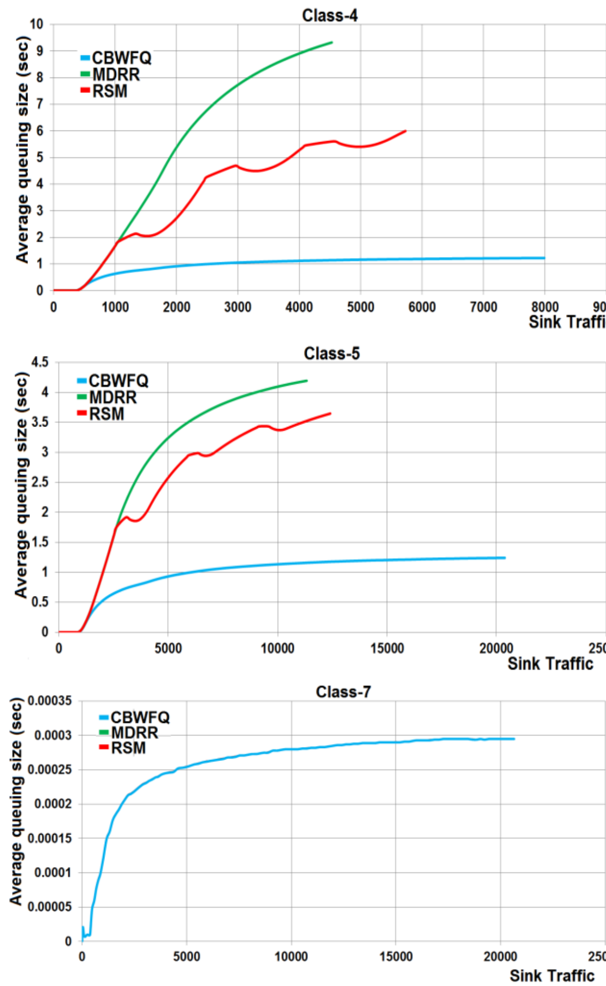


Figure 12. Delay in traffic class 1-6 and 8, under three schedulers in scenario 2

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