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Novel Packet Queuing Algorithm on Packet Delivery in Mobile Internet Protocol Version 6 Networks

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Abstract: New applications such as video conferencing and voice over IP present many challenges to the design of mobile networks. The mobile networks are constantly changing. The latest devices like smart phones, personal digital assistants, and mobile enabled laptops such as Windows Mobile and Windows Phone are truly able to deliver on any mobile broadband. As a July 2010 study by ERICSON, one of the top tier infrastructure suppliers for mobile networks shows, there are approximately five billions cell phone lines in the globe. This survey estimates 3.4 billion smart phone users in 2015. Thus, the Internet service providers must deliver a high quality of service to the customers. The key factor in quality of service is optimization for bandwidth allocation. Various queuing algorithms can be used in case of mobile IPv6 to control bandwidth allocation for instance. In this paper, we present various queuing disciplines in mobile IPv6 network when traffic class field in IPv6 is set to reserved, that is, the packets need quality of service throughout, from source to destination.

Keywords: Mobile IPv6, Queuing algorithm, Quality of Service, End-to-end delay.

1 Introduction

Real time applications such as voice over internet protocol (VOIP) and other multimedia traffic such as Internet Protocol television have driven the demand for increasing and guaranteed bandwidth requirements in the network. Internet service providers are seeking to deploy queuing algorithms [1] to schedule arriving packets in routers buffer between various types of packets and achieve fair bandwidth allocation in congestion conditions. This refers to the capability of a network to provide better service to packets with high priority, or time sensitivity. In this paper, we compare the effect of scheduling algorithms, i.e., first-in-first out (FIFO), priority queuing (PQ), and weighted-fair (WF) to control bandwidth allocation in mobile IPv6 networks. In FIFO [2], the first packet arrives at the intermediate router, and is the first one to be processed and transmitted. Therefore, if the waiting time for a packet is more than the time to live, then that packet will be dropped by the router. Several items can cause this situation, for example, if the buffer space of the router is full and a packet arrives, the router discards that packet. PQ [2–4] is a technique that processes arrival packets based on their priorities. This can be done by "traffic class" in an IPv6 packet. Priorities include reliability, delay, and throughput. Arrival packets with high priority are transmitted faster than packets with lesser priority. This technique ensures that during congestion in the network, packets with higher priority do not get delayed due to packets with lower priority. The main idea in WFQ [5,6] is to maintain a separate queue for each flow; hence, a weight is assigned to each queue. Packets are then serviced by applying a round robin algorithm [7]. WFQ effectively controls bandwidth allocation for each flow. This technique allocates a percentage of output bandwidth to the relative weight of each queue during congestion. The focus of our research is to conduct an in-depth study into the effects of various packet queuing algorithms on packet delivery over mobile IPv6 network. This study includes simulation and performance evaluation of FIFO, PQ, and WFQ disciplines. Simulation is conducted using OPNET IT Guru. The rest of the paper is structured as follows. In the

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next section, a mathematical model for computing finish time over weighted fair queue is presented. This is followed by a discussion and simulation results. A summary concludes the paper.

2 Modeling for Priority Queue

As shown in Figure 1, the arrivals are classified and then inserted to a buffer. Classifier classifies packets to QoS classes and scheduler determines the order of output. In the case of real-time application [8], priority queuing algorithms are used for scheduling traffic. There are two types of performance guarantees in QoS networks, deterministic service and statistical service. A deterministic service guarantees [1] that every packet from a flow satisfy worst case end-to-end delay bound and there is no packet drop in the network, while, a statistical service makes probability service guarantees by this module allows a small fraction of arrival to violate quality of service. Deterministic bounds are easier to determine while they lead to inefficiency in resource management. However, statistical bounds lead to improvement in link utilization and network gain. In general, resources required to service N flows with statistical bounds are much less than the resources required to service N flows in the case of deterministic bounds [9].

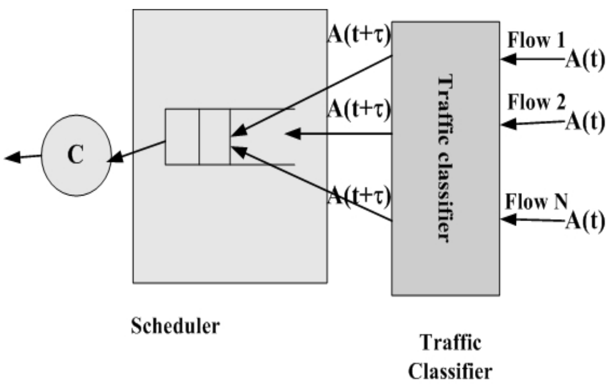

Fig. 1 Traffic Engineering

For considering k class of arrivals at the router, let us consider following functions [10] for the priority queuing,

1. there are n flows with priority;
2. Number of flows from class k (k=1,2,3,...,m) are finite with size N_k
3. the flow i of class k has been generated independently with exponential distributed and arrival time a_k and the arrival process is depicted with parameter $\lambda_k = \frac{1}{a_k}$;
4. each service time is generally distributed with first two moment, $s_k^{(1)} = s_k$ and $s_k^{(2)}$ $(k = 1,2,...,n)$
5. the mean response time of a router q_k for a k priority is as follow:

$$q_k = w_k + s_k + u_k \tag{1}$$

where,

w_k is waiting time for the $flow_k$ to be serviced.
s_k is service time
u_k is interruption time in case a flow of class k is interrupted by a higher priority flow.

the value of interruption time is determined by stream intensity of the higher priority flow and is determined by:

$$u_k = \frac{(\sum_{i=1}^{k-1} l_i)(\sum_{i=1}^{k-1} \Lambda_i)}{(1-\sum_{i=1}^{k-1} l_i)(\sum_{i=1} k-1(N_i\lambda_i)}.s_i \tag{2}$$

where, l_k is the utilization factor for k priority flows, Λ_k is mean rate of k-class of arrivals at the router. Parameter Λ_k can be determined from the following equation:

$$\Lambda_k = (N_k - n_k)\lambda_k \tag{3}$$

Following algorithm describes this queuing algorithm,

Algorithm 1 Priority Queue:

1: A new packet has arrived
2: Acquire it
3: Insert the new packet according to priority in subqueue 0
4: IF the insertion failed THEN {
5: Discard the packet 6: }
7: A request has been made to access the queue
8: IF queue is not busy THEN {
9: Access the high priority packet in the subqueue
10: Forward it to the destination.
11: }

3 Mathematical Modeling for Weighted Fair Queue

Finish time is the time taken by a queuing algorithm to transmit all the packets in the queue. Generally, the finish time for a queuing algorithm is computed by

$$F_i(k,t) = max(F_i(k-1,t), R(t) + P_i(k,t)) \tag{4}$$

Finish time [11] of a packet in the WFQ technique is computed by the following algorithm. Finish time for an active connection is the sum of the maximum finish time in current queue and the size of the arriving packets, divided by the queue weight. Moreover, finish time for an inactive connection is the sum of the maximum finish time in current queue and the size of kth arriving packet divided by queue weight. In other words,

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Pseudocode of algorithm 1: Priority Queue,

```
void acb_prio (OP_SIM_CONTEXT_ARG_OPT){

1,2: pkptr = op_pk_get (op_intrpt_strm ());
3: (op_subq_pk_insert (0, pkptr, OPC_QPOS_PRIO) != OPC_QINS_OK) {
4:        op_pk_destroy (pkptr);
5:        }
6:  if (!op_subq_empty (0))
      {
7:    pkptr = op_subq_pk_remove (0, OPC_QPOS_HEAD);

8:    op_pk_send_quiet (pkptr, 0);
9:     }
```

$$F_i(k,t) = max(F_i(k-1,t), R(t) + P_i(k,t))/W_i \qquad (5)$$

Where, $F_i(k,t)$ is finish number on each packet k in queue i at time t, $P_i(k,t)$ is the size of kth arriving packet in queue i at time t, $R(t)$ is the round number on each packet arrival, and W is the queue weight.

4 Simulation Results

We consider a mobile IPv6 scenario in order to carry out a comparison of the queuing algorithms over mobile IPv6 networks. Various queuing algorithms can be used in mobile IPv6 to control bandwidth allocation, for instance, FIFO, PQ, and WFQ queuing. Figure 1 shows our scenario. Simulation results are conducted using OPNET IT Guru [12]. It consists of one mobile node, which runs a video conferencing application and a voice over IP (VOIP) application simultaneously, two correspondent nodes that run the video conferencing and VOIP servers, one home agent, one access router in a foreign network, and two intermediate routers that interconnect the mobile node to the server. In this scenario, the mobile node runs a video-conference application and is located in its home network at the starting time. This node travels along the defined trajectory to the foreign network [13] and then gets back to its home network which, in this case, is the Computer department. The mobile node's average speed is 10km/h. Figure 1 depicts this scenario:

There are two types of performance guarantees in QoS networks, deterministic service and statistical service. A deterministic service guarantees that every packet from a flow satisfy worst case end-to-end bound and there is no packet drop in the network, while, a statistical service makes probability service guarantees by this module allows a small part of arrival to violate quality of service. Deterministic bounds are easier to determine while they lead to inefficiency in resource management. However, statistical bounds lead to improvement in link utilization and network gain. In general, resources

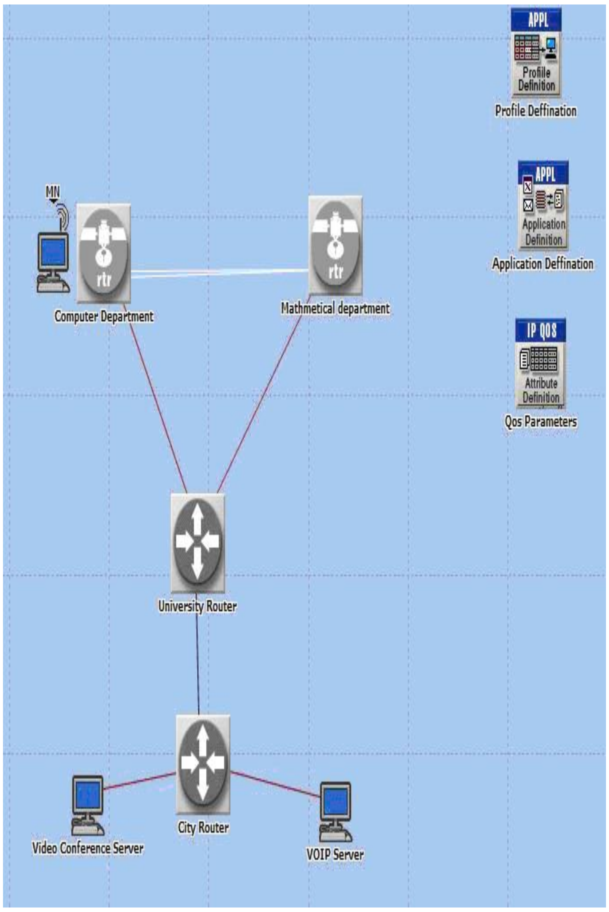

Fig. 2 Simulation Topology

required to service N flows with statistical bounds are much less than the resources required to service N flows in the case of deterministic bounds.

Figures 3 and 4 illustrate the average end-to-end delay for video conferencing and VOIP during periods of 600 seconds. As these figures show, the end-to-end delay

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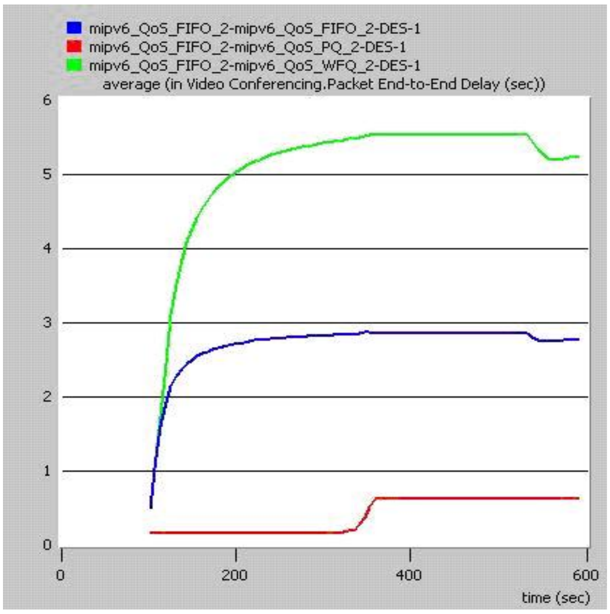

Fig. 3 Average End-to-end Delay, Video Conferencing

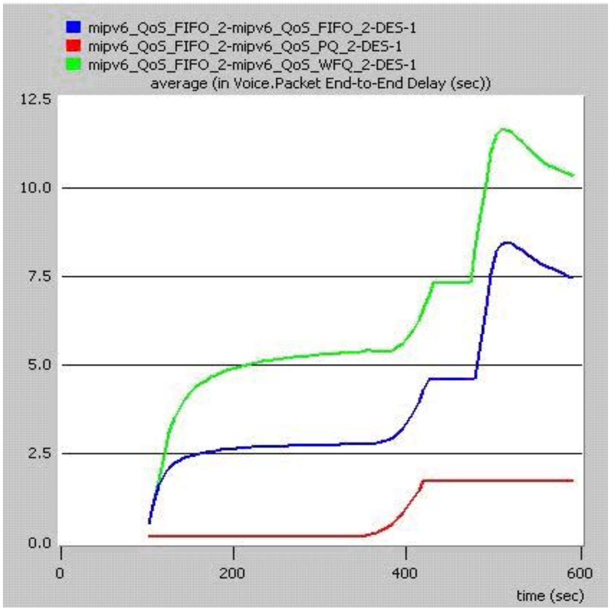

Fig. 4 Average End-to-end Delay, VOIP

[14] for video conferencing and VOIP application is higher in WFQ in comparison to FIFO and PQ. WFQ ensures that each flow has fair access to network resources and prevents burst flows from consuming more than their share of output [15–17] bandwidth. WFQ employs a hashing algorithm [18] that divides the flows over a limited number of queues either to be selected by the user or fixed by default. Thus, one can increase the number of queues as much as possible, which helps the fairness of the algorithm.

Figures 5 and 6 show a comparison of packet delay variation for WFQ, PQ, and FIFO. In both these figures, the video conferencing and VOIP application WFQ have a higher packet delay variation as it tries to maintain a separate queue for each flow by assigning a weight to each queue. Although WFQ had higher end-to-end delay, and a packet delay variation that is affected by the process of flow division over a limited number of queues, it did act better than FIFO and PQ in packet receiving. Figures 7 and 8 illustrate the received packets in video conferencing and VOIP for different algorithms of our study in Mobile Internet [19] networks. Both prove that WFQ has the best performance in packet receiving because it divides flows on different queues and runs round robin algorithms to ensure fairness between different flows. It helps more packets to be received at a destination compared with FIFO and PQ, where packet time to live could run out either due to buffer space limitations or increasing waiting time in the queue. When the time to live for a packet runs out, then the router drops that packet. Therefore, WFQ received more packets by using round robin algorithm.

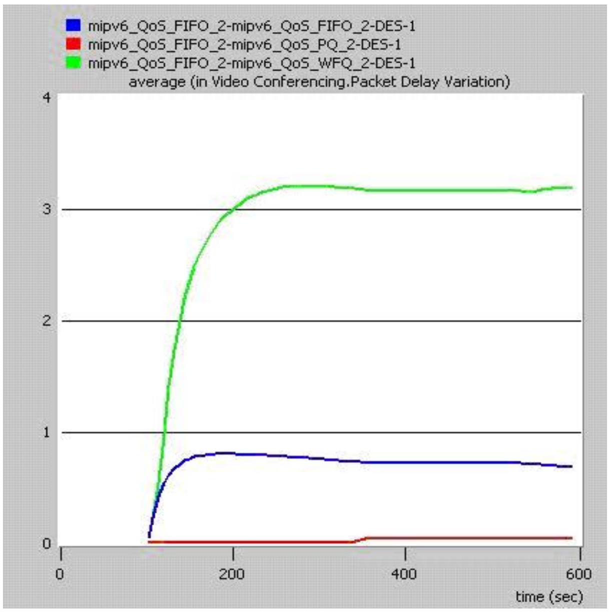

Fig. 5 Packet Delay Variation, Video Conferencing

5 State Probability Evaluation

Let us suppose to have a number of F flows stored on the video server. The number of video streams for each type of flow is supported fixed and the number of streams is N, i.e., $n_1 + n2 + n_3 + n_4 + n_F = N$. A flow is characterized by a number of bandwidth levels. The approach introduced in [20] consists on identifying the aggregation

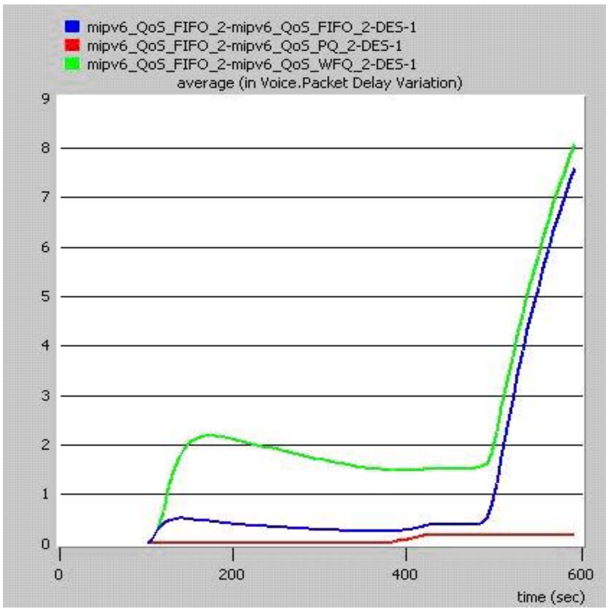

Fig. 6 Average Packet Delay Variation - VOIP

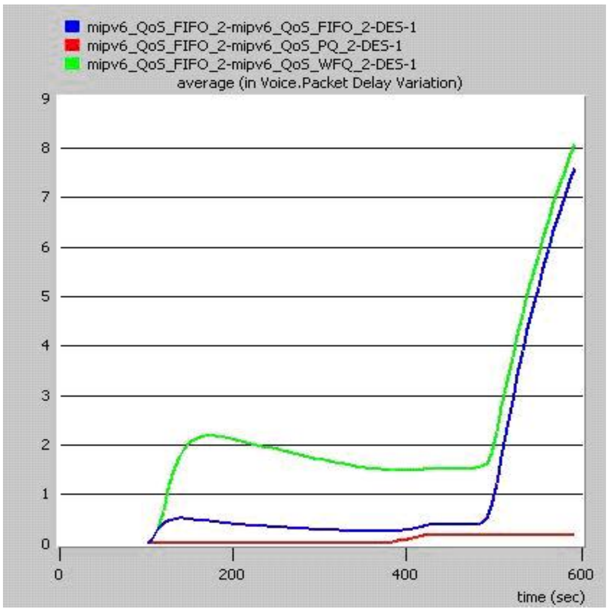

Fig. 7 Packet delay variation - VOIP

of video streams as a multi chain network of queue with different classes of customers [21], in which each of the bandwidth levels represents a service center. We suppose that the system of F flows is characterized by M service centers, corresponding to the total number of different bandwidth levels λ_m with $1 \leq m \geq M$). We suppose that the corresponding service center has a number of classes

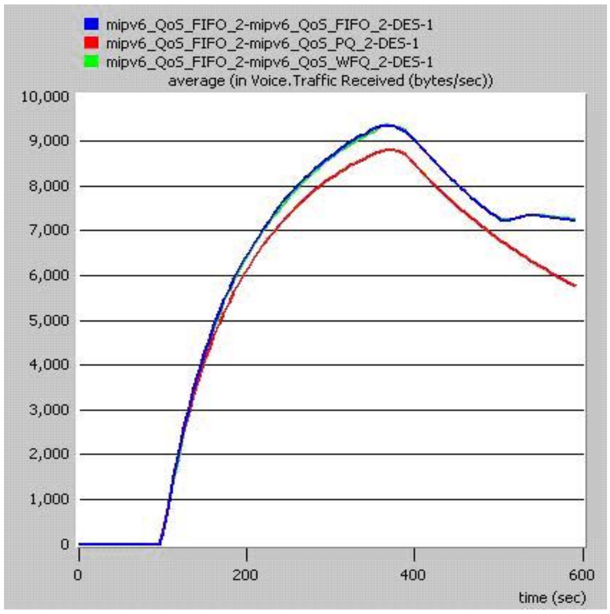

Fig. 8 Traffic received - VOIP

equal to the number of times state can be represented by the following vector[20] :

$$(n_{111}, n_{112}, ..., n_{1F1}, ..., n_{1FR_{1F}}, ..., n_{MF1}, ..., n_{MFR_{MF}} = n_{ijr} \quad (6)$$

where there are n_{ijr} users in the i thbandwidth level λ_i of the j th type of flows of class r, for $1 \leq i \geq M$, $1 \leq j \geq F$ and $1 \leq r \geq R_{ij}$. Then, we have

$$\sum_{i=1}^{M} \sum_{r=1}^{R_{ij}} (n_{ijr}) = n_j \quad j = 1, 2, .., F \quad (7)$$

The relative arrival rate[20] to the i th bandwidth level of the j th type of flow of class r, called r_{ijr}, can be determined as follow:

$$e_{ijr} = \sum_{k=1}^{M} \sum_{l=1}^{F} \sum_{s=1}^{R_{ij}} (e_{kls} P_{k,l,s;i,j,r}) \quad (8)$$

where $p_{k,l,s;i,j,r}$ represents the transition probability from the k th bandwidth level of the l th type of flow of class s to the i th bandwidth level of the jth type of class r.

6 Conclusion

In this paper, we presented end-to-end delay and average and-to-end delay for different queuing algorithms in mobile IPv6 networks and an evaluation of state probability as well. Simulation has been conducted using OPNET IT Guru. The results of this research help us to determine effect of various packet queuing algorithms on

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packet delivery over mobile IPv6 networks. As the results shown, WFQ mechanism is the best address which can guarantee quality of services and bandwidth allocation according to packets requirements. Although weighted-fair queuing offers higher end-to-end delay and packet delay variation, it receives more packets in comparison to FIFO, and PQ. WFQ ensures that each flow has a fair access to network resources and to prevent burst flows from consuming more than its share output bandwidth by round robin algorithm. Meanwhile, PQ is best address to deliver real-time traffics due to less end-to-end delay and packet delay variation.

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