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

**Reza Malekian**\(^1\), **Abdul Hanan Abdullah**\(^1\), **Ning Ye**\(^3\)

\(^1\)Faculty of Computer Science and Information Systems, Universiti Teknologi Malaysia, Johor, Malaysia

\(^2\)Department of Computer Science and Engineering, University for Information Science and Technology "St. Paul the Apostle", Ohrid, Republic of Macedonia

\(^3\)Department of Information Science, Nanjing College for Population Programme Management, Nanjing, China


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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
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. 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].

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

\[ F_i(k,t) = \max(F_i(k-1,t), R(t) + P_i(k,t)) \]  

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,
Pseudocode of algorithm 1: Priority Queue,

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

\[
F_i(k, t) = \max(F_i(k-1, t), R(t) + P_i(k, t))/W_i \quad (5)
\]

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 \(k\)th 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 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

Fig. 2 Simulation Topology

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

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

Fig. 5 Packet Delay Variation, Video Conferencing

Fig. 6 Packet Delay Variation, 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.

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., \( n_1 + n_2 + n_3 + n_4 + n_F = N \). A flow is characterized by a number of bandwidth levels. The approach introduced in [20] consists on identifying the aggregation
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 $l_m$ with $1 \leq m \leq M$. We suppose that the corresponding service center has a number of classes equal to the number of times state can be represented by the following vector[20]:

$$\begin{pmatrix} n_{i11}, n_{i12}, \ldots, n_{i1F}, \ldots, n_{iMF1}, \ldots, n_{iMF} \end{pmatrix} = n_{ijr}$$

(6)

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

$$\sum_{i=1}^{M} \sum_{j=1}^{F} (n_{ijr}) = n_{j} \quad j = 1, 2, \ldots, F$$

(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:

$$e_{ijr} = \sum_{k=1}^{M} \sum_{l=1}^{F} (e_{klr}p_{kl,i,j,r})$$

(8)

where $p_{kl,i,j,r}$ 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 $j$th type of class $r$.

### 6 Conclusion

In this paper, we presented end-to-end delay and average end-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...
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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References

Reza Malekian received Ph.D. in Computer Science from Universiti Teknologi Malaysia (Erasmus partner) in 2012, M.Eng. (with honor) in Information Technology Engineering from Iran University of Science and Technology in 2008, and B.Eng in Computer Engineering from North University in 2006. His research interests include computer and communication networks, design and evaluation of network and transport layers protocols to support of real-time traffics in wireless and mobile communications, mobile Internet, stochastic processing, and Internet engineering/future Internet.

Abdul Hanan Abdullah obtained his PhD degree from Aston University in Birmingham, United Kingdom in 1995. He has been the dean at the Faculty of Computer Science and Information systems for seven years from 2004 to 2011. Currently he is heading Pervasive Computing Research Group, a research group under K-Economy Research Alliances. His research interests include wireless sensor networks, mobile ad hoc networks, network security and next generation networks.

Ye Ning received the BS degree in Computer Science from Nanjing University, China, in 1994, the MS degree in School of Computer and Engineering from Southeast University, China, in 2004, and the Ph.D. degree in Institute of Computer Science from Nanjing University of Post and Telecommunications, China, in 2009. She worked as Visiting Scholar and Research Assistant in the Department of Computer Science, University of Victoria, Canada, in 2010. She is now an Associate Professor in the Department of Information Science of Nanjing College for Population Programme Management, China and holds an adjunct position at Nanjing University of Post and Telecommunications, China. Her research focuses on wireless networks and pervasive computing. She is a senior member of Chinese Computer Federation (CCF).