# Analysis of the algorithms for congestion management in computer networks

Szabolcs Szilágyi
University of Debrecen/Faculty of Informatics
Debrecen, Hungary,
szilagyi.szabolcs@inf.unideb.hu

***Abstract*—This paper presents one of the features of DS (Differentiated Services) architecture, namely the queuing or congestion management. Packets can be placed into separate buffer queues, on the basis of the DS value. Several forwarding policies can be used to favor high priority packets in different ways. The major reason for queuing is that the router must hold the packet in its memory while the outgoing interface is busy with sending another packet. Our main goal is to compare the performance of the following queuing mechanisms: FIFO (First-In First-Out), CQ (Custom Queuing), PQ (Priority Queuing), WFQ (Weighted Fair Queuing), CBWFQ (Class Based Weighted Fair Queuing) and LLQ (Low Latency Queuing).**

**Keywords—Congestion; queuing; OPNET; FIFO; PQ; CQ; WFQ; CBWFQ; LLQ.**

## I. INTRODUCTION

At the beginning computer networks were designed mainly for data transfer such as FTP and email, where delay was considered to be unimportant. In most cases the delivery service was effective, and the TCP protocol dealt with data losses. As the multimedia applications became popular (voice transfer, video conferences), separate telephone and video communication networks were set up. Nowadays, office and company networks are transformed into one converged network (see [1]), in which the same network infrastructure is used to ensure all the requested services.

Although converged networks have many advantages, there are some disadvantages too, namely the competition for network resources (buffers of routers), which leads to congestion. Delay in delivering the packets, jitter, loss of packets are consequences of congestion. Different applications show different sensitivity to these issues. For example, FTP is not impacted by delay and jitter, whereas the multimedia applications (video, voice) are very sensitive to them and the loss of packets too. QoS was introduced to handle this problem, and it is able to provide better multimedia performance (see [2]).

In the IP header there are some fields which can be used to make distinction between the packets of different applications, for example the Type of Service field. Different technics are used for congestion management (PQ, CQ, WFQ, CBWFQ, LLQ). Congestion avoidance (WRED), traffic shaping and traffic policing are also used by the QoS technology in order to control data traffic. This article focuses on the most important component, the congestion management.

Speed mismatches (see Fig.1.) and path aggregations (see Fig.2.) are the main reasons of congestion in computer networks. There are different algorithms which can overcome the mentioned problematic situations. Our purpose is to analyze and evaluate the efficiency of these algorithms using simulations. We are going to examine the following methods: FIFO, PQ, CQ, CBWFQ, WFQ and LLQ. It is important to note that these algorithms have real effect only in the case of congestion (see [3]).

The OPNET IT Guru Academic Edition (see [4]-[5]) was used to perform the simulation. The network topology for the performance evaluation is identical to the one used in former articles (see e.g. [6]-[8]). In this paper we continue to study the queuing technologies for congestion management. In [6] and [7] the authors considered three algorithms: FIFO, PQ and WFQ. The conclusion was that WFQ is the most efficient for multimedia applications. In addition to these we investigate three new algorithms: CQ, CBWFQ and LLQ. The main result of this paper is that for multimedia applications (mainly voice transfer) LLQ is more efficient then WFQ.

The detailed description of the algorithms has been discussed in several papers already (see e.g. [9]), now we would like to provide only a brief summary of them.

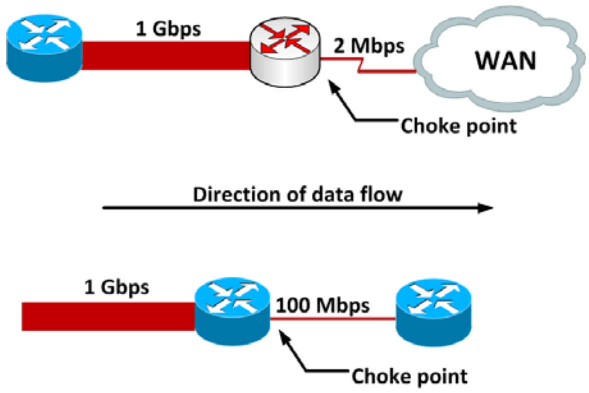


Fig. 1. Speed mismatches example in case of WAN and LAN environment [1]

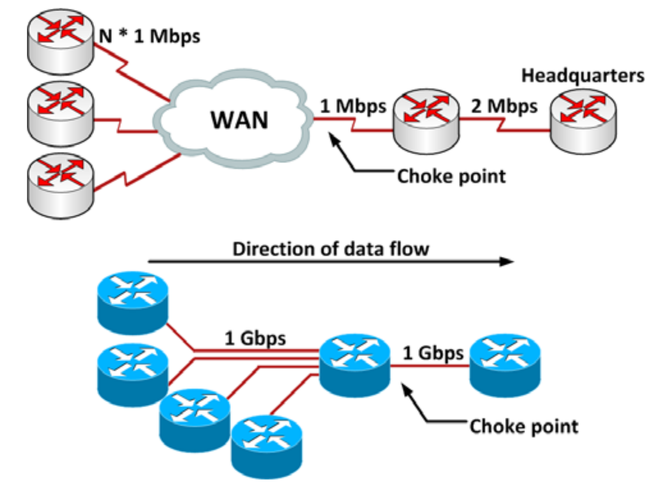


Fig. 2. Aggregation example in case of WAN and LAN environment [1]

### A. FIFO

FIFO is the simplest of all principles (see [10]). The incoming packets are served in the order of their arrival. The resource request of this method is quite small. The major problem is that it cannot provide priority for packets containing multimedia data, thus increasing the risk of jitter, delay.

### B. PQ

The Priority Queuing consists of four priority queues, each handled as a FIFO buffer (see [11]). The first queue has the highest priority, the others are in descending priority order. The packets which arrive into the highest priority queue are processed first, and then the next priority is served. The existence of packets in the highest priority queue is checked by the PQ scheduler after a packet is processed. If there is a new packet in this queue, it will be immediately processed. The strength of PQ is that it is able to ensure highest priority to multimedia applications (especially for voice transfer), but this can also lead to infinite delay for the packets belonging to the lower priority queues.

### C. CQ

The CQ serves to overcome the major disadvantage of PQ. The data flows are categorized into 16 FIFO queues by the network administrators. The buffer length of the queues can be defined. It enables to set the usable percentage of the total bandwidth for each FIFO queues. The Round Robin principle is used to schedule the 16 buffers (see [12]). Using CQ we can avoid the infinite delay, but CQ is not able to ensure priority for multimedia applications. However, the fine tuning of row lengths can help to reach acceptable results.

### D. WFQ

In the case of Weighted Fair Queuing the scheduling is completely automated, offering no tuning possibilities. WFQ works with data flows, which are grouped into a maximum of 256 queues. The data flows are classified by parameters like source IP address, destination IP address, type of transport protocol, IP packet header's ToS field (IP Precedence), source port number, destination port number. The queue index (as an ordinal number) is calculated by a hash algorithm. The WFQ scheduler uses the following notations and formulas (see [13]):

$$SN = Previous\_SN + (Weight * New\_Packet\_Length) \quad (1)$$

$$Weight = 32384/(IP\_Precedence + 1) \quad (2)$$

where *SN (Sequence_Number)* means the *Finish Time*.

For a more detailed description of WFQ and an example see [13].

### E. CBWFQ

The Class Based Weighted Fair Queuing is based on the idea of WFQ with the difference that instead of data flows data classes are considered. The packets are classified into the classes manually by the network administrator (not like in the case of WFQ, where it is performed automatically).

### F. LLQ

Low Latency Queuing is based on CBWFQ algorithm, but a strict priority queue is added to it (LLQ), which can be used for multimedia data transmission. This way the advantages of PQ and WFQ are combined and the disadvantages of these algorithms are reduced.

## II. Simulation Environment and Settings

We used the following network topology in OPNET IT Guru Academic Edition:

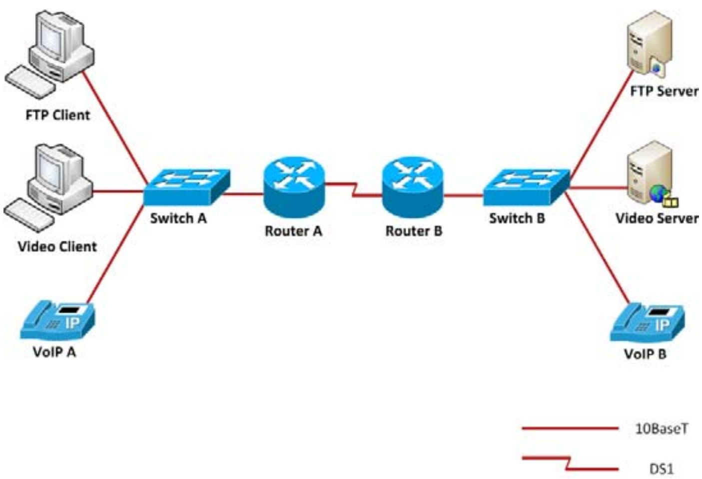


Fig. 3. Simulation network topology

The measurement environment consists of 2 routers, 2 switches and 6 hosts. The routers are connected with a point-to-point link, having the speed of *ppp-DS1*. The rest of the hosts are connected with *10BaseT*. The part between the two routers is actually a narrow cross-section where congestion can happen. For this reason the congestion management algorithm is activated in this area (see [15]-[16]). The next table summarizes the traffic parameter values used in the simulation:

TABLE I. SIMULATION SETTINGS

| Param. | FTP | Video | VoIP |
|---|---|---|---|
| **Start Time Offset (s)** | constant (5) | | |
| **Duration (s)** | End of Profile | | |
| **Repeatability** | Once at Start Time | | |
| **Operation Mode** | Simultaneous | | |
| **Start Time (s)** | constant (100) | | |
| **Duration (s)** | End of Simulation | | |
| **Repeatability** | Once at Start Time | | |
| **Inter-Request Time** | constant (1) | - | - |
| **File Size** | constant (1000000) | - | - |
| **Traffic Type** | High Load | Low Resolution Video | IP Telephony |
| **Frame Interarrival Time** | - | 10 frames/sec | - |
| **Frame Size** | - | 128x120 pixels | - |
| **Encoder Scheme** | - | - | G.729 |
| **Type of Service** | Best Effort (0) | Streaming Multimedia (4) | Interactive Voice (6) |

## III. SIMULATION RESULTS

Simulation makes possible to perform several measurements and statistical analysis. The following main indicators are investigated: the packet loss rate, the number of received packets, the delay of packets between the endpoints and jitter. The length of the simulation was 5 minutes in each case. Congestion management algorithms are activated only in the case of congestion. The graph below presents this fact: it is the 105th second when the congestion management algorithm becomes active and its performance study can be started. Figures 4-12 show the results of the simulation.

The highest rate of packet loss (see Fig.4.) was produced by the FIFO rule, as it could be expected. In the case of PQ, the loss of audio packets is zero, due to the existence of the highest priority queue. The loss of video and FTP packets is extremely high in this case (see Fig. 5.), as the highest priority queue (voice traffic) blocks the video and FTP communication. Using CQ we can get better results than using PQ, concerning the packet loss, but it requires a lot of fine-tuning work. WFQ, CBWFQ and LLQ are the most efficient mechanisms concerning the packet loss.

The following charts analyze the video traffic. Three indicators are examined: the number of received video packets, the delay of packets and the jitter (see Fig. 5-7). The number of received packets is inversely proportional with the loss of packets. That is why Figure 5 is identical with Figure 4. PQ is the only exception (as it was mentioned above).

Concerning the delay of video packets FIFO shows the most disappointing result and it is followed by CQ. The other algorithms (PQ, WFQ, CBWFQ and LLQ) present acceptable results. Considering the jitter in the case of video packets the worst performance is presented by the FIFO algorithm, CQ is a little bit better and the others' (PQ, WFQ, CBWFQ, LLQ) performance is the same.

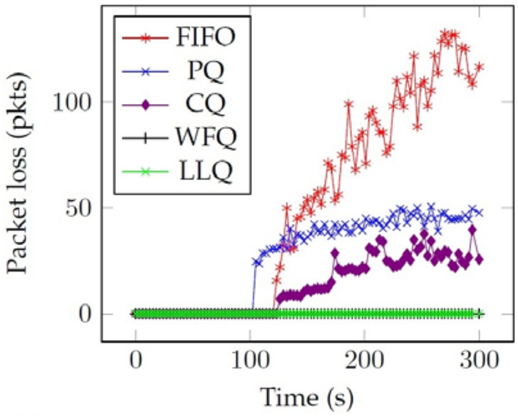


Fig. 4. IP packet loss

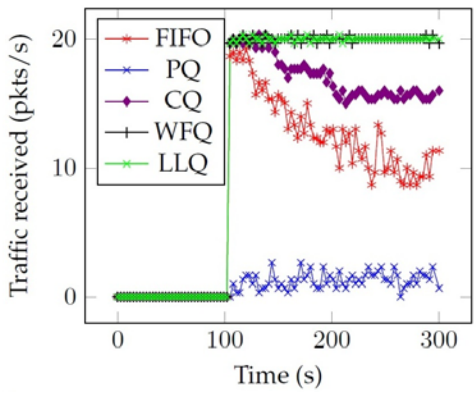


Fig. 5. Video: Traffic received

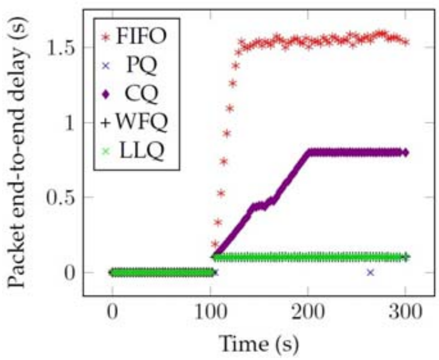


Fig. 6. Video: Packet delay

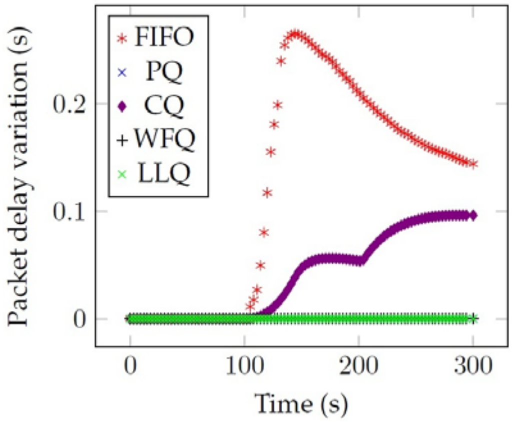


Fig. 7. Video: Jitter

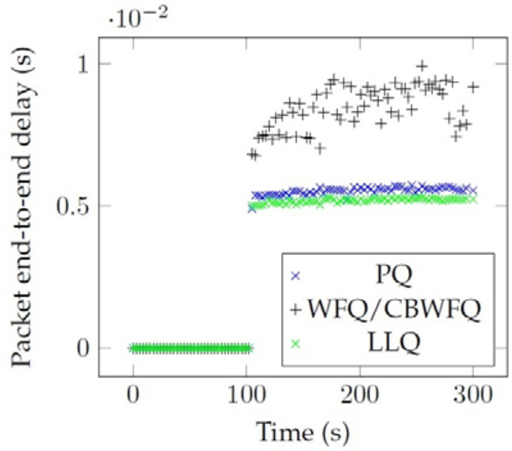


Fig. 10. VoIP: Traffic delay for PQ, WFQ/CBWFQ and LLQ

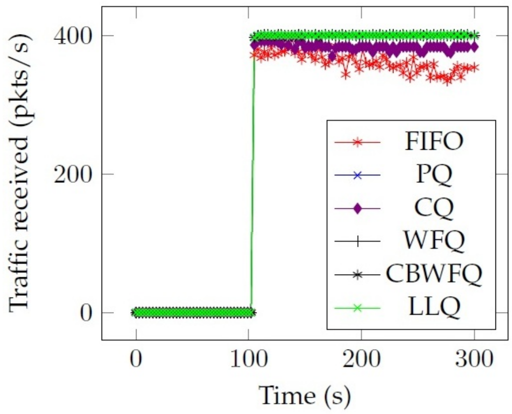


Fig. 8. VoIP: Traffic received

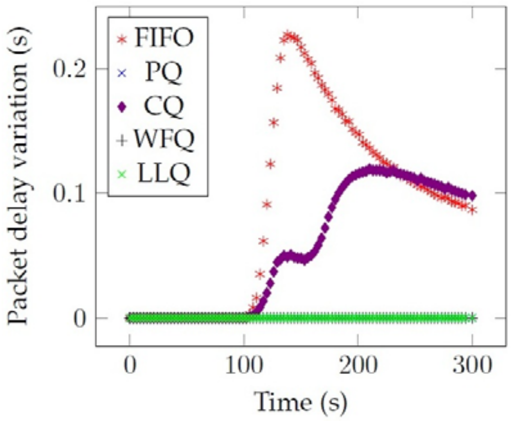


Fig. 11. VoIP: Jitter

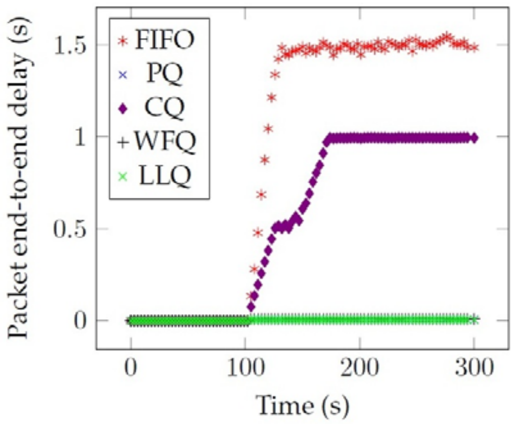


Fig. 9. VoIP: Traffic delay

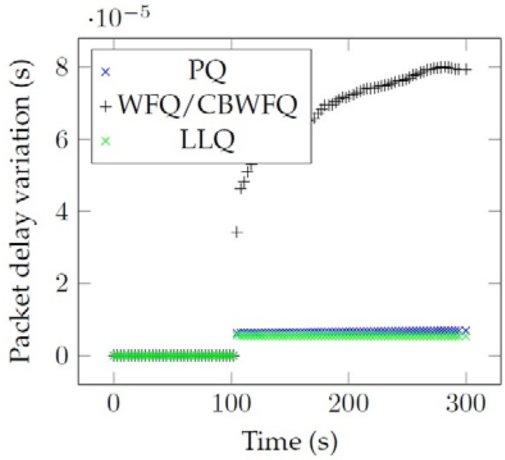


Fig. 12. VoIP: Jitter for PQ, WFQ/CBWFQ and LLQ

The same performance parameters were examined in the case of voice communication, namely: the number of received voice packets, the delay of voice packets and the variation in the delay of voice packets (see Fig. 8-12). Concerning the number of received voice packets the performance of the PQ, WFQ, CBWFQ and LLQ mechanisms is the most efficient. CQ and FIFO show a bit worse results but they are acceptable too. Interactive voice communication requires a maximum of 150 ms delay for voice packets (see [17]). FIFO and CQ cannot fulfill this criterion. QoS also requires that the jitter do not exceed 30ms. FIFO and CQ didn't fulfill this criterion too.

The four most efficient algorithms (PQ, WFQ, CBWFQ and LLQ) were compared in respect of the delay of voice packets and jitter. The reason of choosing the voice packets is that they are very sensitive to delay and jitter. The result is interesting (see Fig. 10 and 12). Previous articles (see e.g. [6]-[7]) showed that WFQ is the most efficient for voice packets. It can be observed that WFQ and CBWFQ have the highest dispersion for the delay. WFQ has the highest value for jitter, much higher than in the case of PQ and LLQ. It is obvious (see Fig. 10. and 12.) that LLQ has much better performance for voice packets than WFQ.

## IV. Summary

In this paper* we tried to present a short overview of congestion management algorithms used by routers. We managed to evaluate three more algorithms beside the ones published in former articles (see e.g. [6]-[7]). The simulation environment was provided by the OPNET IT Guru Academic Edition application, based on mathematical models. We used a generalized, extendable and factual network topology. The article concludes that LLQ is the most efficient algorithm for voice data transfer. Our next research topic is to examine and test the algorithms presented in the current article in a real network environment, as it is also a widely used by researchers (see e.g. [18]-[19]) to perform traffic measurements.

## References


[1] QOS, "Implementing Cisco Quality of Service", Student Guide, Vol. 1-2, Ver. 2.2,©Cisco Systems Inc., 2006

[2] A. S. Ranjbar, "CCNP ONT Official Exam Certification Guide", Cisco Press, 2007

[3] B. Durand, J. Sommerville, M.Buchmann, R. Fuller, "Administering Cisco QoS in IP Networks", Syngress Publishing, Inc., 2001

[4] OPNET official homepage, www.opnet.com, downloaded on 09.09.2013.

[5] A. Kuki, "Tools for Performance Evaluation and Modeling in Higher Education of Informatics", Informatics in the Hungarian Higher Education, Debrecen, Hungary, pp. 70-77, 2010

[6] M. M. G. Rashed, M. Kabir, "A Comparative Study of Different Queuing Techniques in VoIP, Video Conferencing and FTP", Daffodil International University Journal of Science and Technology, Volume 5, Issue 1, January 2010

[7] S. Akhtar, E. Ahmed, A. k. Saka, K. S. Arefin Performance Analysis of Integrated Service over Differentiated Service for Next GenerationInternet, © JCIT, ISSN 2078-5828 (print), ISSN 2218-5224 (online), Vol. 1, Issue 1, 2010

[8] S. Farhangi, S. Golmohammadi, "A comparative study between combination of PQ and MWRR Queuing techniques in IP network based on OPNET", Advances in Computer Science and its Applications, Vol. 1, No. 1, ©World Science Publisher, United States, March 2012

[9] Sz. Szilágyi, B. Almási, "A Review of Congestion Management Algorithms on Cisco Routers", Journal of Computer Science and Control Systems, University of Oradea, Vol.5, Nr.1, ISSN 1844-6043, 2012

[10] T. Svensson, A. Popescu, "Development of laboratory exercices based on OPNET Modeler", Master thesis, Blekinge Institute of Technology, Department of Telecommunications and Signal Processing, 2003

[11] S. Kashihara, "VoIP Technologies", INTECH Open Access Publisher, 2011

[12] M. Barreiros, P. Lundqvist, "QOS-Enabled Networks - Tools and Foundations", A John Wiley & Sons, Ltd., Publication, 2011

[13] W. Odom, J. Geier, N. Mehta, "CCIE Routing and Switching Official Exam Certification Guide", Second Edition, Cisco Press, 2006

[14] W. Odom, R. Healy, D. Donohue, "CCIE Routing and Switching Certification Guide", 4th Edition, Cisco Press, 2010

[15] A. S. Sethi, V.Y. Hnatyshin, "The Practical OPNET User Guide for Computer Network Simulation", CRC Press Taylor & Francis Group, NW, 2012

[16] L. L. Peterson, B. S. Davie, "Computer Networks: A System Approach - Network Simulation Experiments Manual", 3rd Edition, ©Elsevier Inc., 2011

[17] P. K. Verma, L.Wang, "Voice over IP Networks - Quality of Service, Pricing and Security", Lecture Notes in Electrical Engineering, Vol. 71, Springer, 2011

[18] B. Almási, A. Harman, "An Overview of the Multipath Communication Technologies", Proceedings of the Advances in Wireless Sensor Networks (AWSN), Debrecen, Hungary, 2013 (to appear)

[19] C. Mancas and M. Mocanu, "QoS Optimization in Congested Multimedia Networks" proceedings of the 36th TSP Conference, pp. 38-42, Rome, Italy 2013.


* The work was supported by the TÁMOP-4.2.2.C-11/1/KONV-2012-0001 project. The project has been supported by the European Union, co-financed by the European Social Fund.