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IMPLEMENTATION OF BASIC QOS MECHANISMS ON VIDEOCONFERENCING NETWORK MODEL

Časlav Livada, Drago Žagar, Josip Job

Preliminary notes

This research is the outcome of multiannual use of videoconferencing services and the emersion of various problems that come with videoconferencing applications. Video and audio delay, dropped connection, missing audio or video, are just some of the reasons for creating this paper. In this article quality of videoconferencing link in CARNet network is improved by implementing various QoS mechanisms. The obtained results of the videoconferencing simulation are represented in graphs which display dropped packets, delay and other videoconferencing parameters.

Keywords: *quality of service, videoconferencing network, QoS mechanisms*

Implementacija osnovnih mehanizama kvalitete usluge na modelu videokonferencijske mreže

Prethodno priopćenje

Ovo istraživanje je posljedica višegodišnjeg korištenja videokonferencijske veze te pojave raznih problema koji prate istu. Kašnjenje slike i zvuka, pucanje veze, prekid slike ili zvuka samo su neki od razloga zbog kojih je nastao ovaj rad. U ovom radu pokušava se primjenom mehanizama kvalitete usluge na modelu CARNet-ove mreže poboljšati kvaliteta videokonferencijske veze. Na osnovu dobivenih rezultata simulacije videokonferencijske veze prikazani su grafovi ispuštanja paketa, kašnjenja te ostalih parametara bitnih za videokonferencijsku vezu.

Ključne riječi: *kvaliteta usluge, videokonferencijska mreža, mehanizmi kvalitete usluge*

1 Introduction

This article investigates the problem of the quality of service decrease in CARNet network during high videoconferencing traffic.

The main objective of this research is to apply different queuing methods, i.e. FIFO, Priority Queuing and Weighted Fair Queuing on the proposed network. The ultimate objective is to create a model for the quality of service performance measurement based on the CARNet network.

To fully perceive the effects of the queuing mechanisms, a default network model will be simulated with low, medium and high traffic and will be compared to a network model with applied queuing mechanisms. Comparative analysis will yield a conclusion which will determine the best queuing mechanism for the given CARNet network.

One method of QoS improvement is a User-Oriented QoS Streaming System that can achieve perceptible satisfaction based on novel streaming and media differentiation policies in DiffServ networks. This method proposes Dynamic QoS Queue Mapping (DQ^2M) mechanism dynamically controls queue scheduling by adaptively maximizing the utilization of queues and network resources according to the soft states of the DiffServ network. DQ^2M algorithm can improve the fairness and efficiency of resource utilization for low-priority queues [5].

Another approach is a cost-based admission control (CBAC) which is a novel approach to preserve QoS in Internet Commerce systems. CBAC is a dynamic mechanism which uses a congestion control technique to maintain QoS while the system is online. Rather than rejecting customer requests in a high-load situation, a discount-charge model (which is sensitive to system current load and navigational structure) is used to encourage customers to postpone their requests. A scheduling mechanism with load forecasting is used to schedule user's requests in more lightly loaded time periods. The use of CBAC at high load achieves higher profit, better utilization of system resources and service times competitive with those which are achievable during lightly loaded periods [6].

In the context of a reconfigurable transport protocol framework, a QoS-aware Transport Protocol (QSTP) can be used, which is specifically designed to operate over QoS-enabled networks with bandwidth guarantee. QSTP combines QoS-aware TFRC congestion control mechanism, which takes into account the network-level bandwidth reservations, with a Selective ACKnowledgment (SACK) mechanism in order to provide a QoS-aware transport services that fill the gap between QoS enabled network services and QoS constraint applications. QSTP allows applications to reach their negotiated QoS over bandwidth guaranteed networks, such as DiffServ/AF network, where TCP fails [7].

2 Videoconferencing links

Videoconferencing represents two-way audio-visual communication in real time between two or more remote nodes. Videoconferencing can be very simple, i.e. communication between two locations (point-to-point communication) or very complex, i.e. linking between more locations containing multiple codes (multi-point communication). Videoconferencing can also be used, (except for audio or video signal), for data transmission.

Videoconferencing is used for various purposes: in formal lessons (courses, instructions, etc.), for connecting with guests and experts in various fields of expertise, for cooperation with distant schools on one project, for professional activities and social happening.

The core technology used in a videoteleconferencing is digital compression of audio and video streams in real time. The hardware and software that performs this compression is called a codec (coder/decoder) and can achieve compression rates up to 1:500. On the input side, codec has the role of the coder, i. e. codec takes the input signal and codes it (digitalizes and compresses the input signal) and this signal is sent through communication network. At the receiving end, codec takes the role of decoder, i. e. it decompresses the digitalized input and turns it into an analog signal. The quality of the sound and the picture depends greatly on the type of the used codec, since the codec loses some information during the compression process and on the communication link bandwidth. The effects of a bad and slow codec or low communication link bandwidth is broken picture and/or sound delay [1].

Hardware needed for videoconferencing:

- Video input (camera or webcam) and audio input (microphone)
- Video output (computer monitor, TV-set or projector) and audio output (loudspeakers)
- Data transfer (analog or digital telephone network, LAN or Internet).

There are two kinds of videoconferencing systems:

- Dedicated systems that have all required components packaged into a single piece of equipment. Usually it is a console with a high quality remote control that can be controlled in all directions and has the ability to zoom in and out. The console is connected to the computer that contains all the needed software or hardware-based codecs. Omnidirectional microphones, TV monitor with loudspeakers and/or a projector are connected to the console.
- Desktop systems are add-ons to normal PCs that enable videoconferencing, e.g. cameras and microphones. PC contains the necessary codec and transmission interfaces.

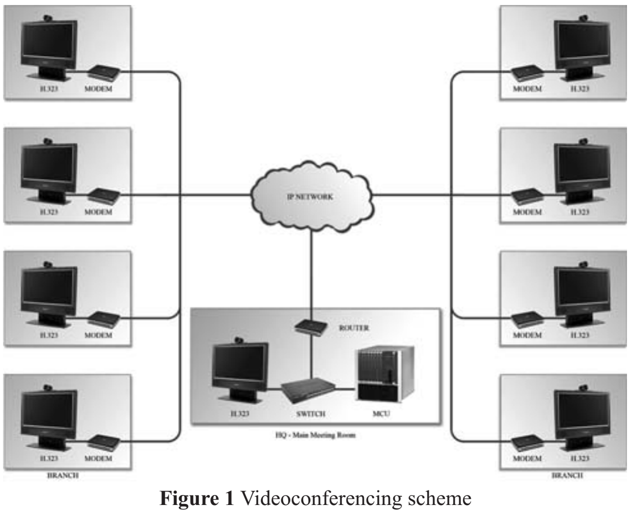

Figure 1 Videoconferencing scheme

Simultaneous videoconferencing (multi-point videoconferencing) refers to the communication among three or more remote points. Multi-point videoconferencing is achieved by using a Multipoint Control Unit (MCU). This is a bridge that interconnects calls from several sources (in a similar way to the audio conference call) as shown in Fig. 1. All parties call the MCU unit, or the MCU unit calls the parties which are going to participate in videoconferencing. The main purpose of MCU is to connect incoming calls and to manage the calls that connect subsequently.

Some systems are capable of multipoint conferencing with no MCU, stand-alone, embedded or otherwise. All of these use a standards-based H.323 technique known as "decentralized multipoint", where each station (in a multipoint call) exchanges video and audio directly with the other stations with no central "manager" or other bottleneck. The advantage of this technique is that the video and audio will generally be of higher quality because they do not have to be relayed through a central point [1].

2.1 Advantages and disadvantages of videoconferencing

As interactive communication medium, two-way video communication enables visual connection and interaction between participants which enhances understanding and helps the participants in creating a bond. Research shows that videoconferencing technology influences students in a positive way [2]:

- Increased motivation
- Better communication and appearance
- Better connection with the exterior world
- Detailed studying.

Two problems that prevent videoconferencing to become the communication standard are:

- Eye contact
- Appearance consciousness.

3 QoS improvement methods

The term QoS refers to a set of control mechanisms and not to the achieved quality of service. Quality of service is the ability to assign priorities to various applications, users or input streams, i.e. quality of service can guarantee requested transfer rate, delay, etc. Quality of service is important if the network bandwidth is insufficient, especially for media streaming, i.e. in our case for videoconferencing [4].

Quality of service is affected by two main factors:

- Human factor (stability and accessibility of service and delay)
- Technical factor (reliability, scalability, efficiency and sustainability).

During the communication, a packet can be late or not come at all and these problems can be classified as:

- Dropped packets (router can drop packets if its buffers are full and the amount of dropped packets depends on the state of network)
- Delay (if the packet is retained in long queues, it can reach its destination later than it should)
- Jitter (packets arrive at destination with different delays and oscillations in delay have bad effect on streaming audio or/and video)
- Wrong packet delivery (when series of packets travel through network in different directions, each with its own delay, finally come to their destination, their arrangement is different from the starting one)
- Error (an error occurs if the packets are routed in a wrong way or if they just become corrupted).

Packet classification is the main quality of service building block because without classification, all packets would be treated equally. Packet classification is a quality of service component that recognizes and differs various streams of data (Fig. 2) [7].

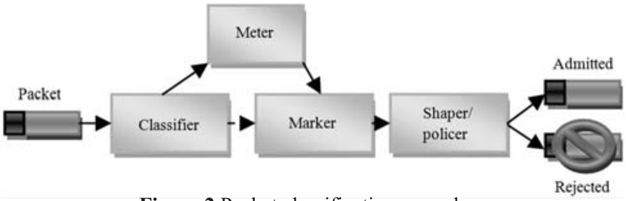

Figure 2 Packet classification procedure

Second stage of quality of service improvement is congestion control. A surge of traffic can overload the network bandwidth and that traffic excess could be dropped. To avoid excess packet dropping, packets are placed in queues. Some advanced queuing algorithms are:

- FIFO
- Priority Queuing
- Weighted Fair Queuing.

3.1 FIFO

FIFO, i.e. First In First Out is the simplest queuing method with the least load on processor. This mechanism does not have congestion control capability but it only forwards packets in the order in which they came [4]. This mechanism is used in interfaces that have throughput greater than 2Mbs, like Ethernet. The FIFO mechanism procedure can be seen in Fig 3. FIFO is good for networks in which packets arrive uniformly so that the queues could not get congested [12, 13].

Figure 3 FIFO mechanism

3.2 Priority Queuing

This mechanism supports rows of queues with various priorities, from high priority queue to low priority queue. Queues are handled in strict order taking priorities in consideration in such manner that high priority queues are serviced first, then the queues with lower priority.

If the low priority queues are being serviced and packet enters a high priority queue, that queue is instantly served. This mechanism is good for important traffic but it can lead to queue starvation. If the number of incoming packets is small, this phenomenon will not be manifested. Fig. 4 shows the procedure of this mechanism [12, 13].

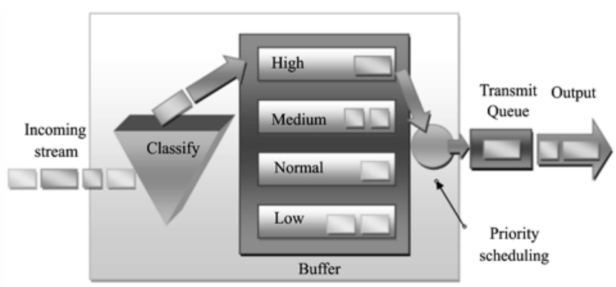

Figure 4 Priority Queuing mechanism procedure

Work algorithm of priority scheduling block can be seen in Fig. 5.

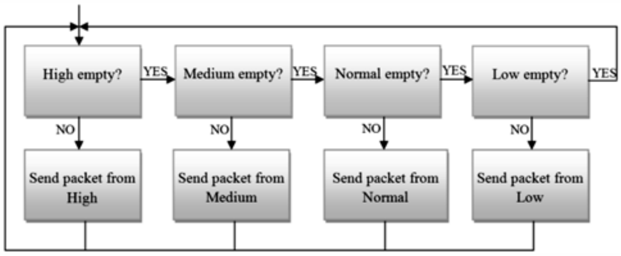

Figure 5 Different priorities packet sending algorithm

3.3 Weighted Fair Queuing

Weighted Fair Queuing mechanism arranges input data stream into rows using various criteria that can be: source address and port, destination address and port, ToS – Type of Service [4]. Weights are allocated to arranged rows of input data depending on the type of service and are serviced by their weights [12, 13]. Weighted Fair Queuing algorithm is shown in Fig. 6.

Incoming traffic is sorted into *N* rows and 1/*N* bandwidth is assigned to each row. Disadvantage of WFQ is that it needs more sorting than other approaches. In WFQ interactive traffic is moved to the beginning of the row to reduce response waiting time and then the residual bandwidth is distributed equally.

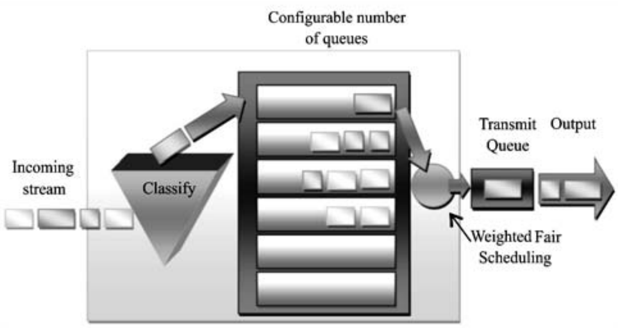

Figure 6 Weighted Fair Queuing mechanism procedure

4 QoS videoconferencing parameters acquiring by CARNet network model simulation

CARNet network structure according to which a model is made can be seen in Fig. 7.

Model consists of three centers: Rijeka, Split and Osijek that connect through their private switches onto a router in Zagreb which is in turn connected to a switch Zagreb connected onto servers that enable video-conferencing.

Individual centers have clients connected to them which participate in videoconferencing:

- Switch Osijek: client Osijek, client Slavonski Brod, client Vinkovci and client Vukovar
- Switch Rijeka: client Rijeka, client Opatija and client Pula
- Switch Split: client Split, client Zadar, client Šibenik and client Dubrovnik.

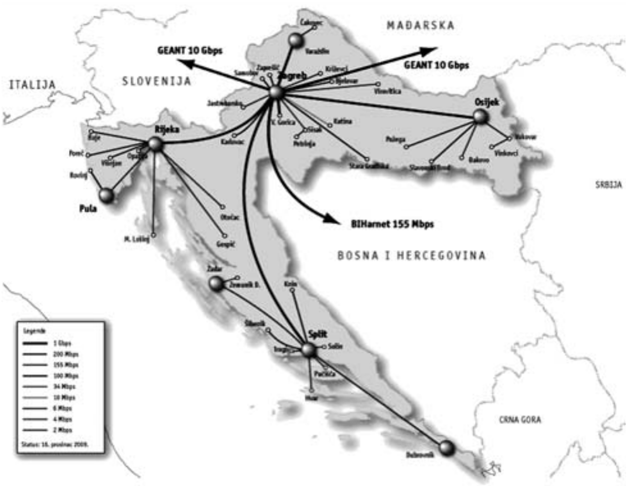

Figure 7 CARNet network

Model of this system is shown in Fig. 8.

Before the simulation, type of service is defined in the application properties which will be used in the simulation. There are eight types of services [5]:

- Best effort
- Background
- Standard
- Excellent effort
- Streaming multimedia
- Interactive multimedia
- Interactive voice
- Reserved.

Besides defining the nodes and their mutual connections, characteristics and specific parameters can be defined in Application Config, Profile Config and QoS Config blocks.

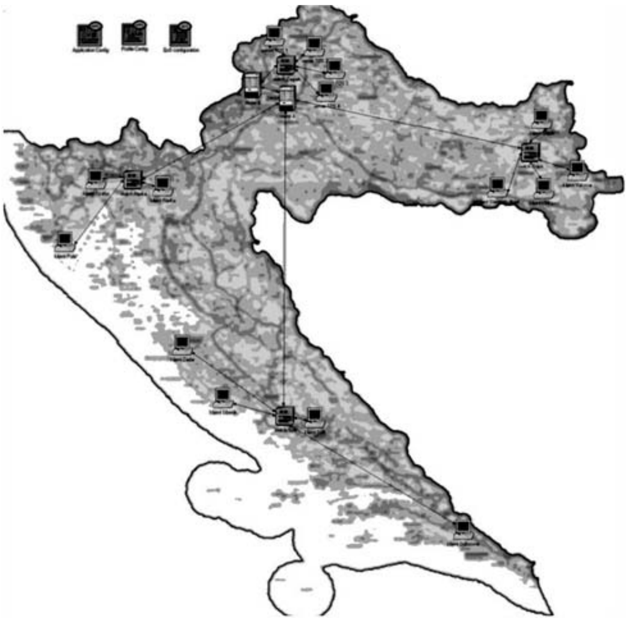

Figure 8 Videoconferencing network model

This model has four kinds of videoconferencing applications:

- Background traffic
- Standard traffic
- Excellent effort traffic
- Streaming multimedia.

Types of applications defined in a model are generated according to Tab. 1 from [4].

Table 1 Videoconferencing application profiles

Videoconferencing application	Parameters		
	Frame refresh rate / frames/s	Frame size / pixel	ToS
Background traffic	30		background (2)
Standard traffic	30	352×240	standard (3)
Excellent effort traffic	30	352×240	excellent effort (1)
Streaming traffic	30	352×240	streaming multimedia (5)

Certain clients/nodes connected to their centers have different types of application (Tab. 2).

A model (called baseline model) without any queuing mechanisms will be simulated first and then models with applied before mentioned queuing mechanisms.

Table 2 Application related to clients

	Application type			
	Background traffic	Standard traffic	Excellent effort traffic	Streaming traffic
Clients	Pula	Osijek	Vinkovci	Vukovar
	Opatija	Dubrovnik	Slavonski Brod	Zadar
	Rijeka		Šibenik	
			Split	

An important parameter for FIFO is the queue length, i.e. maximum number of packets in queue. If the number of packets exceeds the limit, these excess packets will be dropped. In our case queue length is 500 packets.

In Priority Queuing it is important to define the input stream classification method. Packets can be sorted based on the type of service, protocol or port. Priority Queuing classification parameters can be seen in Tab. 3.

Table 3 Priority Queuing classification profiles

	Classification schemes					
	ToS		Protocol		Port	
Priority	Queue length	ToS	Queue length	Protocol type	Queue length	Port type
Low	80	best effort	20	OSPF IGRP EIGRP	20	FTP server
		background				
Normal	60	standard	40	TCP	40	Application server
		excellent effort				
Medium	40	streaming multimedia	60	UDP	60	Voice traffic server
		interactive multimedia				
High	20	interactive voice	80	ICMP	80	Videoconferencing server
		reserved				

Packet classification and weight allocation of Weighted Fair Queuing by type of service is done towards Tab. 4 and when using WFQ classification by protocol type and port type the queue length is 500.

Table 4 WFQ classification by type of service

Weight	Type of service	Queue length / number of packets
1	best effort (1)	500
10	background (2)	500
20	standard (3)	500
30	excellent effort (4)	500
40	streaming multimedia (5)	500
50	interactive multimedia (6)	500
60	interactive voice (7)	500
70	reserved (8)	500

5 Videoconferencing QoS analysis at different traffic densities

Simulation results without and with queuing mechanism will be displayed on the same graph because of easier comparative analysis. Three simulations will be made with difference in traffic density, i.e. a parameter Traffic Scaling Factor will influence the amount of traffic in network. TSF of 1,0 will be default, medium traffic, TSF of 0,5 will be low traffic, and TSF of 2,0 will be high traffic.

Four parameters that depict quality of service in videoconferencing connection will be measured:

- Traffic received
- Packet delay variation
- End-to-end delay
- Dropped traffic.

Results will be shown according to Traffic Scaling Factor.

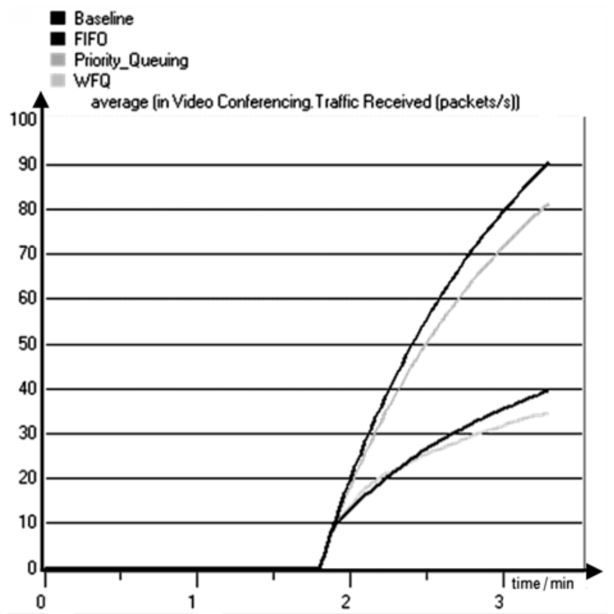

Figure 9 Traffic received, TSF = 1,0

5.1 Medium traffic density (TSF = 1,0)

Time average of the traffic received is shown in Fig. 9 on *y* axis and on the *x* axis (in minutes). Traffic starts in the 100th second of the simulation (1 min and 40 s). From Fig. 9. we can see that the largest amount of traffic is received in the Baseline model because it does not care for the delay nor the delay variation. The FIFO model receives half the amount

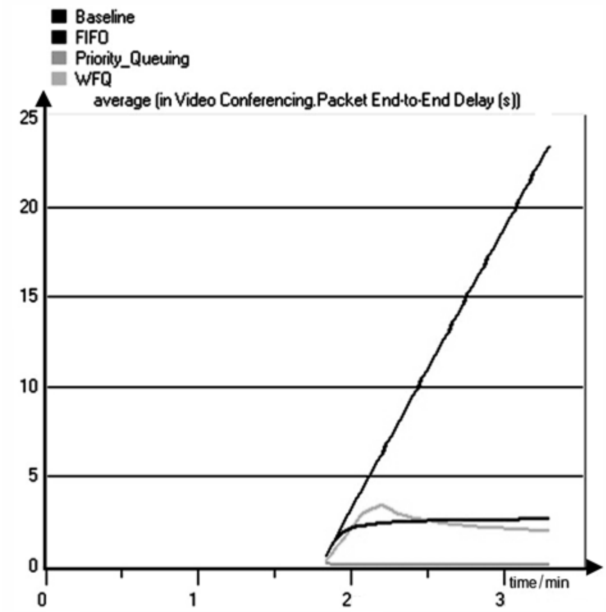

Figure 10 Packet end-to-end delay, TSF = 1,0

of traffic as the Baseline model and the WFQ is even worse than FIFO. Priority Queuing model receives the amount of traffic close to the Baseline model with one little difference: it cares about the delay and delay variation which will be seen in figures to come.

Next figure i.e. Fig. 10 shows the packet end-to-end delay in seconds. The Baseline model does not care for the end-to-end delay which can be seen in Fig. 10 where the delay has linear growth from the point the traffic is turned on. The FIFO and WFQ models have similar results which are far better than the Baseline model. Priority Queuing end-to-end delay is close to zero and that makes this model the best in end-to-end delay for videoconferencing.

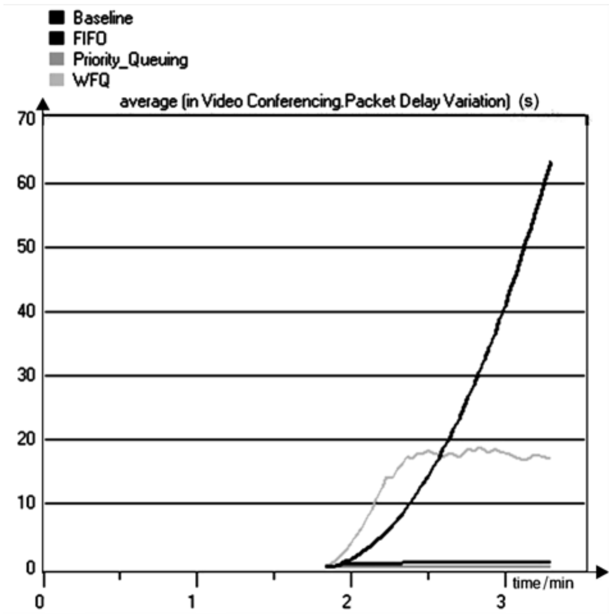

Figure 11 Packet delay variation, TSF = 1,0

Next parameter for measuring the QoS in videoconferencing, the packet delay variation is shown in Fig. 11. The Baseline model manifests (as expected) the worst results where the packet delay variation grows exponentially with time. The packet delay variation for the WFQ model settles in a certain value with little oscillations.

FIFO and Priority Queuing have similar results (Priority Queuing had slightly better packet delay variation).

Fig. 12 shows the traffic dropped in IP network. The Baseline model does not have any queues therefore in that model all traffic is received and no traffic gets dropped but we have seen the repercussions of that in Fig. 10 and Fig. 11. FIFO and WFQ models drop the biggest amount of packets and the Priority Queuing turned out to be superior once again.

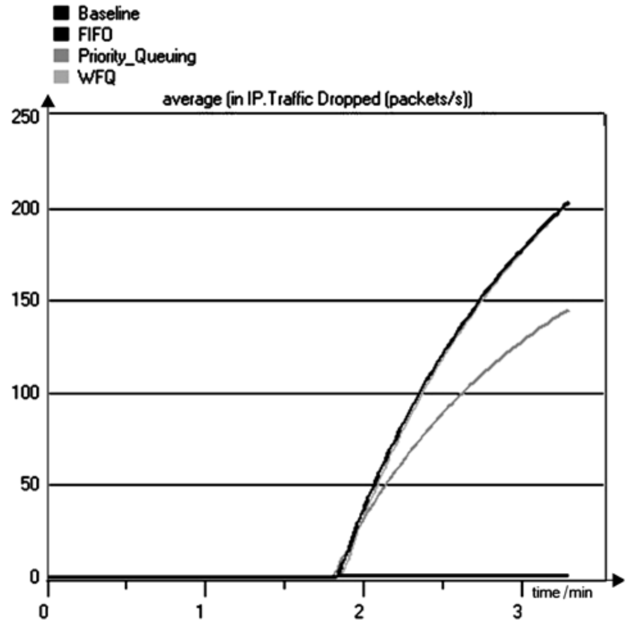

Figure 12 Traffic dropped, TSF = 1,0

The previous four figures (Fig. 9 to Fig. 12) show the simulation results for the network with the medium traffic, i.e. Traffic Scaling Factor is 1,0. We can clearly conclude that the best model for videoconferencing with medium traffic density is Priority Queuing.

5.2 Low traffic density (TSF = 0,5)

The next four figures (Fig. 13 to Fig. 16) will show the simulation results in a network with the half amount of traffic, where TSF is 0,5.

As we can see from Fig. 13, the traffic received for the model where the TSF is 0,5 is equal for all the models in the period of 200 seconds.

The packet end-to-end delay in the model with the half amount of traffic is smallest in the Priority Queuing model. The Baseline model and the WFQ have similar packet end-to-end delay, which has a linear growth. The FIFO model is similar to Baseline model and the WFQ for a short period of simulation and then the packet end-to-end delay is a little bit shorter.

Fig. 15 shows the packet delay variation in the network with the TSF of 0,5. It is clear that the best results are achieved with the Priority Queuing and the worst with the WFQ. The Baseline and the FIFO model have mediocre results and are congenial.

Traffic dropped in the network with the smaller amount of traffic is shown in Fig. 16. The Baseline model, again, has no dropped traffic and the remaining three models start to drop traffic in different time periods. The Priority Queuing starts dropping packets first, then comes the FIFO and finally the WFQ model.

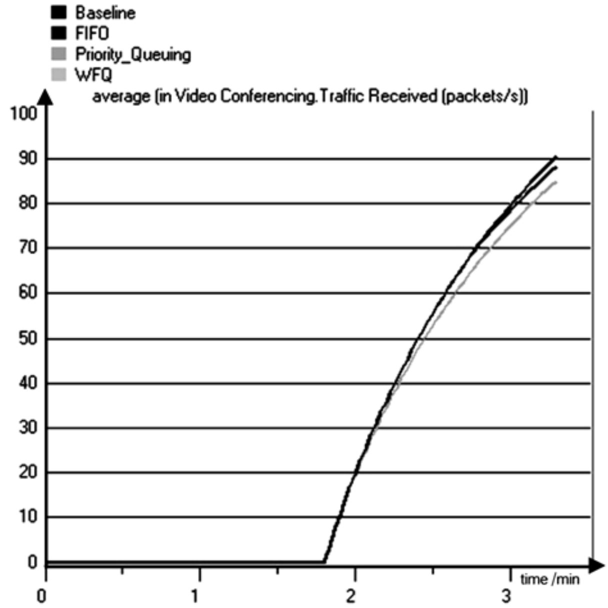

Figure 13 Traffic received, TSF = 0,5

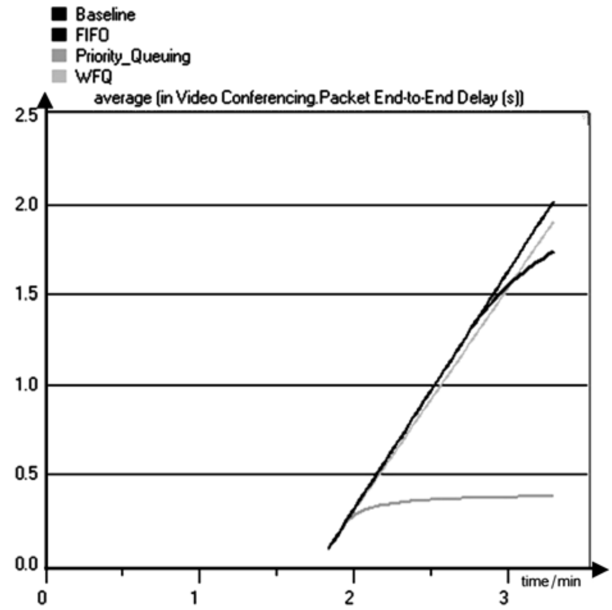

Figure 14 Packet end-to-end delay, TSF = 0,5

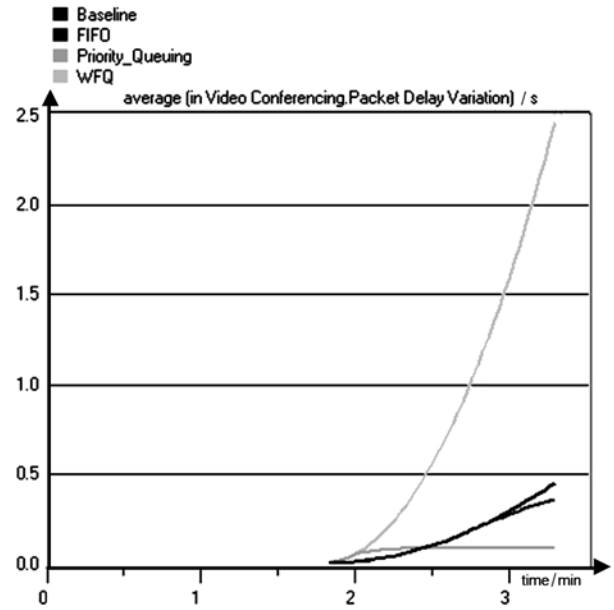

Figure 15 Packet delay variation, TSF = 0,5

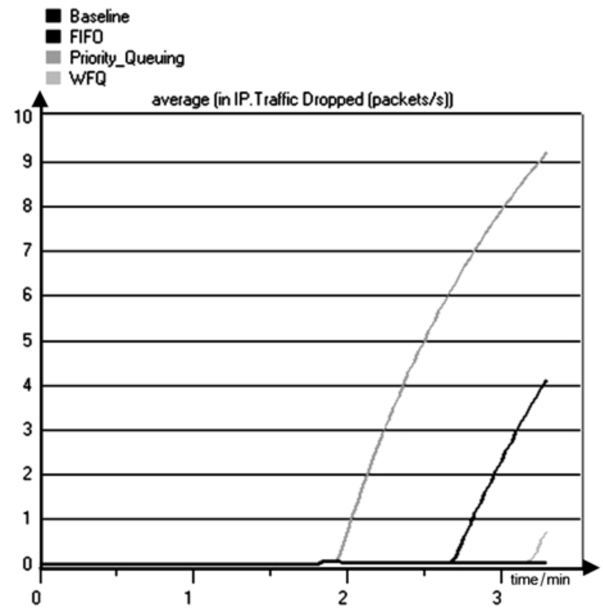

Figure 16 Traffic dropped, TSF = 0,5

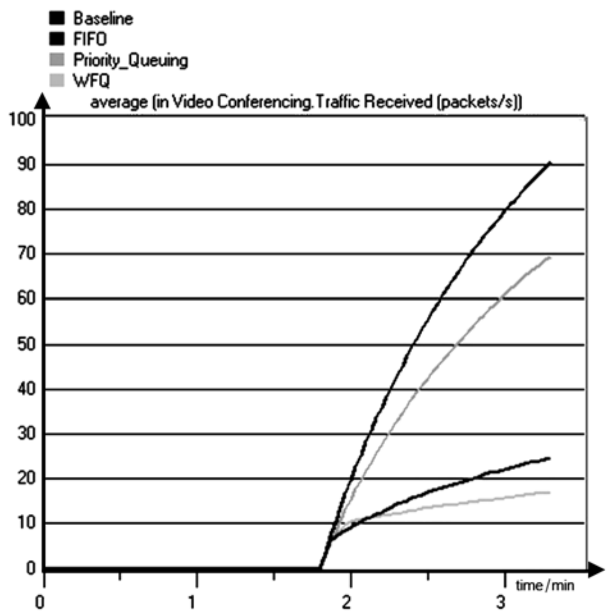

Figure 17 Traffic received, TSF = 2,0

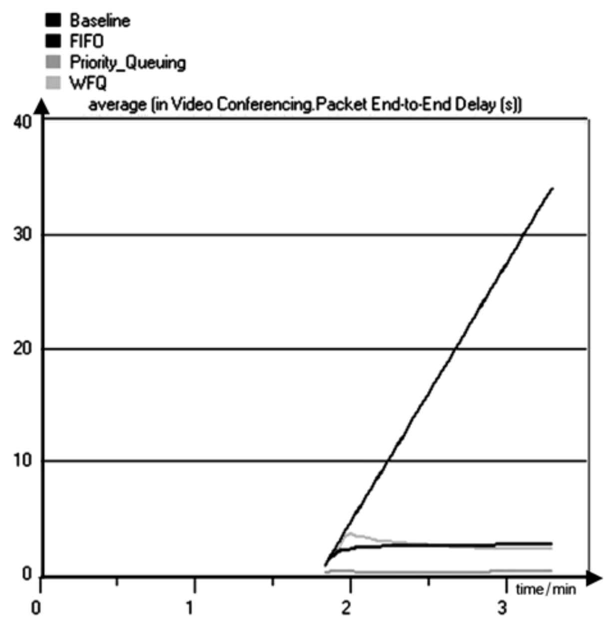

Figure 18 Packet end-to-end delay, TSF = 2,0

Again, as with the TSF 1,0, the best queuing mechanism for this network is Priority Queuing. It offers the smallest end-to-end delay and smallest packet delay variation (two most important parameters for the quality of videoconferencing).

5.3 High traffic density (TSF = 2.0)

Next four figures (Fig. 17 to Fig. 20) show the simulation results when the traffic is doubled, i.e. TSF equals 2,0.

As we can see from Fig. 17, when the traffic load is high, the best queuing algorithm is Priority Queuing, except for the Baseline model which receives most of the traffic. The worst queuing algorithm when TSF = 2,0 is WFQ.

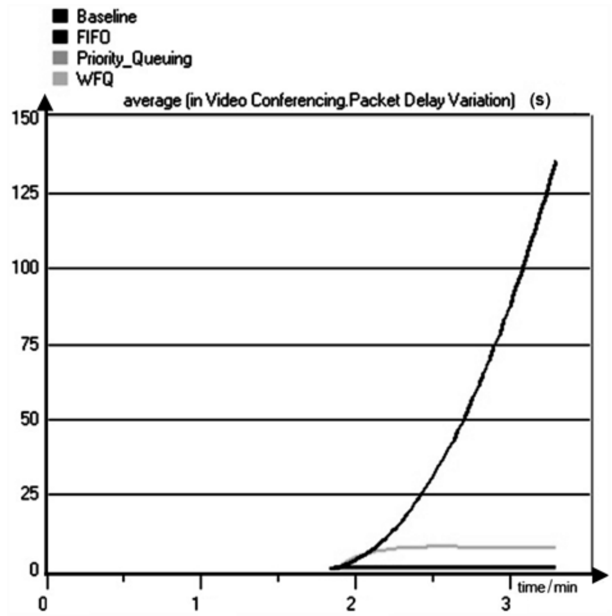

Figure 19 Packet delay variation, TSF = 2,0

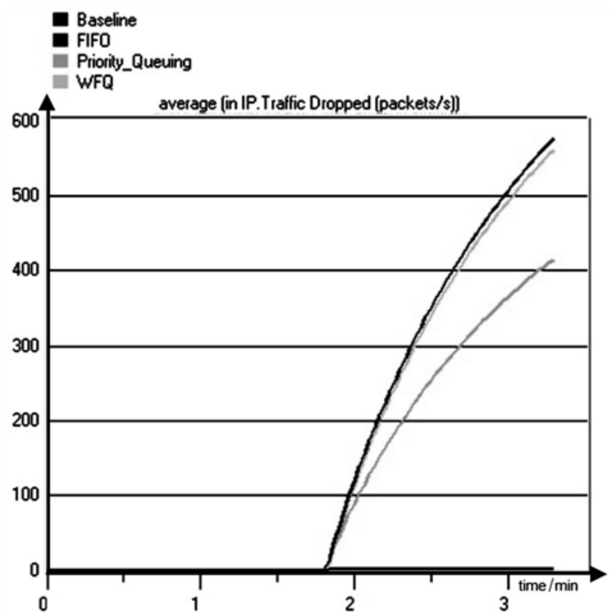

Figure 20 Traffic dropped, TSF = 2,0

The packet end-to-end delay when traffic density is high is very similar to packet end-to-end delay when traffic load is medium, i.e. the best algorithm is Priority Queuing and the worst, of course, is the case without any queuing algorithms. The FIFO and WFQ models have resembling results for the time measured.

Figure showing packet delay variation when the traffic load is high shows that the worst results are in the Baseline model, and the best results are achieved when Priority Queuing is used. FIFO has very similar results to Priority Queuing but is slightly worse. The WFQ is, again, in the middle.

WFQ and FIFO models drop the highest amount of traffic. Priority Queuing is the model with applied queuing mechanism that has the best result in traffic dropped, i.e. it drops the least amount of packets. The Baseline model, of course, drops no traffic.

6 Conclusion

The purpose of this paper was to investigate which QoS mechanism is the most appropriate for the proposed CARNet videoconferencing network model. The tested QoS mechanisms are FIFO (First In First Out), Priority Queuing and WFQ (Weighted Fair Queuing). FIFO mechanism can be used when we want survey the traffic because packets enter the queue one by one. WFQ and Priority Queuing usage depends on the network structure, user requirements and, of course, Quality of Service, i.e. these mechanisms are very customizable and adaptable.

FIFO and WFQ mechanisms gave similar results on improving QoS when the traffic load was medium. Packet delay variation in the WFQ model grows when the traffic is turned on and it reaches one value with minimal oscillations. Measurement results tell us that the Priority Queuing model has the resembling amount of received packets as the Baseline model with three differences: it has the smallest packet end-to-end delay (close to zero), the smallest packet delay variation and the smallest number of dropped packets.

It is recommended that the maximum value for packet delay variation is 30 ms and maximum value for end-to-end delay is 300 ms [15].

Priority Queuing shows by far the best results in all model configurations regardless the traffic. Priority Queuing has the best results in packet end-to-end delay and packet variation according to the given recommendations.

The achieved results show that the model in which Priority Queuing mechanism has been applied is the best QoS improving mechanism for the proposed videoconferencing network model in most of situations which are characteristic for CARNet videoconferencing system.

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Authors' addresses

Časlav Livada, dipl. ing.
University of J. J. Strossmayer
Faculty of Electrical Engineering
Kneza Trpimira 2B, 31000 Osijek, Croatia
e-mail: caslav.livada@etfos.hr

Prof. dr. sc. Drago Žagar, dipl. ing.
University of J. J. Strossmayer
Faculty of Electrical Engineering
Kneza Trpimira 2B, 31000 Osijek, Croatia
e-mail: drago.zagar@etfos.hr

Dr. sc. Josip Job, dipl. ing.
University of J. J. Strossmayer
Faculty of Electrical Engineering
Kneza Trpimira 2B, 31000 Osijek, Croatia
e-mail: josip.job@etfos.hr