

# Simulation of QoS mechanisms in tactical system STORCZYK 2010

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## Abstract

This article is a presentation of adopted solutions, simulations, analyses, and an evaluation of DiffServ mechanisms that are suggested for the STORCZYK 2010 system related to the “Quality of service guarantee method in IPv6 tactical communication system and IPv4 systems integration” project.

The first part of this article deals with the developed parameters of DiffServ mechanisms, such as division into categories of users, HTB and WRED parameters. Specific mechanisms have been parameterized in accordance with the requirements of tactical communications systems. Further on, this article presents the simulation method of verification of previously mentioned mechanisms. Additionally, this article provides the analysis of results for four sample simulations' objectives.

## Introduction

This article contains a presentation of a solution of QoS guarantee in communication system – STORCZYK 2010. The presented QoS idea was developed under the “Quality of service guarantee method in IPv6 tactical communication system and IPv4 systems integration” R & D project.

The proposed QoS mechanisms will be implemented in the STORCZYK 2010 system. This is the next generation of the presently exploited (and still under development) tactical communication system for the Polish army. The first version was based on switching channel technology with modem mode data transfer. The STORCZYK system was repeatedly re-developed. It enables the provision of more advanced services. Presently, the system has been adapted for the IPv4 protocol in best-effort mode. In the STORCZYK 2010, the use of IPv6 routers has been proposed. As the devices responsible for routing of IP packets (prepared as manufacturer's solution) do not guarantee quality of service, it was necessary to propose an effective QoS method, to be implemented within a developed system.

The QoS architecture for a tactical communication system should contain DiffServ, as well as IntServ mechanisms. In this case, it is necessary to employ the reservation and call admission control mechanisms as well as QoS routing protocol. Additionally a signaling protocol is needed for setting end-to-end connections with defined QoS parameters. The use of only DiffServ mechanisms (so called soft QoS) also enables a QoS guarantee, though within a reduced range – it depends on the amount of users and generated traffic.

Attention was devoted to supporting QoS in the data plane mechanisms. In this project, it has been assumed that the proposed statistical QoS mechanisms will be implemented in the STORCZYK 2010 system as the final effect of the project. The

proposal of a full QoS guarantee will be verified only by simulation tests using OPNET Modeler.

This article presents adopted solutions, simulations, analyses, and an evaluation of DiffServ mechanisms suggested for the STORCZYK 2010 system.

## Deployed Solutions

For the purposes of the R&D project, four main network service classes have been adopted and described as CoS, with the basic QoS parameters. They have been divided into users categories – based on the value of the DSCP field in the IP header. Each user category was assigned the appropriate percentage of bandwidth allocation on the router interfaces, by using the HTB mechanism (Hierarchical Token Bucket). A summary of the aforementioned parameters is presented in Table 1.

Traffic class	User Category	Class of network services	DSCP value	Bandwidth percentage for HTB [%]
Real Time (RT)	Cat. I	RT1	CS7 - AF43	24
	Cat. II	RT2	AF42	12
	Cat. III	RT3	AF41	4
Non Real Time-Critical (NRT-TC)	Cat. I	NRT-TC1	AF33	18
	Cat. II	NRT-TC2	AF32	9
	Cat. III	NRT-TC3	AF31	3
Non Real Time (NRT)	Cat. I	NRT1	AF23, AF13	12
	Cat. II	NRT2	AF22, AF12	6
	Cat. III	NRT3	AF21, AF11	2
Best Effort (BE)	Other	BE	DF	10

Table 1: Summary of traffic classes, DSCP values and HTB parameters

The first class of network services (RT) is designed for streaming traffic service (fixed and variable bit rate) with strict requirements with a view of ensuring low latency packet transmission, low jitter and low packet loss level. Traffic with such a profile and QoS requirements is appropriate for applications such as conversational services (telephone,

videophone) and streaming services (audio and video). In this case, traffic to the network from the user is characterized by a defined value of the peak bit rate PBR (Peak Bit Rate). This value results from the employed audio and video codecs.

Two other services (NRT-TC, NRT) are designed for flexible traffic handling, i.e., with the use of TCP. NRT-TC is designed to transmit traffic generated by short TCP connections. NRT service is designed to send traffic with long TCP connections, where the source is greedy. In both cases, traffic is characterized by the value of PBR and SBR (Sustainable Bit Rate). These values should be assigned to the relevant service user.

The last service without QoS corresponds to the "best effort" transfer of packets.

As a distinction between four data streams in the network would be insufficient for ensuring the proper operation of real-time services (for users with different service priorities), it was decided to conduct an additional division into three categories of users for the following classes: RT, NRT and NRT-TC. The main role of this qualification is to link the types of users with classes of network traffic. The first category is for the key users, the second category is for medium-priority subscribers, while the third category applies to other communication system users.

The queuing mechanism proposed for use in the tactical communication system assumes (Figure 1):

- The creation of four HTB queuing groups for 4 classes of traffic: RT, NRT-TC, NRT and BE.
- In each group (except the BE class) – the creation of three queues for three user categories with different priorities: user A (highest), user B (medium), user C (lowest).
- Prioritizing groups in accordance with different bandwidth allocation on the router's output interface: RT - a minimum of 40% of the bandwidth on the interface, NRT-TC - a minimum of 30% of the bandwidth on the interface, NRT - a minimum of 20% of the bandwidth on the interface, BE - a minimum of 10% of the bandwidth of the interface. Within groups, user prioritizing in accordance with available bandwidth allocation for the whole group: user A - 60% of the bandwidth available for the whole group, user B - 30% of the group bandwidth, user C - 10% of the group bandwidth (Figure 1 - marked as "rate").
- The ability to borrow bandwidth between types of users within a given traffic class.
- Creating another level of the hierarchy by integration of groups - the possibility of borrowing bandwidth between groups (traffic classes). In Figure 1, each group is marked with the same color.
- Determining rates of upper limits of occupying nominal bandwidth on the output router interface: RT group - up to 95% of the bandwidth interface, the remaining groups up to 90% of the interface bandwidth (Figure 1 - marked as "ceil").
- Determining the priority of access to a free bandwidth for each group: RT - Priority 1 (highest), NRT-TC - Priority 2,

NRT - priority 3, BE - Priority 4 (Figure 1 - marked as "prio").

- A minimum of 10% of the bandwidth of the interface has been offered to the BE class, due to a lack of division on users within this group.

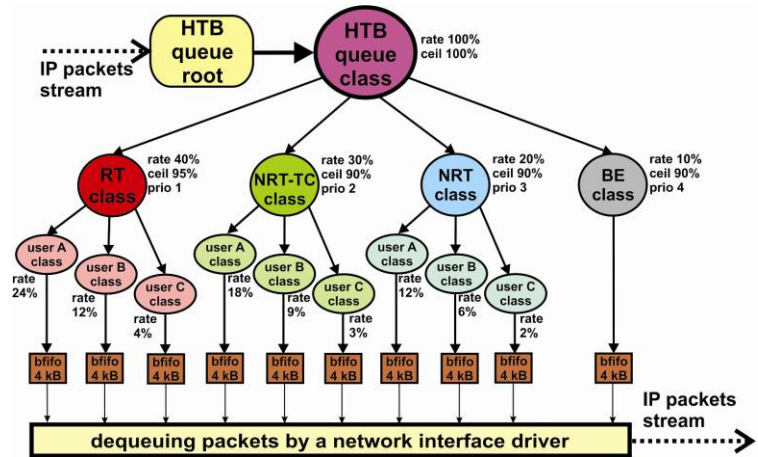


Figure 1: HTB architecture

Additionally, during project implementation, the employment of congestion avoidance mechanisms (WRED) and traffic shaping has been proposed.

The congestion avoidance mechanism, by influencing the process of receiving packets, which depends on the degree of the output queues fulfillment, allows a controlled packet loss and to some extent, the shaping of incoming traffic into an IP router.

For the WRED mechanism, it was decided to assume the following parameters:

- the maximum packet loss level (PLML) for RT, NRT and NRT-TC services is  $10^{-3}$ , and 1 for BE service ;
- the maximum threshold (Lmax) for each service packets (RT, NRT-TC, NRT) will depend on the speed of the interface and for the BE services, it will constitute 90% of the output buffer;
- the minimum threshold (Lmin) for BE and NRT services packets will constitute 10% of the output buffer, and for RT service and NRT-TC, it will be 90% of the maximum threshold Lmax (RT, NRT-TC).

Traffic shaping mechanisms were verified in the real network, and therefore, they shall not be presented in this paper.

### The Objectives of Simulations

Simulation tests should address these issues:

- The ability to differentiate an approach to network services;
- The accuracy of network services maintenance differentiation for compliance with the premises (24% of bandwidth for RT1, etc.);
- The function of QoS, if there is too much traffic in the class;
- The influence of the telecommunication traffic source position in the network graph on the effect of competition for bandwidth access in the class of network services;

- The influence of the proposed WRED mechanism on the improvement of the stability of the user service in the event of network congestion;
- The influence of long communication chains on QoS parameters;
- The influence of QoS mechanisms on an oversized network.

### The Simulation Model and Its Limitations

Simulation models were prepared within the network simulation tool OPNET 12.0, and therefore the following simulation simplifications were required:

- There is no HTB queuing mechanism model in the OPNET network simulation tool. The HTB mechanism was replaced with a Modified Deficit Round Robin mechanism (MDRR), which is functionally very similar to the HTB queue.
- Analysis of network traffic in one direction: from the source (client stations) to the receiver (server stations). The purpose of this limitation is to organise and establish full control over input enforcements in the simulation model during the test.

Four network models were created:

- a two-node network model;
- a four-node network model;
- a seven-node network model;
- an eight-node network model (Figure 2).

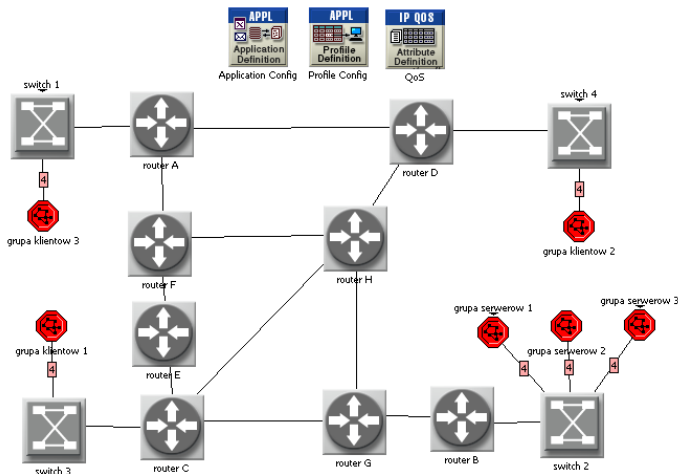


Figure 2: Eight-node network model

There are four different scenarios for each model:

- speed of point-to-point connections between the routers is set at E1, no QoS mechanisms were set;
- speed of point to point connections between the routers is set at E1, and QoS mechanisms on interfaces connecting the routers together were set (marking, queuing and WRED);
- PPP links between routers were set at 100 Mbps without QoS;
- PPP links between routers were set at 100 Mbps with QoS.

The duration of the simulation was set at 3600 sec.

### Results of the Analysis

*The ability to differentiate an approach to network services and accuracy of network services maintenance differentiation for compliance with the assumptions.*

As an example, the time course of telecommunication traffic sent by an edge router in four-node network has been presented (Figure 3). This router is responsible for quality service diversification of all data sources delivered to an edge router from all traffic sources existing within network. In Figure 3, we can clearly distinguish 10 separate sources of telecommunication traffic corresponding to 10 classes of network services. In Figure 3, we can also see that each traffic class is served through a given individual quality resulting from the allocation of different amounts of transition resources in inter-node link. The volume of allocated resources is consistent with the accepted premises, the greatest amount of bandwidth is assigned for RT1 traffic (24% of 2 Mb/s), and smaller amounts for RT2 and NRT1 (12% of 2Mb/s).

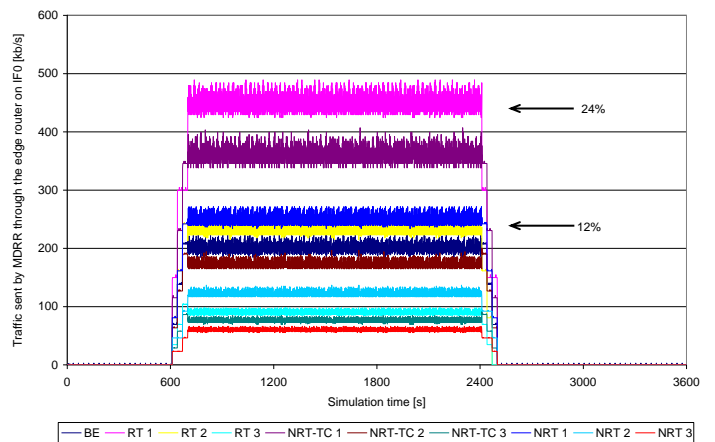


Figure 3: Traffic sent by MDRR through the edge router on IFO in four-node network

*QoS functioning, where there is too much traffic in the network services class.*

Figure 4 presents distinctions in access to transition resources in a four-node network with and without QoS in the event of a progressive overload of selected RT2 network service classes.

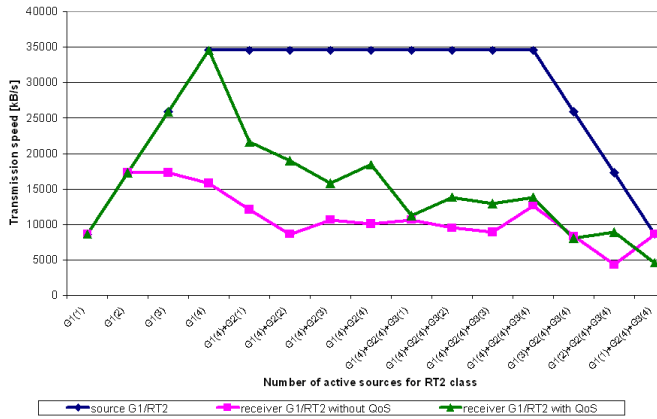
Characteristics:

- G1, G2, G3 – group of telecommunication traffic sources, respectively No. 1, No. 2 and No. 3 – each group was generating a maximum 2.4 Mb/s data stream (Figure 2 shows the locations of the three groups of client - server relationships);
- Number in brackets (1), (2), (3), (4) – number of active users within a defined group of customers;
- Example: G1(3) – means that within customer group No. 1, data are transferred to 3 users at the same time; while G1(4)+G2(4)+G3(4) – means that in all client groups, data are transferred simultaneously by all users.

The chart (Figure 4) illustrates a case of an increasing traffic load of dedicated network service class. The transmission speed

is gradually decreasing for the observed client – server relationship. (In this case, this is group No. 1.) However, at least a minimal bandwidth in networks with QoS is always preserved. In the case of networks without QoS mechanisms, in some circumstances, the transmission was fully blocked within the client–server relationship.

The application of a QoS mechanism allows the continuity of data transmission, even in highly overloaded networks.



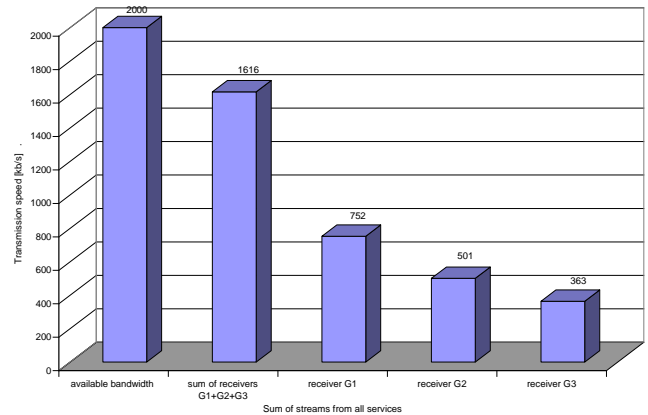
**Figure 4: Number of active sources for RT2 in four-node network with and without QoS**

QoS mechanisms are assigned for “traffic order” in accordance with the network administrator’s expectations. This is done by assigning a greater bandwidth to selected classes of network services at the expenses of other classes.

*The influence of telecommunication traffic source position in the network graph on the effect of competition for bandwidth access within the class of network services.*

In the ideal case, each source of telecommunication traffic attached anywhere within the network should have the same chance to send its data to the receiver. The results of simulation tests have partially confirmed these premises. However, they have also demonstrated the existence of another problem, depending on the available bandwidth from the start of the traffic sources in comparison to other sources of telecommunication traffic.

Figure 5 illustrates that the largest volume of bandwidth has been received by the group of traffic source under No. 1 (G1) and the lowest bandwidth, No. 3 (G3) – all groups were sending data with maximum performance.



**Figure 5: Sum of streams for all services in four-node network**

Source group (clients) No. 1 (G1) is located the closest to its “exit gates” (servers group No. 1); theoretically, the network placement graph appears to imply that it is a favored group. However, of source groups Nos. 2 (G2) and 3 (G3) are at identical distances from their “exit gates” server (groups Nos. 2 and 3), and in theory, they should obtain the same access to the bandwidth. The only difference between source groups Nos. 2 and 3 is the commencement time of sending traffic. Source group No. 2 begins sending 600 seconds earlier than group No. 3.

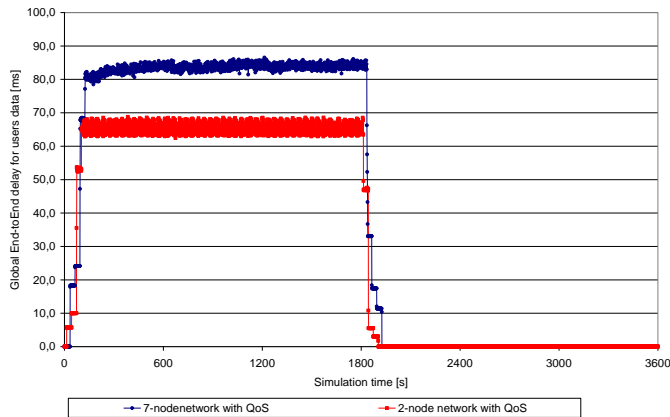
The simulation tests demonstrated that, for example, in the four-node network with QoS, traffic source group No. 2 upon finishing work by traffic source group No. 1, takes over the dominant role of group 1 in gaining access to transmission resources, similar to group No. 1. This is not justified by the position of group No. 2 in the network, but only by the commencement time of work by this group of traffic sources relative to group No. 3.

The phenomenon may stem from the fact that a group of traffic sources (No. 2) transmitting earlier than another group (No. 3) has established its transmission sessions in the client-server relationship. This results in a “reserving” in an informal way the transmission resources in the different classes of network services. Subsequently, another traffic source commencing work with some delay may have a problem entering already overloaded transmission resources. The analysis confirmed that access to transmission resources in the various classes of network services is not dependent on the location of the traffic source within the network graph. However, it does depend on actual network congestion by traffic sources, which had commenced work.

*Evaluation of the influence of long communication chains on various quality parameters of the transmitted data.*

Figure 6 presents a comparison of the global end-to-end delay for a two- and seven-node network with QoS mechanisms enabled. The Figure illustrates that despite an increase of the number of routers to 7, delay has increased only slightly (by about 20 ms). In these networks, a node that has a major influence on delay is the first router in which sent packets meet

the traffic shaping MDRR queue. Other routers only introduce a relatively small delay associated with packet processing time on the router.



**Figure 6: Global End-to-End delay in two and seven-node network with QoS**

## Conclusion

The tests and analysis of the obtained results allow to draw the following conclusions:

- The queuing mechanism provides a differentiation and traffic limits within the class, according to plan. The concept of QoS mechanisms based only on the data plane does not provide a reservation of transmission resources for specific data streams.
- The size of the lost traffic does not depend on the location of the telecommunication traffic source within the network topology, but rather on the time of transmission commencement of the telecommunication traffic source. It would be desirable to develop a mechanism for ensuring even access to transmission resources in a particular class of network services to all traffic sources regardless of the time of transmission commencement.
- QoS mechanisms do not have a negative influence on the quality parameters of data streams sent in oversized networks, up to the moment when the network load will not exceed 60%.
- The proposed QoS mechanisms affect the delay growth in all classes of network services in an identical manner.
- The last phase of this project was the drafting of a recommendation covering data plane mechanisms and control plane mechanisms.

During the project, apart from studies, which have been presented in this article, there have been also carried out other tests such as: WRED mechanisms tests or the impact of DiffServ mechanisms on delay and jitter. The additional research draws the following conclusions:

- The proposed WRED mechanism reduces excessive traffic, just as HTB, and therefore, one should consider the validity of employing both of these mechanisms simultaneously.
- The corrective influence of higher-layer protocols on jitter for real-time services was observed and analysed.

The data plane mechanisms were implemented in STORCZYK system routers and verified in practice. Satisfactory results have been obtained.

## References

- [1] Cisco, Implementing Quality of Service Policies with DSCP, doc. ID: 10103.
- [2] RFC 2475, An Architecture for Differentiated Services.
- [3] RFC 2597, Assured Forwarding PHB Group.
- [4] Y.1291, An architectural framework for support of Quality of Service in packet networks, ITU-T Recommendation.
- [5] Kącik S., Michalski M., Zubel K., „Metoda QoS płaszczyzny danych w specjalnych systemach łączności”, Krajowe Sympozjum Telekomunikacji i Teleinformatyki, Wrocław, Poland, September 2010.
- [6] Strzelczyk K., „Koncepcja gwarantowania jakości usług w taktycznych sieciach IP” PBR nr 0R00002406 – zad. nr 9, nr arch. WIŁ 597/2009/PBR, Zegrze, Poland.
- [7] Kącik S., „Opracowanie modelu kolejowania pakietów w taktycznym systemie łączności wykorzystującym technikę IPv6” PBR nr 0R00002406 – zad. nr 14, nr arch. WIŁ 130/2010/PBR, Zegrze, Poland.
- [8] Zubel K., „Analiza wyników badań symulacyjnych mechanizmów płaszczyzny danych oraz opracowanie wniosków z badań symulacyjnych” ” PBR nr 0R00002406 – zad. nr 16.1, nr arch. WIŁ 301/2010/PBR, Zegrze, Poland.
- [9] Zespół pracowników TRANSBIT, „Specyfikacja wymagań militarnych dotyczących zapewnienia gwarancji jakości usług w taktycznych sieciach IP” PBR nr 0R00002406 – zad. nr 4, nr arch. WIŁ 237/2009/PBR, Warsaw, Poland.