Services for Advanced Communication Networks

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Abstract: Services for Advanced Communication Networks are described in this paper. An Artificial Neural Network will be used for the control of Network Elements, in our case a Switch. Classical sequence data processing is substituted by parallel processing. The classical switch and the switch controlled by a Kohenen Artificial Neural Network are compared. The switch has four ports and its own Quality of Service (QoS) control. Services such as FTP, HTTP, VoIP and QoS mechanisms without QoS, Best Effort QoS, FIFO QoS, Integrated Services (IntServ) and Differentiated Services (DiffServ) are simulated and compared. The level of priority contains the overhead of IP packet as the Type of Service (ToS) or also as Differentiated Services Code Point (DSCP). The Opnet Modeler programme is used for the simulation.

Key-words: Artificial Neural Network, ANN, Network Element, NE, Converged Network, Delay, Jitter

1 Introduction

An Artificial Neural Network (ANN) will be used for the control of Network Elements (NE), in our case a Switch (SW). The classical sequence data processing is substituted by parallel processing [2,3]. This paper starts from [1] and points out the main results from this reference. The extended conference contribution [27] was nominated for this journal usage.

An artificial neuron is a fundamental part of every ANN. The network contains a known number of these neurons. The number of neurons is given by the number of inputs and outputs of ANN topology in the course of ANN mathematical model [2,16]. One neuron contains a limited number of inputs, while outputs are multiplexed and all carry the same value.

ANN are suitable for the converged communication network [2,3]. This network provides user services, especially multimedia services in real time such as videoconferences, Voice over Internet protocol (VoIP) and others.

The most frequent development of the Hopfield network is the optimization problem solution by iteration process [1]. It is a problem solution via sequential repetition of the algorithm and sequential approximation to a result. The Hopfield ANN minimizes the power function of ANN [5,6,13].

The Hopfield networks need a large memory; they contain buffers and control units [7]. The main advantage of the Hopfield networks is the beginning stability of output vector. The main disadvantage is the high number of connections, frequent network overflow and moderate instability.

The Kohenen ANN is formed by one layer of input neurons and a second layer of Kohenen neurons, all of which are interconnected with one another. [1]. The connections of all input neurons come to every Kohenen neuron and for every neuron a vector of synaptic weigh**t** is given. The inputs are multiplexed by synaptic weigh**T**s and the weight space is identical to the input space [7,13,19].

The main difference between the Kohenen and the Hopfield network is in the learning phase. The Kohenen network learns all time, in its traffic. The Hopfield network has first the learning phase and then the using phase.

2 Problem Formulation

A Hopfield ANN is represented as a continuous or discrete system [2,3]. Every neuron is connected with all the other neurons and these connections are symmetrical [14, 19]. Converged networks represent the convergence of telecommunication and computer networks. Classical sequence data processing contains the central processor unit (CPU). When the CPU buffer is overcharged, packets are thrown away, and data loss and delay increase. Parallel data processing is an alternative

[2,3,14] and using a neural network is suitable for it [1].

Network learning is the first step for the Hopfield or the Kohenen ANN. The learning can be initial or gradual, both must solve the optimization task [14,16] fast and effectively. The number of valid solutions increases exponentially with problem. Then the time for optimal solution increases also exponentially [1]. For ANN, optimization is a stochastic process and optimal solution need not be found.

The basic communication protocols such as the Internet Protocol IP and Ethernet protocol do not contain any arrangement for the control of priority reservation in communication channels with the FIFO (First In First Out) queue. The Network Element NE itself must set the priority as a solution to this problem. The active NE controlled by ANN must therefore set itself packet priority [1]. This priority is set by the time point when the packet was received on the input port, by the packet size and type, and by groups of packets streams [2,14]. From these parameters ANN sets whether the packet will be sent out of order (services VoIP) or whether a delay is possible (services FTP - File Transfer Protocol). This arrangement influences the Quality of Service (QoS).

The main function of the switch is to send data from the input port to the output port. If data from more input ports are sent to one output port, the blocking of the network and the ports could be a problem. Therefore every data stream has its own output [2].

The Kohenen network seems to be a good solution to the above problem.

3 The main model of the switch controlled by ANN

The demands on elements in converged networks steadily increase. Elements such as switches must be equally fast for data transmission, together with QoS cover for all connected terminals [1].

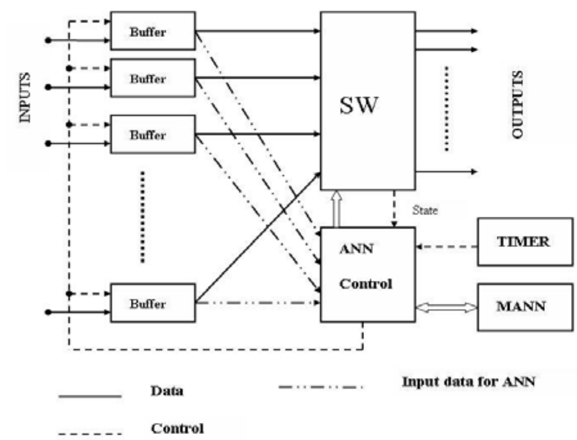
The demands increase even more with higher data transmission using the IPv6 protocol and IEEE 802.3ae and IEEE 802.3z standards. These standards have higher data streams in comparison with IEEE 802.3a to IEEE 802.3y 10/100Mbit/s [2,14,16] .

The basic [switch model controlled by ANN] is in Fig.1 [1,2,14,16]. Data from input ports are immediately sent to input buffers. Input buffers have two functions:

a) sending packet parameters to the ANN block. Overhead from packets IPv4 and IPv6 are sent to the ANN block, where it is decided which of the packets have a higher priority.

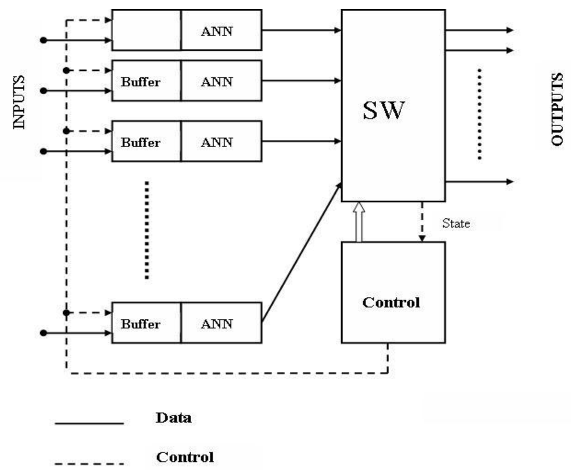
b) sending data to the switch. Data from IP packets are sent from the buffers directly to the switch area. Paths in the switching area are activated to the output port by ANN control.

Switch outputs can for better traffic contain also buffers.



ANN – Artificial Neural Network, SW – Switch, MANN – Memory ANN
Fig.1 Basic switch model controlled by ANN

For the next switch simulation by Opnet Modeler, Fig.1 was transformed into Fig.2 [1]. This transformation is important for the program environment for changing the model of classical switch to a model of ANN-controlled switch. ANN control is separated into single ports because otherwise the calculation is overly slow-paced.



ANN – Artificial Neural Network, SW – Switch
Fig.2 The switch model controlled by ANN in

Opnet Modeler program

4 Opnet Modeler simulation

We compare the classical switch and the switch controlled by ANN [1]. The switch has four ports and its own QoS control. The Kohenen ANN add their own symbols DSCP (Differentiated Services Code Point) to IP packets. Allocation of these symbols is separate for every port. Thus selected services are preferred. Control topology works by Fig.2 [1]. The Kohenen ANN was selected because the calculation time and error number are minimized for the Kohenen network [13,16].

Topology created for the comparison of two switches is in Fig.3 [1]. The switches are controlled either classically or by ANN.

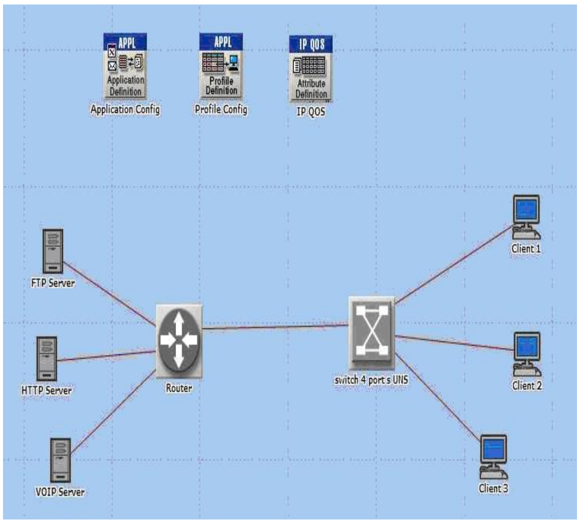


Fig.3 Opnet Modeler topology

All equipment connections are set at a rate of 1 Gbit/s (1000BaseT). Services load the network by these parameters:
- FTP (File Transfer Protocol) – downloading of files of up to 4 MB
- HTTP (HyperText Transfer Protocol) – browsing through internet pages on a HTTP Server and displaying text pages (up to 30 kB)
- VoIP – connection by the H.323 protocol and G.729a codec, loaded by connection with VoIP Server and clients 1 to 3 with one another.

In a network with ANN the QoS is introduced via DSCP marking on the switch. In a network with the classical switch, the QoS is introduced also using the DSCP but on the router.
The following rules are used for DSCP marking:
FTP AF11 – class 1 (AF – Assured Forwarding) – occasional removal of packets
HTTP AF21 -class 2 (AF – Assured Forwarding) – occasional removal of packets
VOIP EF-class 5(EF – Express Forwarding) emphasis on low delay.
The classical switch and the switch with ANN are compared as regards:
Speed of the FTP, HTTP, VoIP services
Total network delay
Delay on the switch ports
Loading of individual buffers
Comparison for the HTTP services is in Fig.4 [1]. The HTTP service has a higher growth rate for the switch with ANN. It is given by the smaller limiting of HTTP service on the switch ports.

lower – classical, upper – with ANN

Fig.4 HTTP network transmission

In this network with QoS the FTP service is in the last place as to the transmission preference. The average service curve for the whole network is in Fig.5 [1]. The switch with ANN has a three times higher transfer than the classical switch. This service is not limited.

Another important parameter is the network packet delay, we can see it in Fig. 6 [1].

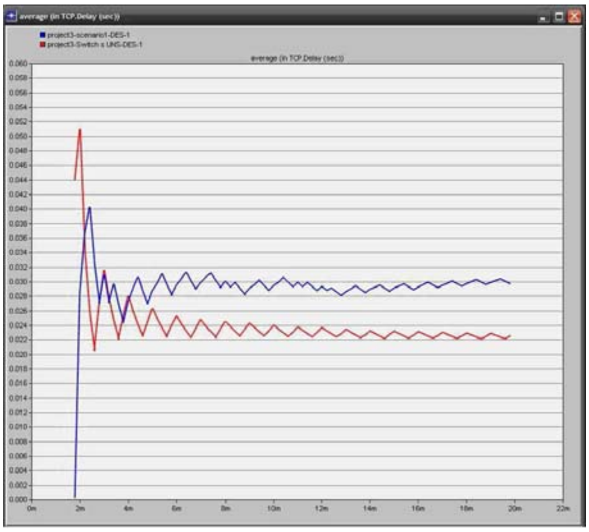
TCP packets in this network form the highest representation. The HTTP and FTP services make full use of them, VoIP only partial use. VoIP uses TCP packets for telephone signalling, selecting subscriber availability and control of network transport. Initial value for the switch with ANN is high, after state fixation it drops to 24 ms.

For VoIP service, not only the network delay but also the jitter are important. The jitter for the switch with ANN is half the jitter for the classical switch. Sometimes the switch with ANN can be without jitter.

lower – classical, upper – with ANN

Fig. 5 FTP network transmission

.Another parameter is the delay on the individual ports. For the switch with ANN there is no delay on the individual ports; the classical switch has some small delay.



lower – with ANN, upper – classical

Fig.6 TCP delay

The last parameter is the usage of input buffers. ANN loads the buffers either roughly in the same or slightly higher extent than the classical switch. For the traffic it has no negative influence.

5 Extended network for QoS simulation

We want to simulate QoS in the extended network as shown in Fig. 7. The Opnet Modeler programme will be used again. The designed network will be loaded by the VoIP, HTTP and FTP services.

The network consists of four nodes – routers in four towns (Praha, Brno, Pardubice, Tábor). These routers are connected by 1Gbit/s (1000BaseT) links and by one alternative path, also 1 Gbit/s (1000BaseT). Each node - town contains a sub-network with one switch and with 25 LAN clients connected by a 100 Mbit/s (100BaseT) link.

The routers contain 4 Ethernet ports and they have a throughput of up to 1Gbit/s. The switches, which are placed separately, have 8 Ethernet ports and a throughput also of up to 1Gbit/s. The switches which are placed in LAN sub-networks have 25 Ethernet ports and a throughput of up to 100 Mbit/s.

The given services load the network by these parameters:

FTP – download of files up to 8 MB in size,
HTTP – continuous browsing of internet pages with large pictures, up to 20 pictures/page (up to 200 kB / 1 client LAN),
VoIP – connection by the H.323 protocol and G.729a codec, loaded by the connection to VoIP Server and LAN networks LAN with on another.

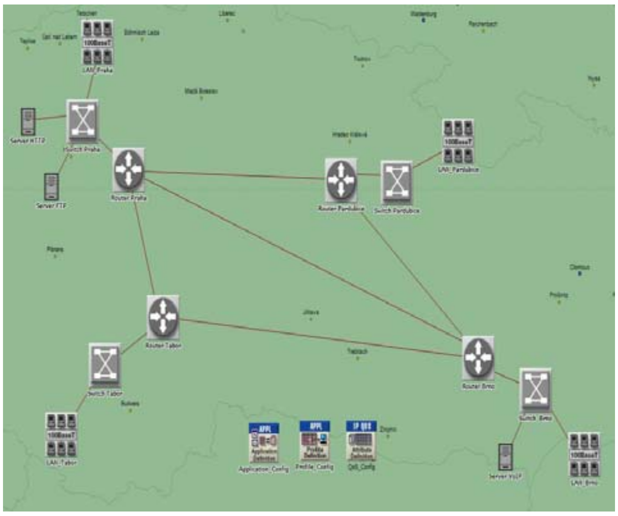


Fig. 7 Extended Network for QoS simulation

The first simulation proceeds without setting the QoS. All services are initially set to the Best Effort quality by the ToS parameter in the headers of transmitted IP protocols. For a higher loading, however, this network does not prefer any among the services. It is possible that some of the services are totally restricted and their traffic is not practicable.

Services as FTP and HTTP can overload the transmission bandwidth of a channel. The VoIP service, which does not carry such a big load, needs a small network delay and a connection without drop-out, and can thus be totally suppressed so that it is not applicable.

The second simulation is set to QoS support. Packets are again marked by the ToS parameter. The following levels of service were given to individual services:

FTP –Background – the network running on the background, priority rating 1,
HTTP – Standard – priority rating 2,
VoIP – Real voice and video (Interactive multimedia), priority rating 5, this service must in addition guarantee delay and jitter below 100 ms.

5.1 QoS Parameters

Four basic QoS parameters are defined as bandwidth, loss, delay, and jitter. These parameters influence the Voice Quality (VQ), see Table 1 [4, 10]

Parameter	Good VQ
Delay	0 – 150 ms
Jitter	0 – 20 ms
Loss	0. 0,5 %

Parameter	SatisfactoryVQ
Delay	150 – 300 ms
Jitter	20 – 50 ms
Loss	0,5 – 1,5 %

Parameter	Not Satisfactory VQ
Delay	over 300 ms
Jitter	over 50 ms
Loss	over 1,5 %

Table 1 Voice Quality and network parameters

Bandwidth is one of the main aspects for the transmission in data networks. It represents a data volume which a transmission channel is able to transport in a certain time. The parameter for the bandwidth is the bit rate in bits per second. The bit rate is given not only by the character of transmission medium but also by the character of technical equipment in the transmission channel. The technical equipment in this channel (modem, switch and so on) can make the total channel transmission slower [4, 10, 11].

Delay is the time which is necessary for data transmission from a transmitting terminal to a receiving terminal. The delay is given in milliseconds (ms). We know total and one-way delays. One-way delay contains:

- packet delay; it is the time of conversion from analogue to digital signal, frame ordering and back transmission.

- propagation delay; it is the time for data transmission from the link input to the link output. It is given by the speed of signal propagation through the transmission medium

- delay of buffer jitter; it originates on the receiver side in the course of storing a datagram in a buffer. The receiver tries to set the delay constant.

Jitter is given by the packet loss in a network in milliseconds (ms). In a network there is a lot of devices which slow down datagrams; also, because of network drop-out an alternative path for datagram transmission must be used. The receiver then receives the datagrams in a different sequence than they were sent. When the buffer capacity is not sufficient, the datagram can be rejected.

Loss is defined as the ratio of transmitted datagrams and error-free received datagrams [10]; it is expressed in (%).

The ITU-T Rec.X.902 standard deals with the quality of service in the telecommunication area. This recommendation contains definitions of terms and an analytical frame for a standardized description of distributed systems. It forms the basis for the ITU-T Rec.X.903 standard [8, 11].

An IP header holds the Type of Service (ToS) and it represents priority for the packet transmission through the network. ToS uses the mechanism of DiffServ [12].

Priority	Transmission
0	Best Effort
1	Background
2	Standard
3	Excellent Effort

4	Streaming multimedia
5	Interactive multimedia
6	Interactive voice
7	Reserved

Table 2 Priority of ToS

5.2 Simulation results

In Fig. 8 we can see the communication of VoIP service in the network with QoS and without QoS(i.e. only by Best Effort). The VoIP service is not practised in the network without QoS, and with increasing demands on FTP and HTTP transmission it is refused altogether.. When it is preferred to other services, it makes no problem in the network and the number of VoIP subscribers can continuously increase. In Fig.9 we can see the average delay of VoIP service in the network with QoS (lower) and without QoS (higher)

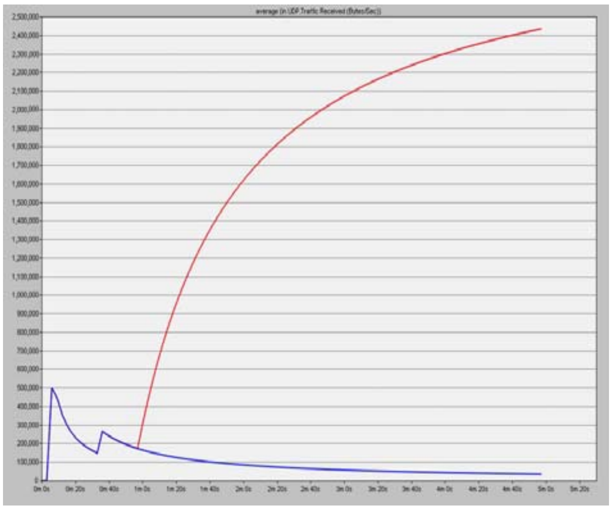


Fig. 8 Communication of VoIP service in the network with QoS (higher) and without QoS (lower)

Another parameter for high-quality communication in the VoIP network is jitter. The result with and without QoS can be seen in Fig.10. We can see a pronounced jitter at the beginning of all services.

Our next simulations were the average behaviour of subjective MOS (Mean Opinion Score) parameter – see Fig.11. Opnet Modeler can estimate this parameter in the network and it shows the transmission quality from the listener's viewpoint (interference, obscurity, echo, and so on).

The G729a codec is economical as to the bandwidth, but it is not very good as to the MOS parameter. VoIP communication with QoS is better than HTTP and FTP.

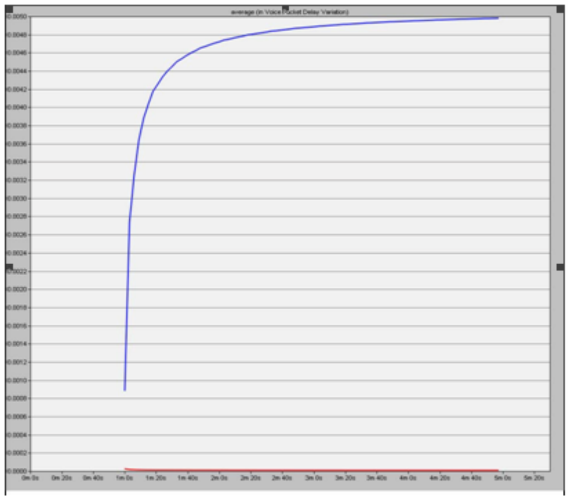


Fig.9 Average delay of VoIP service in the network with QoS (lower) and without QoS (higher)

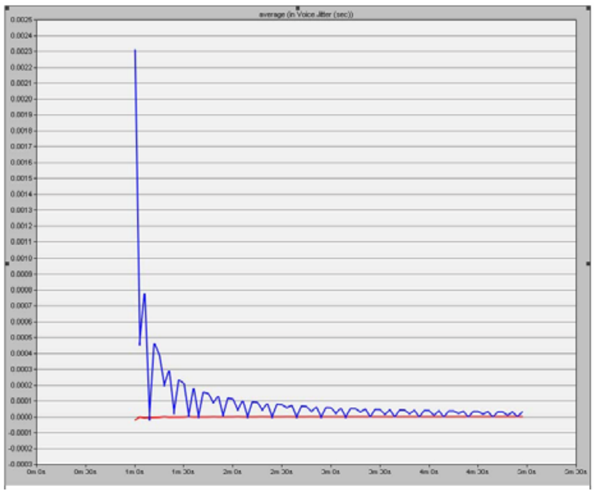


Fig.10 Jitter for transmission of VoIP communication with QoS (lower) and without QoS (higher)

With QoS, the VoIP communication improves. The result is immediate limiting of HTTP transmission and successive limiting of FTP transmission. FTP transmission has a lower priority level in the network than HTTP, but the transfer of data has a higher level. Limiting the FTP and HTTP transmissions with respect to the increasing of VoIP transmission is in Figs 12 and 13.

6 Comparison of mechanisms QoS

The basic mechanisms of QoS in the Ethernet network are:

- Best Effort Service is a method trying to transmit every packet to its destination as effectively and quickly as possible (for instance FIFO QoS). The QoS level is set to 0.

- Integrated Service (IntServ) is a technology based on a bandwidth reservation that is set at the moment of establishing connection. A disadvantage is that this process occupies a bandwidth which it need not require at the given moment while other services are limited by this.

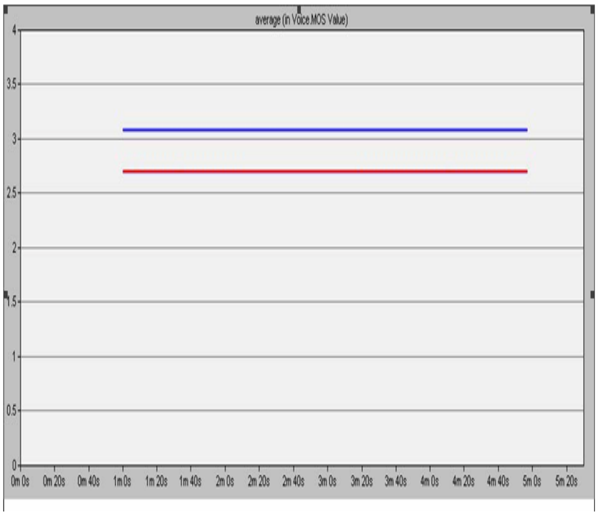


Fig.11 Examination of MOS quality, with QoS (lower), without QoS (higher)

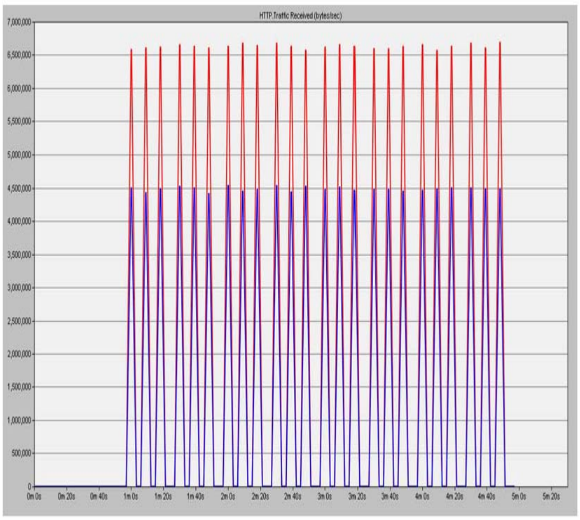


Fig.12 HTTP transmission in the network with QoS (lower), without QoS (higher)

- Differentiated Service (DiffServ): packets are divided into categories by pre-defined parameters. Every packet has its level of priority set and, according to this parameter, the packet is preferentially treated when passing through the network. This level of priority contains the IP header of the packet as the Type of Service (ToS) or also as the Differentiated Services Code Point (DSCP). DiffServ does not reserve the bandwidth reservation itself, the bandwidth is not occupied when it is not used. In Opnet Modeler, ToS-based Priority Queuing and DSCP-based Priority Queuing are examples of the mechanism.

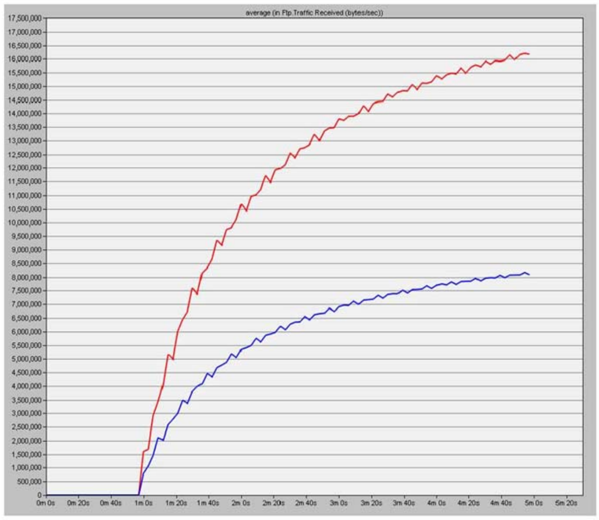


Fig.13 FTTP transmission in the network with QoS (lower), without QoS (higher)

QoS need not be realized when not all of the connectivity is shared or when the network capacity is over-dimensioned. QoS is also very often set dynamically when the network is overloaded and some important services are limited. The bottleneck for QoS is between the transport and the access networks and between the access and the private networks.

The mechanisms of QoS are compared for the network in Fig.14. LAN networks are connected to Switches by 100BaseT links. All other connections are realized by 1000BaseX links. The main elements of our network are connected by 1Gb/s cables while the LAN networks are connected by 100 Mbit/s links. The bandwidth of this network is not sufficient for use by multimedia and therefore the deployment of QoS is important. Each LAN has 10 clients

We simulate the FTP (File Transfer protocol), HTTP (HyperText Transfer Protocol) and VoIP (Voice over Internet Protocol) applications. We use 4 scenarios: “without QoS”, FIFO QoS (First In First Out Quality of Service), DSCP QoS (Differentiated Services Code Point Quality of Service) and ToS QoS (Type of Service Quality of

Service).

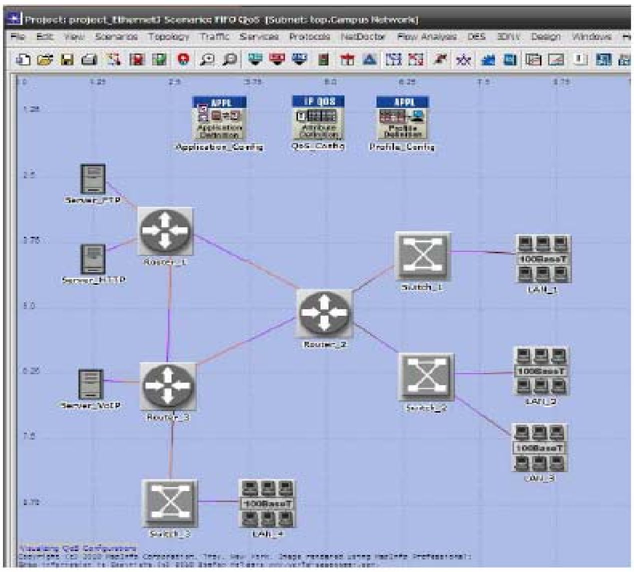


Fig.14 Network Topology

In Figs 15 and 16, we can see the total transport of the HTTP and FTP services in our network. None of the QoS mechanisms is outstanding here. The FTP transport was the highest for the FIFO QoS and lowest for the scenario without QoS. FTP has the lowest priority in the network; no increase in the bit rate is required.

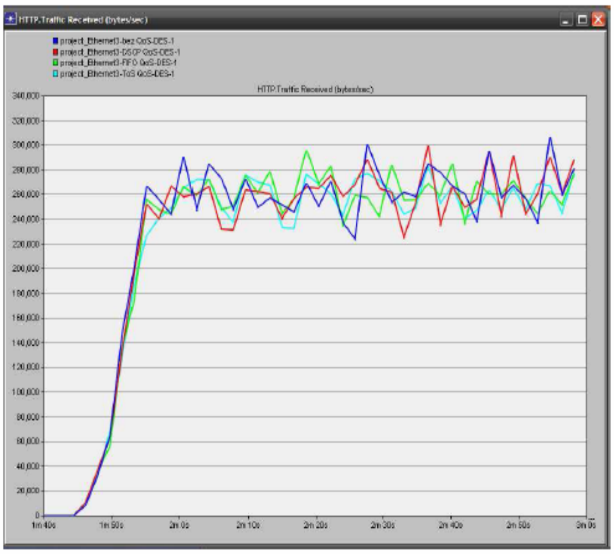


Fig.15 Total transport of HTTP services

full low – ToS QoS, mid-low - DSCP QoS, mid-high – FIFO QoS, full high – without QoS

TCP packets are markedly delayed in Fig.17 for the mechanism of FIFO QoS. The total delay in the Ethernet network in Fig.18 was observable first of all in the scenario without QoS.

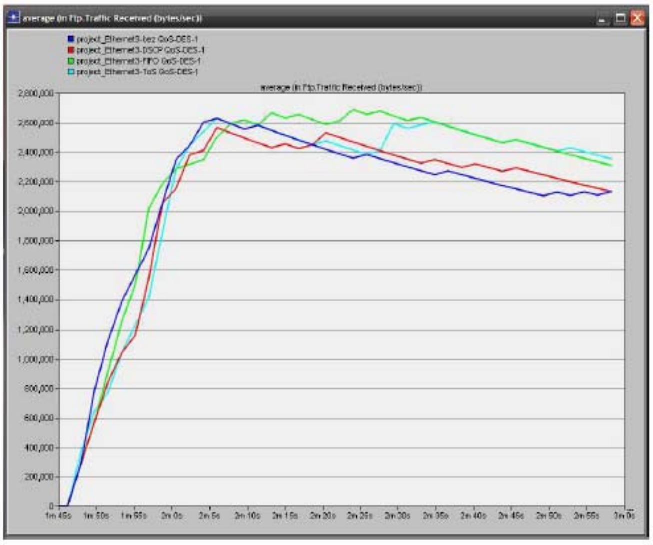


Fig.16 Total services transport FTP

full low – without QoS, mid-low - DSCP QoS, mid-high – ToS QoS, full high – FIFO QoS

The total delay jitter in the network for VoIP service is in Fig.19; the most pronounced value is for FIFO QoS.

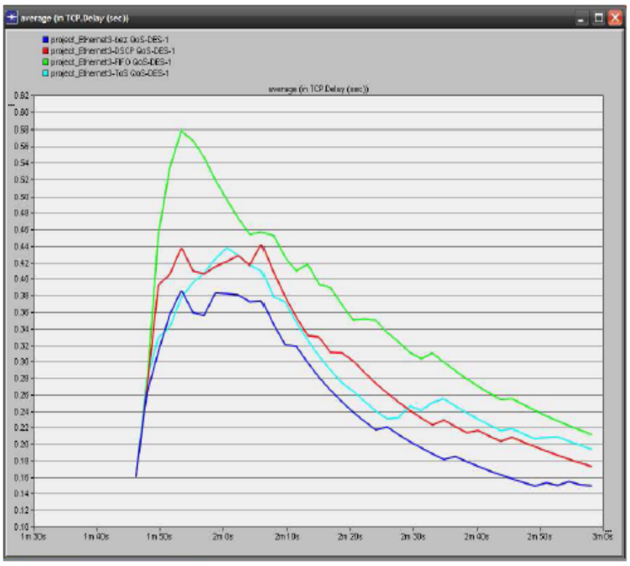


Fig.17 Total TCP packet delay in the network

full low – without QoS, mid-low - ToS QoS, mid-high – DSCP QoS, full high – FIFO QoS

The mechanism of ToS QoS appears to be the most suitable for the simulated network according to Fig.14

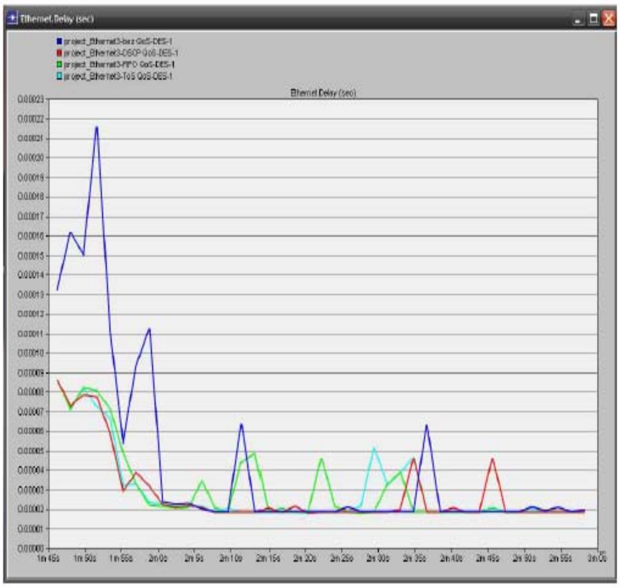


Fig.18 Total delay in Ethernet Network

full low – DSCP QoS, mid-low - ToS QoS, mid-high – FIFO QoS, full high – without QoS

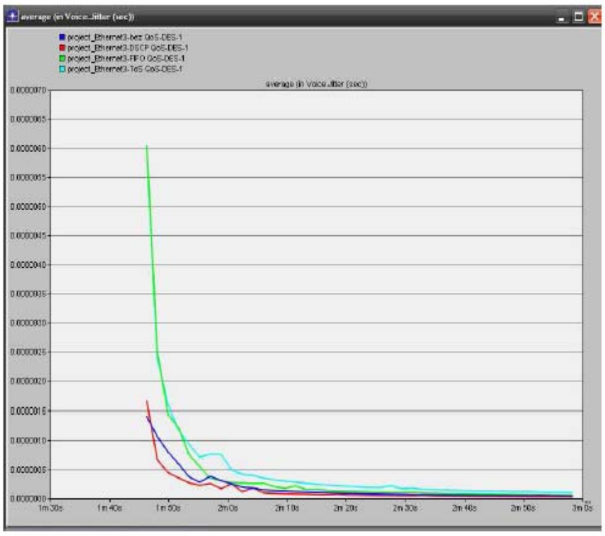


Fig. 19 Total jitter delay in the network for VoIP service

full low – DSCP QoS, mid-low - without QoS, mid-high – FIFO QoS, full high – ToS QoS

7 Conclusion

Using the Opnet Modeler we simulated and compared the classical switch and the switch controlled by ANN. The switch with ANN can be used and has better results for the HTTP, VoIP and FTP services. The bandwidth is without problems for both switches, the delay and jitter are better for the ANN-controlled switch.

The switch with ANN has a higher demand on buffers, but it is not critical. As a conclusion it can be claimed that the ANN-controlled switch is better than the classical switch.

Fig.21 Transmission of FTP service in LAN_4 network

full low – FIFO QoS, mid-low - without QoS, mid-high – DSCP QoS, full high – ToS QoS

The basic behaviour of network elements in the overloaded network and in the network with a lot of demanding services was demonstrated.

Our simulation confirms that the VoIP communication is on a good level when the parameters the QoS mechanism are used in the network. This is necessary for communication by VoIP in an overloaded network.

The first step for the voice optimization of the VoIP communication is to guarantee the QoS mechanism in the network. The QoS mechanism can be triggered as soon as the network is overloaded. Then the other services are not unnecessarily limited. When we use this solution, correct criteria for setting the QoS must be given.

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