Adaptive quality of services management over IP network

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Abstract: - The Internet with multifunctional and variety of structures and dynamics of development is used to provide complex multiple telecommunication services. The biggest shortage of the traditional "best effort" model is that it does not guarantee the quality of services demands, especially for the interactive real time services. In order to ensure the necessary quality of the service (QoS) it must be reacted adequately into the dynamic changes of the load in the network, i.e. the appropriate recourses of the network and the quality of the service must be ensured. In this paper we are proposing the new adaptive (AFQ) scheduling for QoS management model, reasoned by the virtual queue and dynamic weighted queues service coefficient ϕ, changing according to the evaluation of the packet state in the node, reasoned by delay target time and present network load.

Key-Words: - delay, packet scheduling , QoS, queue, Real Time Services .

1 Introduction

Today the Internet provides an access to a variety of common services: form ordinary mail systems to real time interactive services. The problems of new type connected with the service quality protection and management arise now. All the states of managed objects, the possibilities of their dynamics, quality rates describing the effectiveness of management must be taken into account managing the telecommunication service quality. Telecommunication services essentially are multi-parametric, because their quality are described by many parameters [1]. The means of integrated information transmission and processing are in the networks of different operators creating the multi-operational network, so it is not enough to divide the protection needs to the separate subnetwork. In this case the efficiency of all compounds of the network must be evaluated taking into account as the efficiency and reliability of the network equipment and the connecting wires, also the probability of information transmission. In order to achieve this, the dynamic service support procedure between the user and service provider and operators is needed. If during the control time of telecommunication service it is only determined either the parameter has felt or has not felt into the margins of stated service quality norms, so the obtained data efficiency is too small to do the quick conclusions for the separate parameter change dynamics [2]. So the telecommunication service quality control data informatively will grow if the fact of not incoming or not incoming into the permit able margins will be registered, but the obtained meaning of the quantitative parameter value or the information cumulated in the intermediate Internet nodes during transformation transmission will be used.

The question of telecommunication service quality providing improving is important at this time. In order to ensure the necessary quality of the service it must be reacted adequately into the dynamic changes of the load in the network, i.e. the appropriate recourses of the network and the quality of the service must be ensured. The protection of QoS service is the main part of the strategy of network and service providers developing and entrenching in the market while the Internet speed grows. The problems of process management efficiency safety evaluating the influence of network node are solved not enough intensively [10]. In this paper we are proposing the new adaptive (AFQ-Adaptive Fair Queuing) scheduling for QoS management model, reasoned by the virtual queue and dynamic weighted queues service coefficient ϕ, changing according to the evaluation of the packet state in the node, reasoned by delay target time and present network load. This model allowing managing the packet delay in the node and network.

2 The adaptive packet scheduling of IP network

The users of nowadays informational technologies create many short packets in data streams that are irrationally used when transmitting them directly. Othervice, when the channel load is growing, the delay of packets is growing also. Those delay fluctuations negatively influences the quality of the voice and video. The main shortage of the Internet network differentiating stream forwarding quality is that it maintains the quality level of the given task (QoS) only inside the domain. So, only the management of one domain router is not enough in order to ensure the service quality between the end to end users and to improve the efficiency of the Internet network. The new or improved version of quality management allowing using network resources more effectively and assuring the quality of the provided telecommunication service to the final users is needed.

Conventional scheduling is about determining the service order of packets in the output link of a router. The packets may be served from a single queue according to First Come First Served (FCFS) principle or there may be several queues among which some form of service differentiation is performed. Various packet scheduling algorithms have been proposed for quality differentiation during the last decades. These algorithms are Priority Queueing, Earliest Due Date (EDD), Generalized Processor Sharing (GPS), Weighted Fair Queueing (WFQ), Defficit Round Robin (DRR), Class Based Queuing (CBQ) and other [3][4][5][11]. The common shortage for these algorithms is that they rely heavily on static parameterization and thus are not able to adapt to changing traffic dynamics.

The adaptive telecommunication service quality management means for differentiating stream transmitting quality ensurance model to the Internet network reasoned by the quality marginal value and evaluation of dynamic network load was proposed for solving this problem in this paper. Stream service disciplines used in nowadays routings are static and does not respond to the changes of the network load, can not always ensure the quality of network service quality between the final users. The new M/G/1K – AFQ model for the service in the queues, reasoned by the virtual queue and weighted queues service coefficient ϕ, changing according to the evaluation of the packet state in the node and allowing to manage the packet delay in the node and network was proposed.

2.1 The mathematic model of adaptive packet scheduling in the node

The structure of the analyzed node is given in Fig. 1.

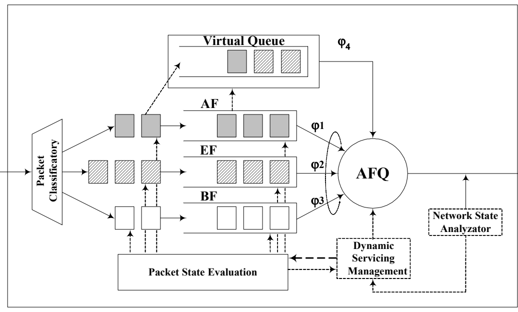

Fig. 1. The structure of the node

Many authors use similar or underlying M/G/1 model not characterizing the queue length for the description streams of different type [9][7]. M/G/1/K model is used for the queue parameters evaluation as the node in the work is described by the buffers of the finite queue [6]. The proposed stream processing in the router is reasoned by the adaptive quality management changing weighting queue service coefficient ϕ that changes according to the evaluation of the state of packets in the node and allows manage the packet delay in the node and network. M/G/1 model modification with multiple vacations is used for the mathematical description of the node. This modification was chosen according to the adaptive node (router) description proposed adaptive service model M/G/1/K – AFQ reasoned by the disciplines of weighting queue service.

The delay time of analyzed node is described by the equation [8]:

$$D = V + \frac{1}{\mu} \cdot \left(N_l + \sum_{j \in J} N_j \right), \qquad (1)$$

where N_l is the number of packets in l-th queue; N_j is the number of packet in all the system, except k-th queue; V is the required time medium for the service of all orders in the system; $\frac{1}{\mu}$ is the average time of packet service.

The parameter V is given by :

$$V = \sum_{j=1}^{K} \left\{ \frac{(1-\rho_{apt})}{\lambda_j \cdot \overline{\overline{T_{nutr_j}}}} \cdot (1+K) + \frac{\rho_{apt} \cdot (\rho - 1)}{\rho} \right\} \cdot \frac{\overline{T_{apt_j}{}^2}}{2\overline{\overline{T_{nut_j}}}} =$$

$$= \frac{1}{2} \cdot \sum_{j=1}^{K} \left\{ \frac{\overline{T_{apt_j}{}^2} \cdot (1-\rho_{apt}) \cdot (1+K)}{\lambda_j \cdot \overline{\overline{T^2_{nutr_j}}}} + \frac{\rho_{apt} \cdot (\rho - 1) \cdot \overline{T_{apt_j}{}^2}}{\rho \cdot \overline{\overline{T_{nutr_k}}}} \right\} = \qquad . (2)$$

$$= \frac{1}{2} \cdot \left\{ (1-\rho_{apt}) \cdot (K+1) \sum_{j=1}^{K} \frac{\overline{T_{apt_j}{}^2}}{\lambda_j \cdot \overline{\overline{T_{nutr_j}{}^2}}} + \frac{\rho_{apt} \cdot (1-\rho)}{\rho} \sum_{j=1}^{K} \frac{\overline{T_{apt_j}{}^2}}{\overline{\overline{T_{nutr_j}}}} \right\}$$

The improved upper and lower bounds of number of packets in node are respectively:

$$N_j \le \min\left(\sum_{\substack{k=1\\j=1}}^{J}\left((N_k+1)\cdot\frac{\phi_j}{\phi_k}\right)+\frac{\phi_j}{T_{serv}\cdot C};\quad N_j+\frac{M}{C}\cdot\lambda_v\right), \tag{3}$$

$$N_j \ge \min\left(\max\left(\sum_{\substack{k=1\\j=1}}^{J}\left((N_k+1)\cdot\frac{\phi_j}{\phi_k}\right)+\frac{\phi_j}{T_{serv}\cdot C}-1,\ 0\right);\quad N_j+\frac{M\cdot\lambda_v}{C}\right), \tag{4}$$

where: φ_j -weight coefficient of queue j; φ_k -weight coefficient of queue k; T_{serv} – packet service time; C – link throughput; M – packet size; λ - packet arrive intensity.

The queue scheduling weight coefficient φ are being changed dynamically. The new set of queue scheduling weighting coefficients for *i-th* session is based by the evaluation results of the packet state in the node. The *k-th* queue weight coefficient change can be written as:

$$\varphi_k(t)=\begin{cases}\phi_k'(t)=\varphi_k(t), & \text{if } \forall N_{i+k}\in Q^{+}\\ \phi_k'(t)=\varphi_k(t)-\Delta\phi(t), & \text{if } \forall \mathrm{N_k}\in Q^{+},\ \text{any one } \mathrm{N_i}\in Q^{+-}\\ \phi_k'(t)=\varphi_k(t)+\Delta\phi(t), & \text{if any one } \mathrm{N_k}\in Q^{+-} \text{ and } \forall N_i\in Q^{+}\\ \phi_k'(t)=0, & \text{if any one } \mathrm{N_{i+k}}\in Q^{+-}\{W_\Lambda<W_k<W_C\}\end{cases}. \tag{5}$$

The change of queue weight coefficient $\Delta\varphi$ is calculated according the equations:

$$\Delta\phi^{+}(t)\ge\frac{N_k\{Q^{+-}\}}{\underset{i\neq k}{N_i}};\quad \Delta\phi^{-}(t)\ge\frac{N_k\{Q^{+}\}}{\underset{i\neq k}{N_i\{Q^{+-}\}}}. \tag{6}$$

2.2 The evaluation of transmission packet state

The existing service quality managing methods limits only by the control between edge routers of one domain not looking into the quality parameters marginal meaning numerical values in the core nodes and dynamic network load. So, only the methods that QoS control and management provide at the same time according to several criterions are more effective for solving the problems of quality supporting. It is proposed to use the combined adaptive QoS management algorithm composed combining three main components:

- DSCP (Diff serv Code Point);
- EIGRP (Enhanced Inter – Gateway Routing Protocol;
- AFQ (Adaptive Fair Queuing).

Having evaluated the possibilities of EIGRP protocol it was chosen to use in the means of the proposed adaptive telecommunication service Internet network quality management.

Three new parameters were additionally evaluated according to our proposed means:

- $T_{\max}$ is the allowed delay norm to the given packet (according to the recommendations it can be given using the differentiating stream transmission quality protection model categories AF, EF, DE or during the sending time are included into the packet);
- T_Σ is the passed total node delay. It includes all the delays of the passed nodes;
- T_s is the static delay in the node for the given service category.

The residual packet delay time is described as the function $T_\Delta = f(T_{\max}, T_\Sigma)$. The residual packet delay time is evaluated, using the first two parameters:

$$T_\Delta = (T_{\max} - T_\Sigma) - T_M\,. \tag{7}$$

The classification margins must be set in order to classify the packets to three states. Those margins are set using the metrics of routing weight. But one more parameter T_s – static delay, i.e. the average node delay to the given category of the service set by the administrator (by the load ρ=0,5÷0,7) must be used for this purpose.

Using those parameters, EIGRP protocol calculates three metrics: Route weight metrics W_k; . route weight metrics, using static (supporting) delay W_c; route weight metrics, using residual delay W_Δ.

The evaluation of packet state is performed according to the obtained metrics means.

Evaluating possible packet states, three gradations are used:

- "good" state Q^+;
- "satisfactory" state Q^{+-};
- "bad" state Q^-.

Those packet state conditions are the function from the weight metrics $Q = f(W_i, W_c, W_\Lambda)$.

$$Q=\begin{cases}Q^{+} \text{ if } W_k<W_C<W_\Lambda,\\ Q^{+-} \text{ if } W_C<W_k<W_\Lambda \cup W_C<W_\Lambda<W_k \cup W_k<W_\Lambda<W_C \cup W_\Lambda<W_k<W_C,\\ Q^{-} \text{ if } W_\Lambda<W_C<W_k.\end{cases} \tag{8}$$

Every packet state is converted into the composed binary code that is included into the free places of DSCP.

3 Simulation model and results

Simulation was provided using telecommunication network program package Opnet. The structure of imitational simulation is given in Fig. 2.

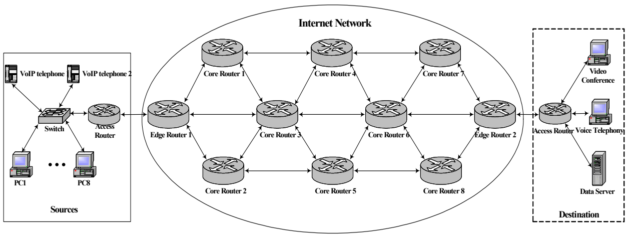

Fig. 2. Structure of simulation network

Simulation network of differentiated service consists of ten routers: two edge and eight core routers. The 100 Mb/s transmission speed is set between the final network nodes and edge routers. The speed is limited to 2 Mb/s (Fig. 2) in the network of core differentiated service network. Seven different services, such as two of real time (video conference and voice telephoning) and five of non real time (FTP, data base, e-mail, HTTP, and remote login) are in the simulation network. The routing protocol EIGRP is used in the entire network for the exchange of the data.

The initial parameters for service determination are given in Table 1.

Table 1. Service parameters determination

Service	Stream intensity			Prior.
	Codec	**File size in bytes**	**The law of the dynamics**	
Voice	G.711	-	Poiss (30)	EF
Video	H.261	-	Poiss (20)	AF43
FTP	-	700000	Exp (36)	BE
Data base	-	327680	Exp(12)	BE
E-mail	-	20000	Exp (36)	BE
HTTP	-	10000	Exp (12)	BE
Remote login	-	15000	Exp (36)	BE

Simulation was made in three cases, when the load of the network (network use) ρ changed: ρ = 0,2 to 0,3; ρ = 0,5 to 0,7; ρ = 0,75 to 0,9.

The main parameters given in Table 2 were set during the imitative simulation using the proposed adaptive quality managing means AFQ.

Table 2. The parameters of AFQ queues service methods

Service	Allowed delay margin $T_{\max}$	The set static medium delay time meaning of one node T_s	Service coefficient ϕ of initial weight queues
Voice	40 ms	2ms	20
Video	80ms	4ms	15
Other	250ms	15ms	5

Taking into account the proposed service parameter for quality evaluation in ITU-T and ETSI recommendations, during experiment was registered:

- Delay time between the final users to the different service classes;
- Network load change during the simulation;
- Weight queue service coefficient ϕ change using the proposed AFQ means.

The obtained simulation results using standard queue service discipline WFQ and proposed adaptive quality managing means AFQ are given. Coding/encoding, spreading and bufferization delay time-frames are not included performing the simulation and analyzing the packet delay time between the final users. The components of delay time-frames are allocated to the fixed delay and do not depend on dynamic load; so during the simulation we analyze only the rotational delay (packet delay in queues of the node). So, the allowed delay margin for the voice is not 150 ms, but 40 ms.

In the case ρ = 0,20 to 0,30 we present only average delay time (Fig 3).

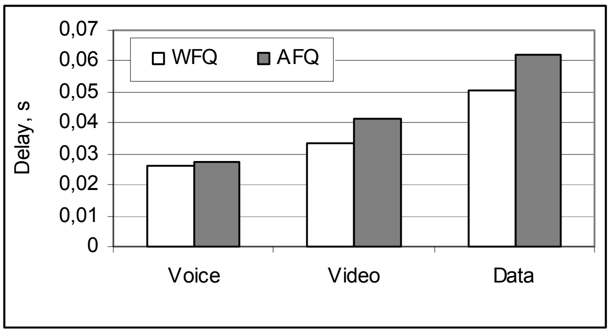

Fig. 3. Average delay time (ρ = 0,20 to 0,30)

Analyzing the obtained results of the simulation(Fig.3) we can observe that all the provided service provided in the network does not exceed posed allowed delay target time when the loading of the network is small (ρ = 0,20 to 0,30) using standard quality methods (WFQ) (voice – 40 ms; video – 80 ms; HTTP, FTP – 250 ms). So, in this case the proposed use of the method scarcely enlarges the packet average delay time between the end to end users (with WFQ – 26 ms, with AFQ – 28 ms). Analogues situation is with the packets of video, HTTP, and FTP data. This can be explained that using the proposed AFQ means, the additional delay related with the information exchange and packet state evaluation is added. Weight queue coefficients remain unchanged as the packet delay time does not exceed the allowed margins. The inconsiderable additional delay related with information processing ads using AFQ means; for this case the packet mean delay time exceeds marginally.

The service of real time (voice and video) not always meets the allowed quality demands when there is medium network use (ρ = 0,50 to 0,70). The simulation result are shown in Figures 4 – 5.

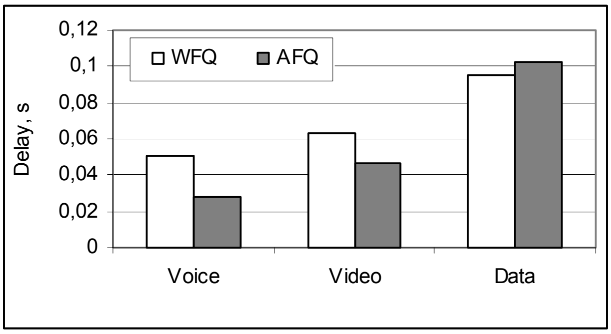

Fig. 4. Average delay time (ρ = 0,50 to 0,70)

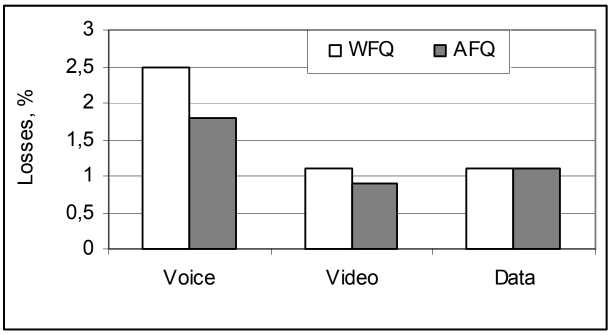

Fig. 5. Packet loses (ρ = 0,50 to 0,70)

In this case, the losses of voice service packet because of delay margin exceed is 2,5 % using WFQ. The mean delay in the common case is 51 ms . Having used the proposed adaptive quality managing means, the mean delay decreases up to 28 ms. Using AFQ the packet losses are 1,8 %. The losses occur because of "thrown packets" in transitional routers that are evaluated as having "bad" state. In the transitional nodes providing the packet throw we make the conditions for the other packets better. The analogical situation is transmitting the other real time service. In the common case the mean delay 63 ms. The mean delay decreases up to 46 ms having used the proposed adaptive quality managing means. The losses because of allowed delay margin transcendence reach 1,1 %. Having used the proposed adaptive quality managing means they are 0,9 %.

In the case ρ = 0,75 to 0,9 the obtained results are given in Figures 6 – 9.

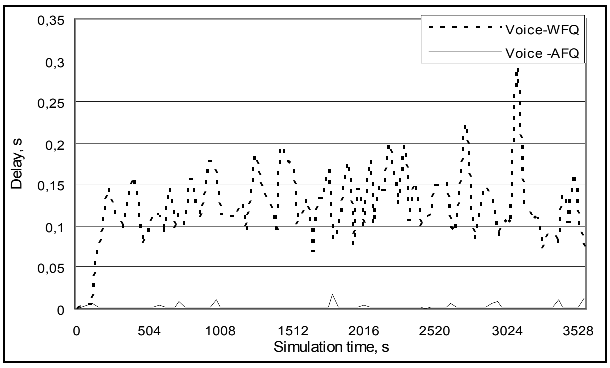

Fig. 6. Voice packet delay time between the users

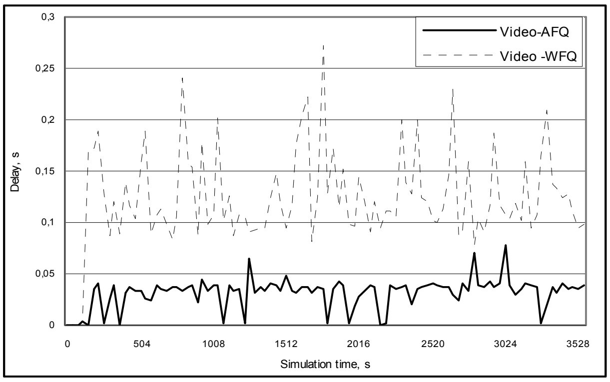

Fig. 7. Video packet delay time between the users

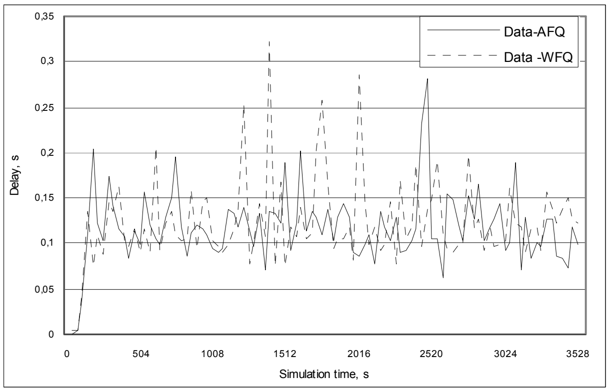

Fig. 8. Data packet delay time between the users

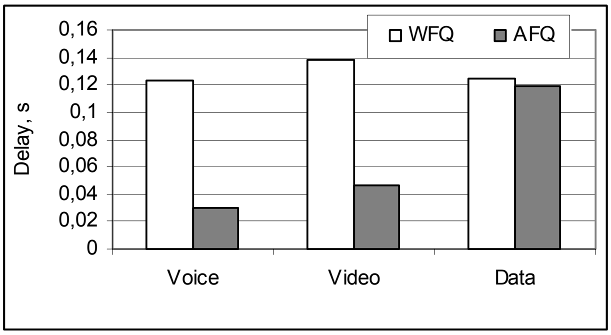

Fig. 9. Average delay time (ρ = 0,75 to 0,9)

4 Conclusions

Proposed new adaptive (AFQ) scheduling for QoS management model, reasoned by the virtual queue and dynamic weighted queues service coefficient ϕ, changing according to the evaluation of the packet state in the node, reasoned by delay target time and present network load.

The average delay of packets with the highest priorities decrease from 51 ms to 28 ms and losses from 2,5 % to 1,8 % during the imitation simulation when the network is loaded (ρ = 0,50 to 0,70). In this way, the mean delay of video packet of lower priority decreases from 63 ms to 46 ms having used the proposed adaptive quality management means. The losses because of allowed delay limit having used the proposed adaptive quality management means decreases from 1,1 % to 0,9 %.

Using proposed adaptive QoS management method AFQ , the network users with the ensured services quality number can be increased from 30% to 65 % with the same network resources.

References:

[1] Jha Sajay, *Engineering Internet QoS.* Artech house.2002 - 314p.

[2] G. Armitage, *Quality of Service in IP Networks: Foundataion for a Multi-Service Internet*, Macmillan Technical Publishin. 2000. – 309.

[3] F.Risso. *Quality of services on Packet switched network.* Report. Italy. 2001. - 151p.

[4] B. Dekeris, L. Narbutaite, T. Adomkus. The Analysis of Service Disciplines of Differentiated Services Flows // Electronics and Electrical Engineering.– Kaunas: Technologija, 2004.- No. 6 (55).- P. 34–39.

[5] B. Dekeris, T. Adomkus. Evaluation the Quality of IP Network Using Differentiated Services // Telecommunications and Electronics. – Riga: Scientific Proceedings of Riga Technical University in series 7, 2003. – Vol. 3. – P. 11-16.

[6] B. Dekeris, T. Adomkus, A. Budnikas. Analysis of QoS assurance using weighted fair queueing (WFQ) scheduling discipline with low latency queue (LLQ) . *ITI 2006 : Proceedings* .2006. pp. 507-512.

[7] Dawid Nowak. Dynamic Bandwidth Allocation Algorithms for Differentiated Services enabled Ethernet Passive Optical Networks with Centralized Admission Control. // Annual Report. Dublin City University .2005.-225p.

[8] Villy B. Iversen. *Teletraffic engineering and network planning.* DTU. 2005.-350p

[9] B. Dekeris, L. Narbutaitė, T. Adomkus. Packet delay evaluation using M/G/WFQ model *WSEAS Transactions on communications, Issue 2, Volume 4, February* 2005. – P. 51-56.

[10] J.-S. Li and C.-S. Mao. Providing Flow-Based Proportional Differentiated Services in Class-Based DiffServ Routers. IEE Proc.-Commun., Vol. 151, No. 1, February.2004.- P81-88.

[11] Vishal Phirke. Traffic Sensitive Active Queue Management for Improved Quality of Service. Report. Worcester Polytechnic Institute. 2002.- 61p.