End-to-End Queue Dimensioning in IP Network

Rossitza Goleva [1] and Seferin Mirtchev [2]

Abstract – **This paper proposes analytical and simulation approach for QoS management in IP networks. It uses flexible queueing bounds for delay and loss combined with priorities. The analytical solution uses Priority Queueing. It can be applied also to Weighted Fair Queueing, Round Robin or other technique. The simulation is done for IntServ – RSVP and DiffServ. The results demonstrate better traffic shaping on real time traffic.**

Keywords – **Queueing, IP Quality of Service management, Shaping, Wait and loss bounds.**

I. Quality of Service Management

Internet traffic is changing continuously and requires dynamic Quality of Service (QoS) estimation and management [1], [2]. TCP and UDP segments are different by size and QoS requirements. Packet sizes between 20 and 1500 bytes influence the servicing time. Output scheduling algorithms like Priority Queuing (PQ), Weighted Fair Queuing (WFQ), Round Robin (RR) or other applied in the queues change the distribution of the packet flows [4], [5]. Traffic policing and traffic shaping mechanisms, packet dropping and early random detection can change the characteristics of the flows.

The traffic profile is changing from hour to hour, weekly and due to the season. Effects like self-similarity and long tailed queues are a matter of long research. Internet Service Providers need to configure their equipment dynamically in order to meet the Quality of Service requirements. The mixture of voice and data traffic requires specific prioritization and reservation scheme that will allow conformance to the QoS requirements. Wired and wireless parts of the connections, access and transport devices gather the traffic in a different way. It is difficult to predict end-to-end QoS in a typical IP connection with many hops.

In his paper, Atov [2] has presented a combined DiffServ/MPLS approach that classifies the traffic depending on the delay and delay jitter requirements. MPLS is used for resource reservation to support delay bounds. DiffSerf is applied for prioritization of different aggregated traffic.

A comparison between Weighted Round Robin (WRR) and Weighted Fair Queueing (WFQ) for resource allocation is shown in [8]. A demonstration of the FTP traffic requirements and its management to fulfil the QoS requirements and channel capacity utilization criteria is presented in [10].

UDP and TCP traffic sources can generate symmetric or asymmetric traffic. UDP is symmetric in VoIP and asymmetric in TFTP service. UDP is also simplex in IPTV. TCP traffic is asymmetric in HTTP, email, FTP. It can be also symmetric for client-server applications. Bounds on delay in real time and non real time services can vary in quite high ranges. The delay can be close to the upper limits depending on the number of hops, processing delay, scheduling in the queues, credit fluctuations.

IntServ – RSVP and DiffServ occupy waiting places in the queue in different ways. Patchararungruang [10] proposes a simplified method of router representation applying fuzzy logic. His results are approximate. The model is not applicable for fast calculation and dynamic resource reservation. The capability of Pareto distribution to model data traffic and especially heavy tailed effect in the router interfaces is demonstrated in [1]. Chen [11] shows the capability of Multi-Reservation Multiple Access (MRMA) scheme to guarantee the delay bounds in access networks.

End-to-end delay and delay variation in TCP and UDP services under bursty traffic depends strongly on the traffic distribution, policing and shaping applied [6], [8]. Series length is important in delay and queueing places bounds. In paper [5], numerical results after simulation for QoS techniques IntServ – RSVP and DiffServ are presented. Similar approach can be also seen in [8].

The aim of this paper is to show course-grained and fne-grained approach for traffic shaping that will allow dynamic configurations and end-to-end QoS management. We show how the QoS requirements can be kept using combined DiffServ and Priority queueing approach.

II. Traffic Sources and Simulation Model

Three types of traffic sources are assumed in the network model – Voice over IP, LAN emulation, email. LAN traffic has lower priority in comparison to the VoIP traffic and higher priority in comparison to the email. In VoIP service silence and talk intervals are exponentially distributed. On-off model is applied. Short packets are considered as between 64 and 100 bytes. They carry up to 15 ms voice [12]. LAN emulation is specific with its TCP sessions. Sessions are established for any Internet connections [5]. LAN packets are between 20 and 1500 bytes. Emails are specific with packet exchange mostly in one direction. Packets are taken to be as long as 1500 bytes.

End-to-end Quality of Service requirements for real time services allow up to 150 ms end-to-end delay and up to 30 ms

[1] Rossitza Goleva is with the Faculty of Telecommunications at Technical University of Sofia, 8 Kl. Ohridski Blvd, Sofia 1000, Bulgaria, e-mail: rig@tu-sofia.bg

[2] Seferin Mirtchev is with the Faculty of Telecommunications at Technical University of Sofia, 8 Kl. Ohridski Blvd, Sofia 1000, Bulgaria, e-mail: stm@tu-sofia.bg

delay jitter. Packet loss probability is considered to be less that 0,1. The loss and delay bounds limit the total number of hops. The bounds for waiting times for TCP traffic are calculated on slow start timer that has typical value of 3 seconds. Servicing times per packets are simulated on 100 Mbps line interface.

III. NETWORKS OF QUEUES APPROXIMATION

In typical router interface, few queues classify the traffic depending on the port number. Many priority levels can be applied according to different Quality of Service mechanisms like IntServ – RSVP and DiffServ. All of them classify the traffic in input queue and identify scheduling technique to the output line (Fig. 1).

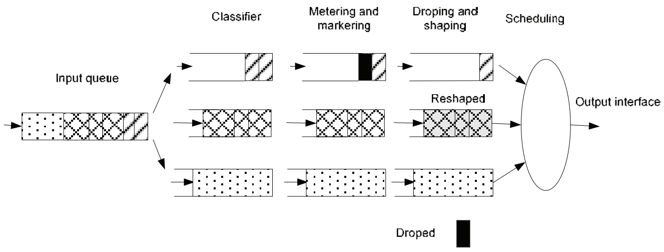

Fig. 1. Parallel Queues

Typical IP connection has many hops. It can be modelled with cascaded Leaky Bucket queues with priorities (Fig. 2). The sequence of queues can be considered to have equal characteristics with exceptions in access queues at both ends of the connection. In access part of the model limits for waiting place and waiting times should be more relaxed in comparison to the others. Access routers are usually less powerful than core network routers.

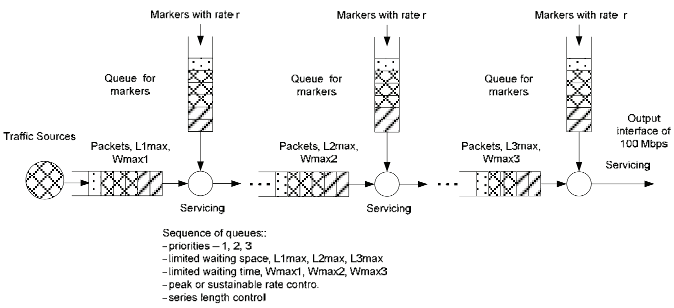

Fig. 2. Sequence of Queues

The output interface can be easily considered as single Leaky Bucket queue with priorities (Fig. 3). Limits to the queue length L_{imax} and waiting times W_{maxi} are calculated in order to satisfy requirements for Quality of Service. L_i are the current lengths of the queue fractions depending on the type of packets.

Cascaded queues accumulate waiting times and end-to-end losses. The overall throughput of the connection is the minimal throughput of the nodes in the connection. All routers in the connection see round trip delay and are capable to count total loss. So, the structure of the model shown on Fig. 2 can be applied end-to-end keeping in mind that the delay and loss bounds are cumulative products of the delays and losses bounds in the nodes of the connection.

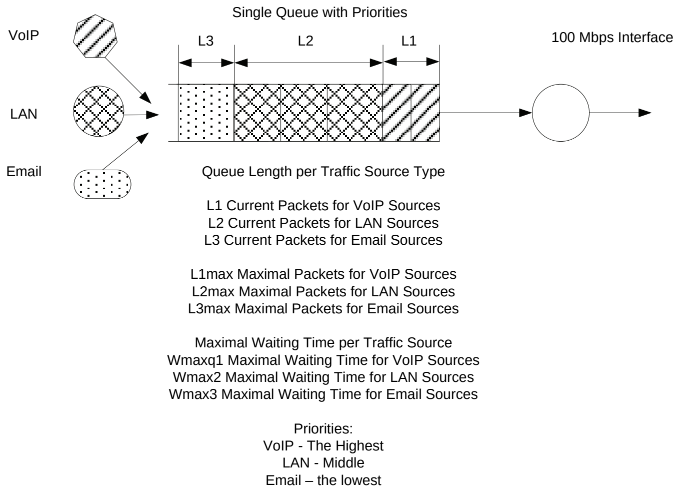

Fig. 3. A single node QoS model and end-to-end QoS model

Different scheduling techniques like Priority Queuing (PQ), Weighted Fair Queuing (WFQ) and Round Robin (RR) can be considered for scheduling in parallel queues. Priority queueing technique is presented by Kleinrock in [3]. He derived the formula for the priority scheme with preemptive and non preemptive disciplines taking into consideration second moments of the queueing delay and propagation delay as well as distribution of the packets. The idea of Kleinrock is not directly applicable to the router interfaces because of its complexity.

Below, we try to simplify the idea and make it more applicable for dynamic interface dimensioning. We show that queue scheduling influence significantly the end-to-end behaviour. The model shown on Fig. 3 is considered. Delay bounds depend on slow start procedure for TCP traffic and real time transmission requirements for VoIP. Calculation of the waiting times depends on type of priority, IntServ-RSVP/DiffServ application and scheduling algorithms. For UDP traffic short queue for VoIP is modeled. Long queue is applied for LAN emulation traffic that is mostly http. SMTP traffic can be transmitted with lower priority in comparison to the http traffic.

Parameters of the Leaky bucket queues depend on type of services. In the simulation model, bigger packetization and depacketization delays in the access routers are considered. By changing the delay and loss bounds and priorities it is possible to adjust router interfaces behaviour depending on the real time circumstances in the networks. It can be done per aggregated service and aggregated packet level. The model applies three priorities.

IV. RESULTS FROM SIMULATION

Simulation is performed on C++ language. The waiting time and loss bounds are calculated in accordance to DiffServ [14], [15]. Queue behaviour is complex due to the priorities and limits on waiting times and places. Many parameters have been derived from the model including probability of packet loss due to the lack of place in the queue, probability packet to be dropped due to the exceed of waiting limit, probability to wait for different types of traffic, observations on of the distribution of packets intervals, queue lengths, delay, delay jitter. Statistical accuracy of the derived results is proven by batch mean method for output results analysis with Student distribution and confidence probability 0.95 (Table I).

TABLE I. NUMERICAL RESULTS ON UTILIZATION AND LOSS PROBABILITIES FOR DIFFSERV

VoIP maximal delay, s	0.002	0.001	0.00004	0.0001	0.0001
VoIP queue limit	10	5	7	3	2
LAN maximal delay, s	0.07	0.06	0.0004	0.04	0.03
LAN queue limit	50	25	24	5	5
Email maximal delay, s	0.09	0.08	0.0004	0.06	0.04
Email queue limit	50	25	24	5	5
Queue size	110	55	55	13	12
Probability of losses due to the lack of place	0	0	0.18272	0.0265	0.0881
Prob. of place losses for VoIP	0	0	0	0.0001	0.0056
Prob. of place losses for LAN	0	0	0.18038	0.0309	0.0978
Prob. of place losses for email	0	0	0.54851	0.0017	0.1389
Prob. of losses due to the exceed of waiting limit	0	0	0.00009	0	0
Prob. of waiting losses for VoIP	0	0	0.00062	0	0
Prob. of waiting losses for LAN	0	0	0	0	0
Prob. of waiting losses for email	0	0	0	0	0
Prob. to wait observed	0.495	0.4673	0.6945	0.6436	0.7175
Prob. to wait observed for VoIP	0.522	0.5097	0.88853	0.6934	0.8259
Prob. to wait observed for LAN	0.492	0.4634	0.69823	0.6380	0.7065
Prob.to wait observed for email	0.489	0.4643	0.29602	0.6312	0.6336
Utilization	0.473	0.4585	0.82179	0.6132	0.752
Utilization for VoIP	0.045	0.0437	0.13892	0.0681	0.0997
Utilization for LAN	0.423	0.41	0.64798	0.523	0.6150
Utilization for email	0.005	0.0048	0.03489	0.0221	0.0373

Interesting results that influence directly interfaces and queue management are derived based on queue length per service type. The queue fractions of the three services are observed. Under almost the same utilization factor, the fractions of the queue per service is changed. Scenario limits are shown on the first six lines.

On the third column of the table, it can be seen that tight bound in waiting time of “0.00004” will keep the VoIP queue short with probability of waiting time loss of “0.00062”. The total load on the interface is significant (0.82179) but the occupation of the interface by VoIP packets is very low in comparison to other cases (0.13892). This means that the influence of the waiting time limit is significant in the system. The system also shows the deterministic nature of the IP traffic. The same effect can be seen on the last column.

The adjustments of the waiting time and waiting place bounds are considered equal along the connection with exception for access routers. It can be done also proportionally depending on the results for round trip delay from traceroute measurement.

V. ANALYTICAL MODEL FOR QUALITY OF SERVICE MANAGEMENT

The simulation model is used for the end-to-end cumulative QoS analyses of the model in Fig. 2. The proposed simple analytical algorithm supports end-to-end maximal limits calculations per packets, per aggregated traffic and per scheduling algorithm.

Kleinrock in his study on Internet traffic [3] has derived formulae for different packet traffic. His derivations are based on end-to-end round trip delay. They can be applied for end-to-end analyses of the traffic with known distribution and profiles. It cannot be applied dynamically at the router interfaces. The proposed approach is simpler and applicable for fast dimensioning.

Queue limits for TCP traffic is calculated from number of hops and slow start timer. ICMP requests and traceroute requests return the minimal, maximal and mean delay. Upper limits of waiting time for VoIP traffic is also calculated using number of hops and 150 ms end-to-end maximal bound. Waiting time bounds calculated analytically are compared to the results from simulation. VoIP traffic is considered to be modelled like ON-OFF source. 1250 VoIP on-off traffic sources with intensity of 0.05 Erl are aggregated.

In Priority Queuing (PQ) mechanism, every packet from the queue with higher priority will be send before any packet from the queue with lower priority. The nonpreemptive queueing is considered. In case the VoIP traffic is dominated in the network, other traffic will suffer of big delays. Minimal delay on fragmentation at both ends is equal to the voice buffer, i.e. 20-30 milliseconds. Maximal waiting time per queue $W_{maxVoIP}$ is calculated with Eq. (1) (Table II). Maximal waiting time bound for LAN packet $W_{\max LAN}$ depends on existing number of VoIP packets because they are of higher priority. In equations, t_1 and t_2 are the time necessary for the router interface to send VoIP and LAN packet, *n* is the number of VoIP packets, *m* is the number of LAN packets, *p* is the number of email packets in the queue. The delay bound for real time services is kept on the favour of the non real time services. Every packet of type 3 will wait packets from type 1, 2 and 3 in the schedule to be served. Furthermore, new packets from type 1 and 2 arriving in the queue input will be served before the waiting packet of type 3. This discipline is serving high priority traffic in a very good way.

$$W_{\max VoIP} \geq nt_1 \quad W_{\max LAN} \geq nt_1 + mt_2 \tag{1}$$

$$W_{\max email} \geq nt_1 + mt_2 + pt_3 .$$

The limit for queue length calculated as an upper one is considered as course-grained bound. The waiting time limit is considered to be fine-grained bound. Two simple instruments for interface adjustments are obtained. The maximal waiting time for LAN traffic is corrected with the maximal waiting time for VoIP traffic. The value is less that the one practically calculated from RTT. The limits for email and other lower priority services are shared with the other TCP traffic. Time for acknowledgement service is considered negligible. Calculations are made for 100 Mbps interface.

TABLE II.

NUMERICAL RESULTS ON DELAY BOUNDS IN PQ

VoIP service (traffic)		
Low delay bound	7.62939E-06	sec
High delay bound	0.0048	sec
Servicing time for 100 bytes packet	7.62939E-06	sec
Number of hops	25	
Packetization delay	0.03	sec
Upper number of packets in queue	629.1456	packets
n –VoIP packets	600	
Queue length for VoIP packets	62914.56	bytes
Waiting time for VoIP more than	0.004577637	sec
LAN service traffic		
m - 200 LAN packets	1460.4352	packets
Packets length	1000	bytes
Servicing time per packet	7.62939E-05	sec
Low delay bound	7.62939E-05	sec
High delay bound per router	0.111422363	sec
Waiting time for LAN packets more than	0.116	sec
Queue length for LAN	2190652.8	bytes
p - email packets	50	
Servicing time packet	7.62939E-05	sec
Low delay bound	7.62939E-05	sec
High delay bound per router	0.111422363	sec
Waiting time for email packets more than	0.119814697	sec
Queue length for email	75000	bytes

VI. CONCLUSION

In this paper, we demonstrate similar to [2] approach for aggregated DiffServ traffic without use of MPLS capabilities. We show that due to the delay bounds in the queues techniques like Weighted RED algorithms can be avoided. Instead of WRED packets that delay will be dropped. The approximate approach to network dimensioning ([6], [8]) with single server queue with priorities and place and time bounds is proven to be effective. The approach can be used for TCP and UDP traffic as well as for sequence of queues and parallel queues.

The capability of the algorithm at router interface to shape traffic from different traffic sources is shown. The approach allows some services to obtain better service conditions on the favour of others. Limits criteria for queue management as coarse-graned and fine-grained adjustments are evaluated. Cumulative traffic with different distribution and without specific distribution of the packets is studied. Combination with PQ prove possibility to implement the ideas on traffic shaping on the interface and end-to-end.

With numerical results, we also prove that shaping is effective under interface load above 0.6. Therefore, it can be applied as congestion management technique. Because of the simplicity of the proposed algorithm, it can be implemented in real time queues for fast reconfiguration and QoS dynamic management. The derived results are applicable to the routers that are capable to classify packets and manage dynamically queues and priorities.

ACKNOWLEDGEMENT

This paper is sponsored by the COST Action IC0703 “Data Traffic Monitoring and Analysis (TMA), research programme at Technical University of Sofia, and Mtel Ltd., Bulgaria.

REFERENCES

[1] Ariffin, S.: Accelerated Simulation for Packet Buffer with a Non FIFO Scheduler. Submitted for the Degree of Doctor of Philosophy, Queen Mary University of London, March (2006)

[2] Atov, I., Harris, R.J.: Capacity Provisioning in DiffSrv/MPLS Networks Based on Correlated Traffic Inputs. Performance, Computing, and Communications, 2004 IEEE International Conference, pp. 187- 193 (2004)

[3] Kleinrock, L.: Queueing Systems. Volumes I and II, John Wiley and Sons (1976)

[4] Ralsanen, V.: Implementing Service Quality in IP Networks. John Wiley & Sons, Ltd. (2003)

[5] Goleva, R., Atamian, D.: End-to-End Delay Analyses in IP Networks. ICEST 2008, pp. 40-43, Nish, Serbia (2008)

[6] Peng, G.: Availability, Fairness, and Performance Optimization in Storage Virtualization Systems. A dissertation, The Graduate School, Stony Brook University (2006)

[7] Iversen, V.: Teletraffic Engineering Handbook. ITU-D (2005)

[8] Victor, V., Záruba, G., Balasekaran, G.: RECHOKe: A Scheme for Detection, Control and Punishment of Malicious Flows in IP Networks. IEEE GLOBECOM (2007)

[9] Fernekes, A., Klein, A.: Influence of High Priority Users on the System Capacity of Mobile Networks. WCNC (2007)

[10] Patchararungruang, S., Halgamuge, S., Shenoy, N.: Optimized Rule-Based Delay Proportion Adjustment for Proportional Differentiated Services. IEEE Journal on Selected Areas in Communications, Vol. XX, No. Y, January (2005)

[11] Chen, H., Fei Yu, Chan, Henry C. B., Leung, Victor C. M.: A Novel Multiple Access Scheme over Multi-Packet Reception Channels for Wireless Multimedia Networks. IEEE Transactions on Wireless Communications, Vol. 6, No. 4, April, pp.1501 (2007)

[12] Mirtchev, S.: Stud y of Queueing Behaviour in IP Buffers. iTech’07 Conference, Varna, St. Kostantin and Elena, June, pp. 187-193 (2007)

[13] Mirtchev, S., Goleva, R.: A Discrete Time Queueing Model with a Constant Packet Size. ICEST 2009, pp. 173-176, V. Tarnovo, Bulgaria (2009)

[14] Mirtchev, S., Goleva, R.: Study of Discrete Time Pareto/D/1/k Queue by Simulation. Telecom 2009, 8-9 October, pp. 306-313, St. Constantine, Varna, Bulgaria (2009)

[15] Mirtchev, S., Goleva, R.: Evaluation of Pareto/D/1/k Queue by Simulation. International Book Series “Information Science&Computing” No.1, Supplement to the International Journal „Information Technologies & Knowledge”, Vol.2, pp.45-52 (2008)