TRAFFIC MODELLING IN DISRUPTION-TOLERANT NETWORKS

Rossitza Goleva*, Seferin Mirtchev**

*Assistant Professor, Department on Communication Networks, Technical University of Sofia, Bulgaria, rig@tu-sofia.bg

**Associate Professor, Ph.D., Department on Communication Networks, Technical University of Sofia, Bulgaria, stm@tu-sofia.bg

ABSTRACT

End-to-end Quality of Service management and analyses in Disruption-Tolerant Networks requires a dynamic and flexible dimensioning approach. Connection is usually not performed on single path. Multipathing and multihoming in aggressive environment is the only way to transmit reliably traffic bundles. This paper proposes analytical/simulation solution for end-to-end performance analyses. The input flows are modeled by Polya distribution that is applicable for continue and discrete time systems. It is also appropriate for peakedness and packet long range dependence simulations. We use flexible queuing bounds for packets' loss and delay combined with priorities. The bounds depend on cross-layer Quality of Service mapping at application, bundle, transport and IP traffic characteristics. The solution is combined with scheduling techniques at IP level such as Priority Queuing, Weighted Fair Queuing and Round Robin. The results for DiffServ algorithms are shown. The results for single interface with Polya distribution traffic sources and fixed packet length show the necessity of short queues for real time services. When queues with more than 50 places are applied the long range dependence between packets in the queue is seen. Furthermore, we calculate end-to-end delay bounds for real time and non real time services to show how the delay bounds could be used for traffic regulation for delay intolerant traffic. Our calculations demonstrate the capability of the approach for better traffic shaping of real time service on the favor of non real time services.

Keywords – Queuing system, Queuing analyses, Discrete time queue, Polya distribution, IP traffic modeling

1 QUALITY OF SERVICE MANAGEMENT IN DTN

Disruption-Tolerant Networks (DTN) use heterogeneous connectivity. Because of the instability of the end-to-end connections they use multipathing and multihoming approaches. Performance of different paths per bundle is investigated. Synchronised bundle streams can be rerouted dynamically [1]. Due to the changing technological circumstances a dynamic Quality of Service (QoS) estimation and management [2] is necessary. The bundle sizes per service depend on reliability of the transmission channel. The distribution of the packet flow influence end-to-end performance in a different way. The scheduling algorithms like Priority Queuing (PQ), Weighted Fair Queuing (WFQ), Round Robin (RR) applied in the queues change the distribution of the packet flows significantly [3], [4]. Traffic policing and traffic shaping mechanisms, packet dropping and early random detection can favor or not real time services.

The mixture of voice and data traffic requires specific prioritization and reservation scheme [5]. Delay, delay jitter and loss requirements are calculated easily within given platform but it is hardly to predict them end-to-end. QoS parameters have to be calculated on many paths simultaneously.

In his paper, Atov [2] has presented a combined DiffServ/MPLS approach that classifies the traffic depending on the delay and delay jitter. DiffServ is applied for prioritization. We propose the use of bundle time characteristics for packet management in the queues. A comparison between Weighted Round Robin (WRR) and WFQ for resource allocation is shown in [6]. A demonstration of the FTP traffic requirements and its management in order to fulfill the QoS requirements and channel capacity utilization criteria is presented in [7].

IntServ – RSVP and DiffServ occupy waiting places in the queue in different ways. In his paper, Patchararungruang [8] proposes a simplified method of router representation applying fuzzy logic. 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 [9, 10]. Chen [7] shows the capability of Multi-Reservation Multiple Access (MRMA) scheme to guarantee the delay bounds in access networks.

In [8] the authors change TCP time parameters and achieve successful transmission over Disruption-Tolerance Networks. In [11] the capability to apply Polya distribution for packet flows that have long range dependence and high peakedness is shown.

Ott and Kutscher show in their work [12] how different RTT can influence overall QoS. Cerf [1] defines three priorities in DTN. The only attempts for QoS definition in DTN are made by Wood and discussed in DTNRG forum partially. In [12-14] the inapplicability of the TCP timers for QoS support in DTN is shown. Balasubramanian shows different components of the end-to-end delay and its measurements [14]. System with priorities is shown in [15]. QoS techniques IntServ – RSVP and DiffServ are investigated in [16]. The necessity of bundle protocol over already reliable TCP protocol for long connections is shown in [17]. The effect of Round Trip Delay (RTT) over Quality of Service for TCP connections is demonstrated in [18, 19].

In this paper, we demonstrate an approach to network dimensioning that uses single server queue with priorities, place and time bounds with Polya traffic sources and fixed packet lengths. The approach is applied for application, bundle, transport, Internet traffic as well as for sequence of queues and parallel queues. Quality of Service requirements are mapped on a cross connection level in accordance to the DTN Architecture Reference Model (Fig. 1). We show that the distribution of the flows after the traffic sources and the queuing mechanisms like PQ, WFQ and RR are important in queue analyses. We show how the QoS requirements can be kept using combined reservation approach not only for non real time but also for real time services with high peakedness and long range dependence between packets in the queue. The derived results are applicable to the routers that are capable to classify packets and manage dynamically queues and priority.

2 TRAFFIC SOURCES AND SIMULATION MODEL

Voice over IP, FTP and email traffic sources are assumed in the network model. FTP traffic has lower priority in comparison to the VoIP traffic and higher priority in comparison to the email. The number of traffic sources and services are aggregated at service level (Fig. 1). The distribution of the traffic at bundle, transmission and packet level with Polya distribution is observed and applied in the end-to-end dimensioning. In VoIP service silence and talk intervals are exponentially distributed. On-off model is applied. Short packets with up to 15 ms voice are considered [4]. FTP and email work with TCP sessions. They are specific with packet exchange mostly forward and acknowledgements backward.
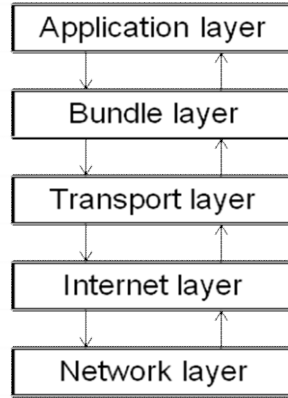
Fig. 1. DTN Architecture Reference Model.

End-to-end QoS requirements for real time services are up to 150 ms end-to-end delay, up to 30 ms delay jitter, packet loss less that 0,01. The bounds for waiting times are calculated under consideration of end-to-end delay for the service. It depends mostly on the bundle duration or TCP Slow Start Timer (STT).

3. POLYA ARRIVAL PROCESS AND NETWORK APPROXIMATION

The Polya arrival process is a pure birth process with an average arrival rate λ. The probability *Pi(t)* of *i* arrivals in an interval with duration of *t* seconds is given by

$$P_o(t) = \left(1 + \beta \lambda t\right)^{-\frac{1}{\beta}}$$

$$P_i(t) = \left(\frac{\lambda t}{1 + \beta \lambda t}\right)^i \frac{1(1+\beta) \ \dots \ [1+(i-1)\beta]}{i!} P_o(t) \ .$$

(1)

The Polya distribution is a variant of the negative binomial distribution. Its mean value (the average number of arrivals in an interval of length *t*) is

$$M(t) = \sum_{i=1}^{\infty} i P_i(t) = \lambda t \ .$$

(2)

The variance of the number of arrivals in an interval of length *t* is

$$V(t) = \sum_{i=0}^{\infty} [i - M(t)]^2 P_i(t) = \lambda t (1 + \beta \lambda t) \ .$$

(3)

The peakedness of the Polya input flow is

$$z(t) = \frac{V(t)}{M(t)} = 1 + \beta\lambda t > 1. \tag{4}$$

When $\beta = 0$, $M(t) = V(t) = \lambda t$ i.e. it is a regular Poisson process. When $\beta = 1$ the Polya distribution is a geometric distribution.

Let us consider a single server queue Polya/D/1 with a Polya input stream with arrival rate λ, constant service time τ and unlimited waiting places. This queueing system is a non-Markovian model or renewal process. We will analyse it using the theory of embedded Markov chain. The finite state-transition probability diagram for the embedded Markov chain is shown on Fig. 2, in which we show only transitions in state n $(0 < n < \infty)$.

Let τ be the length of an observed interval that falls at random in time. The packet in service at time t should have left the system at time $t + \tau$ because of the constant service time. Then, we have the relation

$$P_n = P_o Q_n(\tau) + \sum_{i=1}^{n+1} P_i Q_{n-i+1}(\tau) \quad n = 0,1,2,...,\infty. \tag{5}$$

where: P_n is the steady state probability that n packets exist in the system

$Q_n(\tau)$ is the probability that n packets arrive during the service time

τ is the service time.

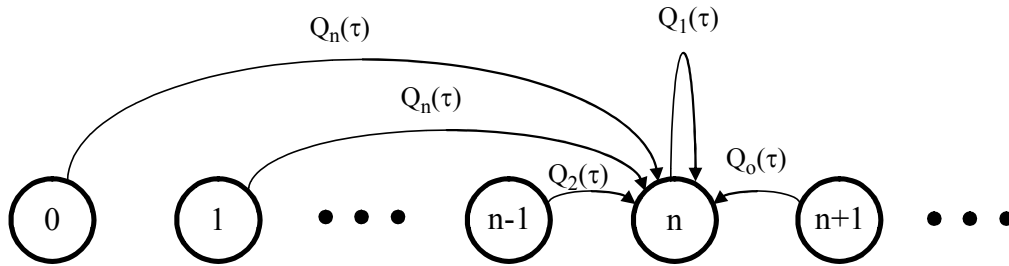*Fig. 2. How to reach state n (0<n<∞) at the end of an observed time interval of length τ.*

Because of the Polya arrival process, the probability that n packets arrive in the service time is

$$Q_o(\tau) = (1 + \beta\lambda\tau)^{-\frac{1}{\beta}} \tag{6}$$

$$Q_n(t) = \left(\frac{\lambda\tau}{1 + \beta\lambda\tau}\right)^n \frac{1(1+\beta) \ \dots \ [1+(n-1)\beta]}{n!} Q_o(\tau) .$$

We can calculate the steady state probability of this queueing system.

In typical router interface, few queues classify the traffic depending on the port number. Many priority levels can be according to different QoS mechanisms like DiffServ. All of them classify the traffic in input queue and identify scheduling technique to the output line. It can be modeled with cascaded Leaky Bucket queues with priorities.

Parallel and cascaded queues are approximated with single server queue [4] with priorities, finite queue and delay bounds (Fig. 3). Limit to the queue length is denoted as L_{imax} and waiting time as W_{maxqi}. Values of these bounds are calculated in accordance to the time characteristics of service, DTN bundle, TCP or UDP transmission buffers and IP queues requirements. They should conform to defined end-to-end QoS requirements. L_i are the current lengths of the queue fractions depending on the type of service.
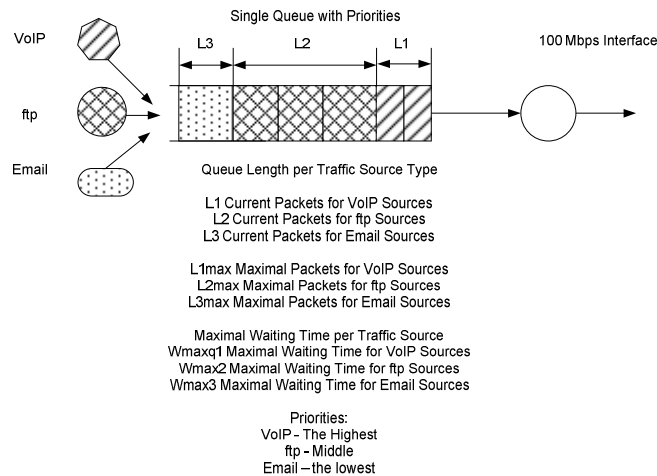

Fig. 3. End-to-end QoS Model Approximated as Single Node

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. 3 can be applied end-to-end keeping in mind that the delay and loss bounds

are cumulative products of the delays and loss bounds in the nodes of the connection.

Three scheduling techniques - PQ, WFQ, RR and combined approach are considered. We show that by changing the delay and loss bounds, priorities and scheduling it is possible to adjust router interfaces behavior depending on the real time circumstances in the networks. It can be done per aggregated service, aggregated bundle and aggregated packet level.

4 RESULTS FROM SIMULATION

The numerical results for obtained for congestion probability of the single interface with Polya distribution are shown on Fig. 4. The peakedness z influence on a highly occupied interface with 0.8 erl offered traffic for Polya/D/1 system is significant. The results are obtained on a short real time queues.

Figure 5 presents the normalized mean system time *(W/τ)* as function of the offered traffic intensity with different peakedness of the Polya input flow. The influence of the peakedness on the mean system time is high even in a short real time queue. The peaked input flow changes the waiting time distribution significantly.
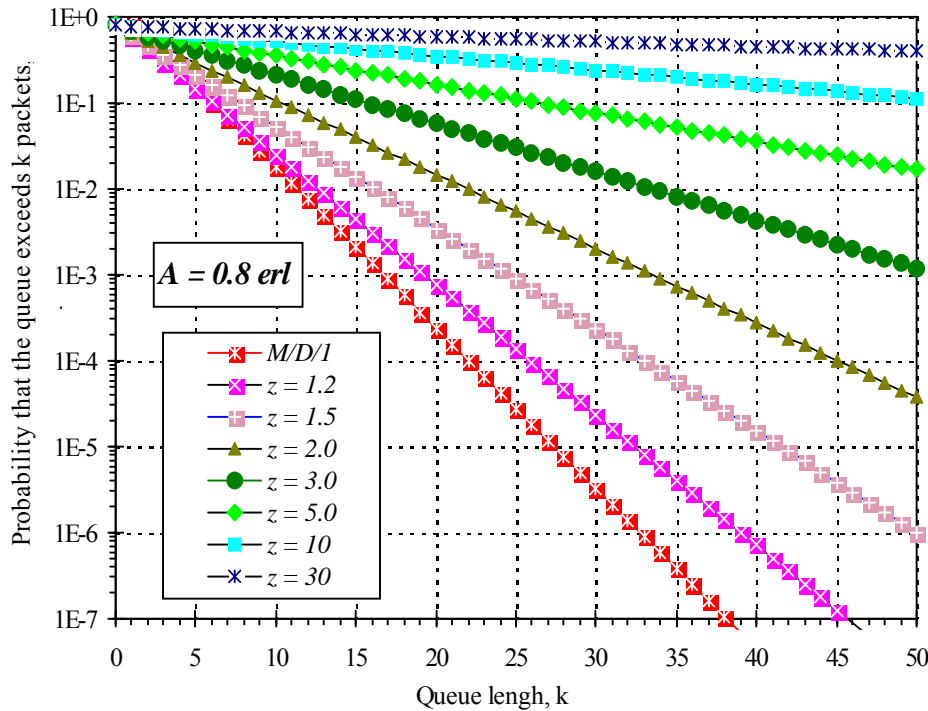

Fig. 4. Polya/D/1 time congestion probability with different peakedness.
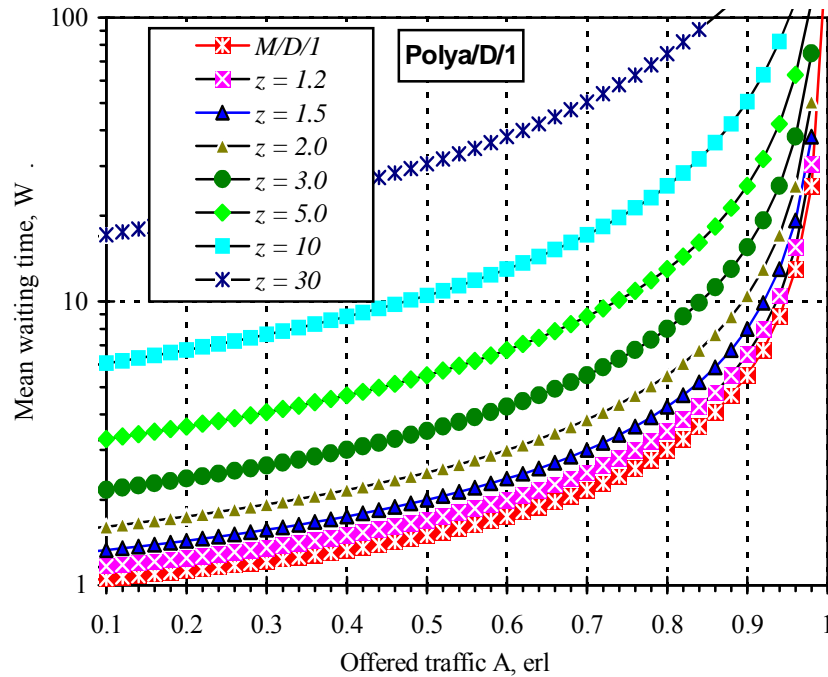
Fig. 5. Polya/D/1 normalised mean system time with different peakedness.

In this paper it is shown that the influence of the variance of the input stream over the performance measures is significant. The peaked input flow contributes to raise the congestion and waiting time. This is of importance for real-time traffic.

In end-to-end simulation model the waiting time and loss bounds are calculated in accordance to DiffServ [3-4] and QoS requirements. Many parameters have been derived from the model including probability of packet loss due to the lack of place in the queue, probability packets to be dropped due to the exceed of waiting limit, probability to wait for different types of traffic, interface utilization, queue lengths, and delay. Statistical accuracy of the derived results is proven by batch mean method for output results analysis with Student distribution and confidence probability of 0.95. The results are shown on Table 1. Under almost the same utilization factor, the fractions of the queue per service are changed. Scenario limits are shown on the first six lines. On the last column 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 (0.82) but the occupation of the interface by VoIP packets is very low in comparison to the other (0.139). The influence of the waiting time bound is significant during congestion.

Table 1. Numerical Results on Utilization and Loss Probabilities

VoIP time bound, s	0.002	0.001	0.0001	0.0001	0.0001	0.00004
VoIP queue bound, packets	10	5	3	3	2	7
FTP time bound, s	0.07	0.06	0.04	0.04	0.03	0.0004
FTP queue bound, packets	50	25	5	5	5	24
Email time bound, s	0.09	0.08	0.06	0.06	0.04	0.0004
Email queue bound, packets	50	25	5	5	5	24
Queue size, packets	110	55	13	13	12	55
Prob. of loss due place bound	0	0	0.0006	0.0265	0.0881	0.1827
Prob.of loss due place bound for VoIP	0	0	0	0.0001	0.0056	0
Prob.of loss due place bound for FTP	0	0	0.0007	0.0309	0.0978	0.1804
Prob.of loss due place bound email	0	0	0	0.0017	0.1389	0.5485
Prob. of loss due to the time bound	0	0	0	0	0	0.0001
Prob. of loss due to the time bound for VoIP	0	0	0	0	0	0.0006
Prob. of loss due to the time bound for FTP	0	0	0	0	0	0
Prob. of loss due to the time bound for email	0	0	0	0	0	0
Utilization	0.4732	0.458	0.468	0.6132	0.752	0.8218
Utilization for VoIP	0.0453	0.044	0.0447	0.0681	0.0997	0.1389
Utilization for FTP	0.423	0.41	0.4185	0.523	0.6150	0.648
Utilization for email	0.0045	0.005	0.0046	0.0221	0.0373	0.0349

5 SCHEDULING TECHNIQUE INFLUENCE

The influence of the scheduling algorithms is investigated with Priority Queueing, Weighted Fair Queueing and Round Robin. The length of the queue fraction per VoIP packets $Q_{LenVoIP}$ is made equal to the series length. In PQ every packet from the queue with higher priority will be send before any packet from the queue with lower priority. The nonpreemptive queuing mechanism is taken into account. In case the VoIP traffic is dominated in the network, other traffic will suffer of big delays. TCP connections will

enter slow start state with high probability. Maximal waiting times per queue $W_{maxVoIP}$, W_{maxftp}, $W_{maxemail}$ 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 FTP packet, n is the number of VoIP packets, m is the number of FTP packets, p is the number of email packets.

$$W_{\max VoIP} \geq Q_{LenVoIP} t_1 \quad \text{or} \quad W_{\max VoIP} \geq n t_1 \qquad W_{\max ftp} \geq n t_1 + m t_2 \qquad (7)$$

$$W_{\max email} \geq n t_1 + m t_2 + p t_3$$

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 FTP traffic is corrected with the maximal waiting time for VoIP traffic. The limits for email and other lower priority services are shared with the other TCP traffic (Table 2). Calculations are made for 100 Mbps interface. Delays are acceptable for VoIP service.

Table 2. Numerical Results on Delay Bounds in PQ

Parameter	VoIP	FTP	Email
Low delay bound, sec	7.63E-06	7.63E-06	7.63E-06
High delay bound, sec	0.0048	0.1114	0.1114
Servicing time for 100 bytes packet, sec	7.629E-06	7.629E-06	7.629E-06
Packetization delay, sec	0.03	0.03	0.03
Upper number of packets in queue, packets	629.15		
Number of packets per type, packets	n=600	m=1460	P=50
Queue length, bytes	62914.56	2190652.8	75000
Waiting time for service more than, sec	0.00458	0.116	0.1198
Packets length, bytes	60	1000	1000

In WFQ case every packet of type 1 will wait before serving all packets of type 1 existing in the queue as well as proportional existing packets of types 2 and 3. Let us denote with r_1, r_2 and r_3 the number of packets derived from the queue for VoIP, FTP and email in WFQ scheduling. Let us note also "n/r_1" as "c_1" (credit of type 1 packet because r_1 packets will be send from this queue). "c_2" is equal to the "m/r_2", respectively "$c_3=p/r_3$". Waiting time for the new packets per service is calculated (Table 3). Numerical results are calculated for the same initial data as in the case of

PQ above. We also suppose the ratio 50-30-20% of the distribution of packet scheduling for 1-2-3 types of packets in WFQ. The numbers 50-30-20% means that on every 5 packets of type 1, 3 packets from type 2 and 2 packets from type 3 will be transmitted. Some VoIP packets can wait until FTP and email packets will be transmitted depending on the situation when they enter the queue.

$$W_{new1} \geq r_1 c_1 t_1 + r_2 c_2 t_2 + r_3 c_3 t_3 \qquad (8)$$

$$W_{new2} \geq r_1 c_2 t_1 + r_2 c_2 t_2 + r_3 c_3 t_3, \quad W_{new3} \geq r_1 c_3 t_1 + r_2 c_3 t_2 + r_3 c_3 t_3$$

The ratio between c_i is important for the results. When $c_1 > c_2 > c_3$ every new packet of type 1 will wait all packets of type 1. It will also wait all packets of type 2 and 3 below its credit. This means also big VoIP queue and small TCP queues. The case when c_1, c_2 and c_3 become almost equal like in Round Robin (Table 4) is the worst case. Limits on w_1 are more relaxing in comparison to the PQ case.

Table 3. Numerical Results on Delay Bounds in WFQ

$c_1 > c_2 > c_3$			$c_1 > c_3 > c_2$		
w1	0.0293	Unacceptable for VoIP	w1	0.0286	Unacceptable for VoIP
w2	0.0216	Acceptable for FTP	w2	0.0198	Acceptable for FTP
w3	0.0104	Acceptable for email	w3	0.02098	Acceptable for email
$c_2 > c_1 > c_3$			$c_2 > c_3 > c_1$		
w1	0.0219	Unacceptable for VoIP	w1	0.01984	Unacceptable for VoIP
w2	0.0238	Acceptable for FTP	w2	0.02861	Acceptable for FTP
w3	0.0198	Acceptable for email	w3	0.02670	Acceptable for email
$c_3 > c_2 > c_1$			$c_3 > c_1 > c_2$		
w1	0.0198	Unacceptable for VoIP	w1	0.02098	Unacceptable for VoIP
w2	0.0244	Acceptable for FTP	w2	0.01984	Acceptable for FTP
w3	0.0248	Acceptable for email	w3	0.02136	Acceptable for email

Let us consider combined scheduling between PQ for VoIP traffic and WFQ for FTP and email traffic. It this case email traffic will not suffer from FTP traffic. Both of them will suffer from VoIP traffic. We have two cases depending to the ratio between credits for FTP and email traffic c_2

and c_3 accordingly. When $c_2>c_3$ then numerical results are much more acceptable (Table 4).

$$W_{new1} = n_1 t_1, \quad W_{new2} \geq nt_1 + r_2 c_2 t_2 + r_3 c_3 t_3, \text{ and}$$
$$W_{new3} \geq nt_1 + r_2 c_3 t_2 + r_3 c_3 t_3. \tag{9}$$

Table 4. Numerical Results on Delay Bound in RR and Combined PQ and WFQ Approach

Round Robin, $c_1=c_2=c_3$			Combined PQ and WFQ, c2>c3		
		Unaccept. for			Unaccept. for
w1	0.04005	VoIP	w1	0.00457	VoIP
w2	0.04005	Accept. for FTP	w2	0.02708	Accept. for FTP
w3	0.04005	Accept. for email	w3	0.02517	Accept. for email

When $c_3>c_2$ numerical results are of similar value. On the same table, combined scheduling with reservation is applied. In this case *x%* of the capacity of the interface are reserved for data. This amount of capacity guarantees minimal throughput between nodes for data traffic. P_1 and P_2 are the probabilities to use or not this reserved capacity. The delay requirement for VoIP traffic is kept regardless the quantity of the data traffic. Data traffic is also transmitted with acceptable delay requirements. For simplicity we suppose that $r_1 \gg r_2$, $r_2=r_3$ (Table 5). Numerical results show that this scheme degrades slightly the VoIP delay requirements but ensure data traffic transmission.

$$W_{new1} = P_1 r_1 c_1 t_1 + P_2 (r_1 c_1 t_1 + r_2 c_2 t_2 + r_3 c_3 t_3), \tag{10}$$
$$W_{new2} = W_{new3} \geq W_{new1} + r_2 c_2 t_2 + r_3 c_3 t_3, \text{ and } P_1 + P_2 = 1.$$

Table 5. Numerical Results on Delay Bound in Combined PQ and WFQ Scheme and Scheme with x% Reservation

Combined PQ and WFQ, $c_3>c_2$			x% reservation on combined PQ and WFQ		
			p_2	0.1	Data
			p_1	0.9	**VoIP**
			w1	0.00648	**Near accept**.
			p_1 %	0.00457	Accept.
w1	0.00457	**Accept. for VoIP**	p_2 %	0.0236	Not accept.
w2	0.0228	Accept. for FTP	w_2	0.0255	Accept. for FTP
w3	0.0236	Accept. for email	w_3	0.0255	Accept. for email

In a typical DTN connection there is no sense to keep packets in queues in case the delay and loss is increasing. We demonstrate the instruments to change in a flexible way delay bounds in the network. Bundle end-to-end delay can be used for delay bound in the network nodes. This can be done per estimated path in multipathing connection, may depend also on routing protocol. This technique shows that even in a rough interface it is still possible to keep end-to-end QoS requirements for real time services.

6 CONCLUSION

In this paper we demonstrate the capability of the simple algorithm at router interface to shape and police traffic from different traffic sources. Cumulative traffic with different distribution of the packets is studied. Simulation of single server queue with priorities and bounds in waiting times and waiting places is shown. Combination with analytical approach for WFQ and PQ prove the possibility to implement the ideas on traffic shaping on the interface. The proposed combined solution is proven to be the most applicable.

With numerical results, we also show that time bound provides good result in congested interfaces. Because of the simplicity of the proposed algorithm, it can be implemented in real time queues for fast reconfiguration and QoS dynamic management. The approach allows some services to obtain better service conditions on the favor of others.

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