

# Markovian Queueing Model to Estimate Bandwidth Allocation in MPLS over IP Networks

NagamallaMounica , Duddilla Vedashruthi, Moguram Revathi, Sampath Kumar k<sup>+</sup>, T Adilakshmi,  
Department of C.S.E, Vasavi College of Engineering, Ibrahimbag, Hyderabad - 500031, Telangana, India,  
<sup>+</sup>Tata Consultancy Services, SynergeyPak, Hyderabad -500035, Telangana, India

*Abstract—In this paper, we estimate the bandwidth that can be allocated on each link along the Label Switched Path (LSP) of a flow of IP packets, so that an end-to-end packet delay is statistically bounded in Multi-Protocol Label Switching (MPLS) over Internet Protocol (IP) network. We develop discrete event simulations to estimate required bandwidth to be allocated to the link and delay in MPLS over IP Network. This network is modelled as a tandem queue of Label Switched Routers (LSR). The arrival process is assumed to be bursty and strongly correlated and it is generated by 2-MMPP arrival process. The service times are exponentially distributed, as IP packets are of variable in length. The simulation results are more promising and the results are validated with 95% confidence intervals range.*

**Keywords-Bandwidth, MPLS, IP Networks, Markovian Models, Queueing Systems, Quality of Service.**

## I. INTRODUCTION

Typically bandwidth allocation involves reserving part of the transmission rate of the output port of each router along the path of a connection for the traffic flow associated with the connection. The problem of allocating bandwidth under quality of service (QoS) restriction has been analyzed extensively in the literature [1]. Most of the proposed methods calculate the necessary bandwidth so that the packet loss is bounded. Several queueing theoretical papers have presented the packet loss probabilities against finite buffers or the queueing tail probability in infinite buffers, Kim and Shroff model the input traffic as general Gaussian process and derived an approximate expression for the loss probability in a finite buffer system [2]. The review of some of these techniques is as follows.

A well-known approach of allocating bandwidth so as to guarantee end-to-end delay is called equivalent bandwidth, proposed originally for ATM networks [3]. In addition, a survey of various call admission algorithms can be found in Wright [4]. Chara et al [5] compare three methods for the evaluation of end-to-end delays for avionics network architectures. The new generation avionics embedded systems are marked by the characteristics of increased number of integrated functions, increased number of connections between these function and hence the growth of the number of

multipoint communication. Airbus, A 380, has adopted the Switched Ethernet technology with bi-directional links (AFDX, 802.ID) and static routing. The authors are interested in finding out the worst case end-to-end delay of the system.

Koij et al [6] improve on the percentile upper bound on the end-to-end delay as experienced by various real time CBR sources inside a homogenous network. The node model consists of two queues (one for real time and one for low priority traffic), served by a non-pre-emptive head of line scheduler. Assuming that the two queues are independent and the packet lengths are constant, the r-percentile of the upper bound on the end-to-end delay of a K node network is calculated as the arithmetic sum of the r-percentile of the K times convolution of the delay of single M/D/1 node and K times the service time of a single data packet. Experimental evidences show that in access networks (low bit rates) the improvement over the upper bound, due to the later method, can be up to 40% and in the order of tens of milliseconds, whereas in core networks (high bit rates) the upper bound is reasonably accurate in predicting the end-to-end delay.

Vleeschauer et al [7] have described four different approximations to compute the r-percentile of the queueing delay in a heterogeneous network where each node can be represented by an M/G/1 queue. The methods proposed are the most complicated but it works very well but not for very high percentiles because of the discretization and truncation errors which are inherent in the method.

Goyal et al [8] present a way to determine an upper bound on the end-to-end packet delays of flow that passes through a network that employs Guaranteed Rate Scheduling algorithms. Lechoczky and Yeung [9] work on a new queueing theory methodology called real time queueing theory, which allows one to keep track of the deadlines associated with each of the tasks/packets in the system. The authors provide closed form expressions for the proportion of late packets for two scheduling disciplines EDF and FIFO and for constant and uniform distributed deadlines. Simulations illustrate good accuracy of the closed form expressions.

An exact numerical expression of the end-to-end delay in a tandem queueing network can also be obtained by calculation its Laplace transform and then inverting it numerically to obtain delay percentiles. This approach was used by Xiong and Perros, [10] within the context of resource optimization of web services using a tandem queueing network with Poisson arrivals and exponential distributed services.

Bandwidth allocation schemes based on packet loss do not provide guarantees for the end-to-end delay. We are concerned with the calculation of the bandwidth to be allocated at the output port of each router along the path of a connection in an MPLS-enabled IP network. A connection is typically represented by a series of queues forming a tandem queueing network, with each queue representing the output port of a router along the path. Consequently, calculation of the end-to-end delay can be modeled as a problem of evaluating the end-to-end delay in a tandem queueing network.

This paper is organized as follows, in section II, we describe the tandem queueing network under study, and in section III we detail the important simulation algorithms. In section IV, we present the statistical analysis for the calculation of percentiles for validating the simulation results. Section V provides numerical results validating the simulation results, and the conclusion is given in section VI.

## II. TANDEM QUEUEING ANALYSIS

The queueing network under investigation is an open tandem network consisting of N-infinite capacity queues as shown in Figure-1. The queueing network shown below models the delays in an MPLS connection over an IP network. Each queue represents queueing being encountered by the packets of connection at the output port of the each router along the path of the connection. Each queue is assumed to have infinite capacity. The service discipline is in FIFO and the service time is exponentially distributed with the same service rate ' $\mu$ ', which represents the bandwidth allocated to the connection. The propagation delay between the routers is not included in the model since it is fixed.

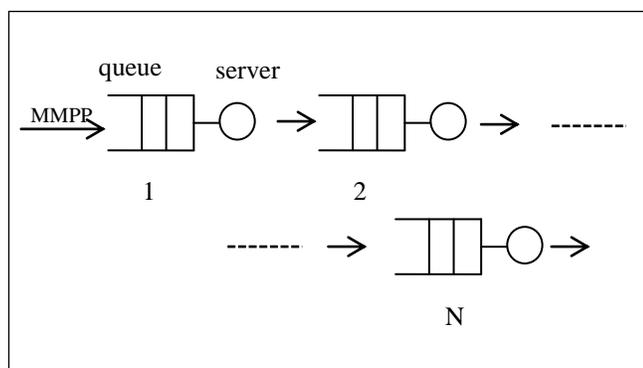


Figure 1 Tandem queue of N routers

In general, in high speed networks packets arrive in a bursty manner and often the successive inter-arrival times are correlated. MMPP is the most widespread traffic model which is capable of capturing burstiness and autocorrelation characteristics commonly present in the Internet traffic [11, 12]. It has been widely used in the literature to capture the

correlation characteristics of multimedia sources in broadband integrated services digital networks [13, 14]. Also, appropriately constructed Markov models using 2-state MMPP appear to be a viable modeling tool also in the context of generating self-similar traffic over time scales as developed in [15-17].

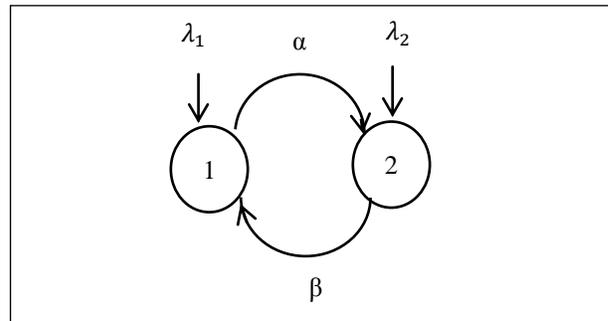


Figure 2 2-state MMPP

MMPP is doubly stochastic Poisson process, where the rate of Poisson process is defined by the state of a two-state continuous-time Markov chain, as shown in Figure 2.

## III. SIMULATION MODEL

As detailed in previous section, the MPLS connection over IP network is being modelled as a Tandem network. The simulation of tandem network is being developed using discrete-event simulation technique. The basic idea is to track the occurrence of the different events in the system that changes the state of the system. In the proposed simulation model, we have six important events as described in Table-1.

Event	Description
1	Completion of the time the arrival process is in state 1 (if it is in state 1)
2	Completion of an arrival during state 1
3	Completion of the time the arrival process is in state 2 (if it is in state 2)
4	Completion of an arrival during state 2
5	Arrival at a queue i
6	Service completions at a queue i

Table-1. Number of Events

The occurrence of the one event may trigger the occurrence of one or more events. In order to track the simulation, a master clock (MC) is maintained. Each event is associated with the time of completion in the future. Also, a random value from an exponential distribution with a given mean can be generated by the below expression.

$$-(\text{mean of exponential distribution}) * \log(\text{random}()) \dots (1)$$

The simulation algorithms for various events are described in the following sections.

A. Arrival Process: end of state  $i:1,2$

Step - 1. Assume , the arrival process is in state-1  
 Step - 2. Generate exponential time 't', using the equation (1) with a mean '1/α'  
 Step - 3. Set the time, that this event will occur in future is 'MC+t', where MC is the current clock time.  
 Step - 4. When the time is reached 'MC+t', the event "end of state-1" has occurred and process shifts to state-2  
 Step - 5. [Now, need to decide how the process will stay in state-2]  
 Step - 6. Repeat step-2, with the mean '1/β' and step-3  
 Step - 7. When the time is reached 'MC+t', the event "end of state-1" has occurred and process shifts to state-1

B. Arrival during state  $i:1,2$

Step - 1. Generate new inter-arrival time of a packet using equation (1) with mean '1/λ<sub>i</sub>'  
 Step - 2. For every new arrival , immediately generate a new inter-arrival time 't' with the same mean and mark its time of occurrence 'MC+t', where MC is the current time  
 Step - 3. Repeat step-2, until the " end of the state " event occurs  
 Step - 4. The newly generated packet joins the first queue  
 Step - 5. If the queue is empty, generate service time using equation (1), with a mean '1/μ', set the service completion time, 'MC+t'  
 Step - 6. If the server is busy and /or there are packets waiting in the queue, the packet will join the queue and no further action will be taken.

C. Service Completion at queue  $i:1,N$

Step - 1. After service completion at queue 'i', the packet moves to the next queue 'i+1', if  $i < N$ .  
 Step - 2. If the queue 'i+1' is empty , generate a service time using equation (1) with a mean '1/μ' and set service completion as 'MC+t', where MC is the current time  
 Step - 3. If the server is busy, and/or there are packets waiting in the queue, the packet will join the queue and no further action will be taken.  
 Step - 4. If the packets waiting in the queue i, then the next packet will start service its service and for this generate a new service time t at queue 'i' using equation (1) with a mean '1/μ'. and the set the service completion time as 'MC+t', where MC is the current time.

D. Main Simulation Algorithm

In this paper, we consider five routers, i.e N=5. In this case, the total numbers of events are eight, service completion at 5 queues, completion of the arrival process in state  $i:1,2$ , and arrival of a packet. However, the arrival process is could be either in 1 or 2, and therefore maximum number of events that can be scheduled in future at any time  $i:1,2$ . In this scenario, the main simulation algorithm is presented below.

Step - 1. All the future events are stored in an array, in this case we have seven events.  
 Step - 2. Check the future events to see which of them will occur next and then advance the clock to that time.  
 Step - 3. Service the events, as described in A,B and C.  
 Step - 4. Repeat Step 2 and 3, for all the events stored in the future events array.  
 Step - 5. When the packet departs from the last queue, calculate the time it spent in the entire network and store it in end-to-end delay array for further processing

IV. STATISTICAL ANALYSIS

A. 95% percentile of the end-to-end delay time

Given a random variable with a probability distribution, the 95<sup>th</sup> percentile is the value of this random variable such that only 5% values of this random variable are greater than itself. Let X indicates the end-to-end delay time in the queueing network, and let  $x_i$  be the end-to-end delay time of  $i^{\text{th}}$  packet. Then the 95<sup>th</sup> percentile of the end-to-end delay

time  $T$  is a value such that  $\text{Prob}[X \leq T] = 0.95$ . Now, suppose that the total number of observations is  $n$ , i.e.,  $x_1, x_2, \dots, x_n$ . To calculate the percentile  $T$ , sort the observation in ascending order. Let  $y_1 \leq y_2 \leq \dots \leq y_n$  be the sorted observations. Then, the 95<sup>th</sup> percentile  $T$  is the value  $y_k$  where  $k = \text{ceiling}(0.95 \times n)$ , where  $\text{ceiling}(x)$  is the ceiling function that maps the real number  $x$  to the smallest integer not less than  $x$ . For instance, if  $n = 50$ , then  $k = 48$ , and the percentile is the value  $y_{48}$ .

**B. Confidence interval of the 95th % of the end-to-end delay time**

A confidence interval provides an indication of the error associated with the estimate. In this section we evaluate the error associated with the simulated end-to-end delay times. In this case we find the 95th percentile. To calculate the confidence interval, we have simulated 50 different percentiles of the end-to-end delay time for the same input values. This has been computed using batch method. We run simulations for one lakh departures, and then grouped results as follows. We divided all the departures into 50 batches and each batch is of 2000 arrivals. For the first 2000 departures in batch 1, we calculate the 95th percentile of the end-to-end delay time and store this number in an array. Repeat this process for the next 2000 arrivals without changing anything in simulation, and so on until we obtained 50 different percentiles. Now, let  $T_1, T_2, T_3 \dots T_n$  be the calculated percentiles for  $n=50$  batches. Then the mean of the percentiles is

$$T_{mean} = \frac{1}{n} \sum_{i=1}^n T_i$$

and the standard deviation ‘S’ is :

$$S = \sqrt{\frac{\sum_{i=1}^n (T_{mean} - T_i)^2}{(n - 1)}}$$

The confidence interval at 95% confidence is given by:

$$\left( T_{mean} - 1.96 \frac{S}{\sqrt{n}}, T_{mean} + 1.96 \frac{S}{\sqrt{n}} \right)$$

**V. NUMERICAL RESULTS**

In every simulation, we assume that there are no packets in the queueing network, i.e the system is empty. But, in this scenario, the initial behavior of the simulation will be affected by this “System empty” condition. In order to eliminate the effects of the initial condition, we run simulation for first 1000

departures, after that started batch method as described in previous section. We run the simulations to obtain the specified percentile of the end-to-end delay of the packets and its confidence interval. We considered the following inputs [18] for the simulations.

Inputs	Values	Probability Distribution
2-MMPP : $\alpha$	0.3	Exponential
2-MMPP : $\beta$	0.5	Exponential
2-MMPP : $\lambda_1$	3	Exponential
2-MMPP : $\lambda_2$	4	Exponential
Percentile	95%	
No.of Departures	101000	
Number of Batches	50	
Service rate: $\mu$	5	Exponential

For the input values as mentioned above table, we run simulations to get the stable system and percentile result with  $\sim 0.10T_{mean}$  and until the confidence interval contain the value 5.75.

To estimate the bandwidth, we run simulations for different values of the service rate  $\mu$  (5.0Mbps, 5.5Mbps, 6.0Mbps and 6.5Mbps). This will represent the bandwidth allocated to the connection, so that the 95% percentile of end-to-end delay ( $T_{mean}$ ) is less than or equal to 4. This has been repeated for 80%, 85%, 90%, 95% and 99% percentile. The results are depicted in the following Figure 3. From the simulation results it is evident that, the required value of bandwidth ( $\mu$ ) to obtain the 95<sup>th</sup> percentile of end-to-end delay less than or equal to 4ms, at  $\mu=6.5$ ,  $T_{mean} = 3.972$ , confidence interval is (3.971,4.152) and Error = 0.18. From the same figure it is evident that, the end-to-end delay increases exponentially with more and more confidence expectations. This indicates, there is a trade-off between the percentile point confidence desired and value of the end-to-end delay as the confidence interval increases with more and more expected confidence in the values.

Figure 4, shows the variation of Error % against Percentile point, for various values of service rate  $\mu$  (bandwidth). As the bandwidth allocation increases, i.e value of  $\mu$ , the network becomes more and more stable; the error % reduces with

increase in bandwidth allocated having a closer confidence interval.

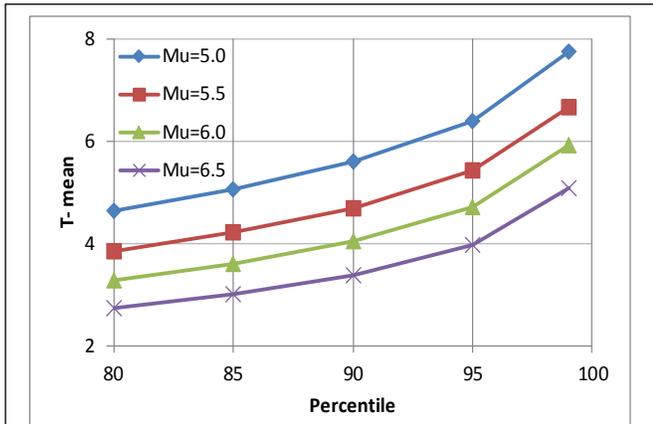


Figure 3 T-mean vs Percentile points

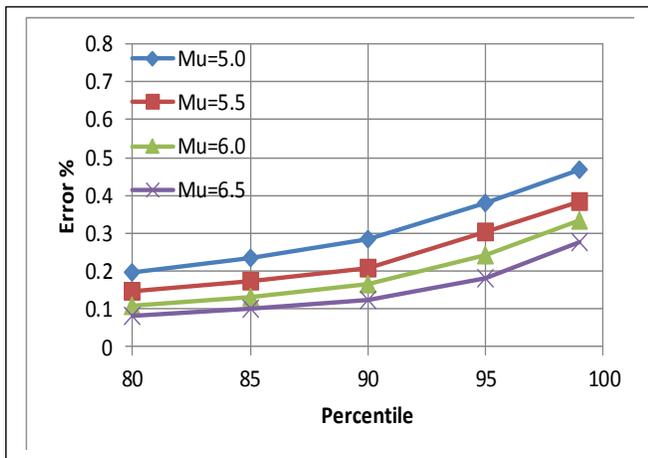


Figure 4 Error % Percentile

confidence interval, 95% and the relative error is 0.18, which is reasonably less.

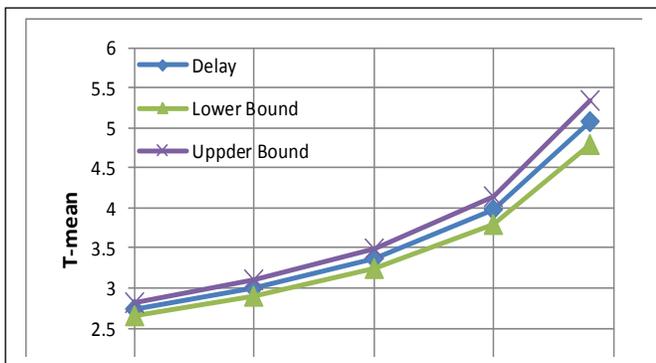


Figure 5 Confidence Intervals, when  $\mu = 6.5$

## VI. CONCLUSIONS

In this paper we proposed a simple simulation model for bandwidth estimation that should be allocated on each link of an MPLS connection in an IP network so that end-to-end delay is bounded statistically. The IP network is being modeled as a tandem queue of five routers. The arrival process is assumed to be bursty and correlated and it is generated by 2-MMPP. The service times are exponentially distributed, as IP packets are of variable in length. From the simulation results, based on the given inputs it is concluded that, at each router one can allocated 6.5 Mbs bandwidth, with the expected average delay of 3.75 ms, 95% of the time this delay is less than or equal to 4 ms.

In this work, it is assumed that all the queues are with unlimited buffer. In future, we would like to explore these simulations with the finite buffer and also buffer bounds against the self-similar traffic.

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