# Improving Throughput in TCP/IP Network

Roopam Gupta

*Abstract*-**With the Internet's evolution from a best-effort network to a network with guaranteed quality of service (QOS), the transport of integrated IP services over asynchronous transfer mode (ATM) is required, which is being adopted as the backbone infrastructure to support any application. Real time applications such as video and audio have become popular. Transmission of real time video has requirements of minimum delay, loss and maximum bandwidth.**

**In this paper we emphasized on to improve channel throughput of TCP traffic in the integrated environment by optimizing delay, queue size and window size. No modification is done at the IP/ATM interface.**

**By optimizing widow size, queue limit and delay, in TCP the packet loss is reduced thus improving channel throughput.**

## I. Introduction

ATM is a technology, which provides voice, data and video services using a single integrated broadband network. ATM's goal is to support diverse services, traffic mixes and to efficiently share resources.

By deploying TCP/IP for Internet also called best effort service, it is required to ensure Internet stability along with fair and efficient allocation of the network bandwidth. Unfair bandwidth allocation to competing network flows arises in the Internet for a variety of reasons, one of which is the existence of applications that do not respond properly to congestion. The impact of emerging streaming media traffic on traditional data traffic is of growing concern in the Internet community. Streaming media traffic is unresponsive to the congestion in a network, and it can aggravate congestion collapse and unfair bandwidth allocation. Thus by optimizing parameters the packet loss should be reduced to improve channel throughput.

## II. Simulation

To evaluate our solutions, we implemented these on the NS-2 simulator and measured the performance.

### A. Overview

The simulation demonstrates that increase in channel throughput can be made in response to iteration of simulation parameters. The packets transferred and packets lost can be visualized in the results [1]shown by the simulator.

### B. Simulation Topology

For the simulation topology, we use the cell known dumbbell topology, with senders and receivers on either side of a single bottleneck link. The bandwidth and delay for each link is given in the simulator parameters:

n1, n2, n5, n6- TCP/IP clients
n3, n4- bottleneck nodes of TCP/IP network
r1, r2 – routers
s1, s2- ATM switch

### C. Simulation Parameters

We consider the system shown is fig1. Simulation Topology. The best effort sources have the following characteristics where packet size = 1000 bytes, Bottleneck Bandwidth = 2 Mbps, Bottleneck delay = 10 ms, Delay of other links =3 ms, Bandwidth of other links =5 Mbps, bandwidth of ATM link = 155 Mbps, Delay of ATM Link =10ms, Type of queuing = Droptail, Simulation time = 5 sec.

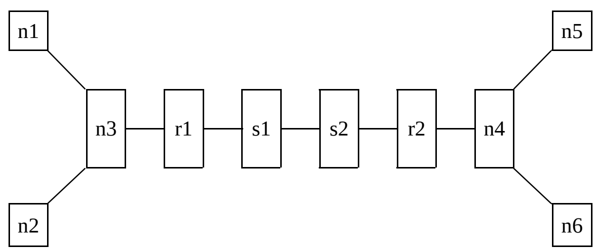



Fig. 1. Simulation Topology

### D. Simulation Results and Graphs

The topology of our simulation is shown is figure 1. The s1 and s2 represents the ATM switches, which are integrated with TCP/IP network. The nodes n1, n2 represents the TCP/IP clients and n3, n4 represents the bottleneck point where most of the packets drops occur. The Switch implements FIFO scheduling and Drop tail queuing. We have used FTP session as the TCP flow, sending from n1 to r1 and n2 to r1. The TCP flows share the bottleneck bandwidth from n3 to r1. For a fair comparison, the end-to end delays for the flows are same. Link delays are also same except the bottleneck link.

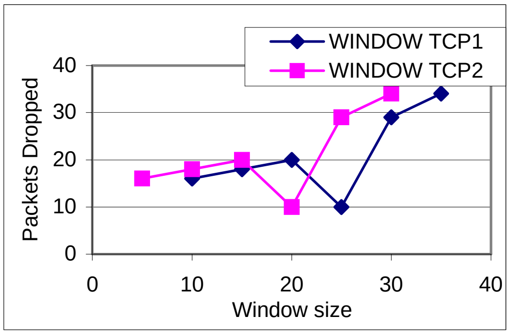
### E. Figures and Tables

Table 1. Parameters for Simulation iteration summarizes that by varying the window size TCP1 and TCP2, delay on the links and queue limit simultaneously the number of packets dropped is obtained. The minimum value of packets dropped fixes the window size as 25 and 20 respectively.

Roopam Gupta is presently working as Reader in University Institute of Technology , Rajeev Gandhi Technical University

TABLE I
PARAMETERS FOR SIMULATION ITERATION

| S.NO | WINDOW TCP1 | WINDOW TCP2 | DELAY on n1-n3 and n2-n3 link | Queue limit | Packets dropped |
|---|---|---|---|---|---|
| 1 | 10 | 5 | 2 | 11 | 16 |
| 2 | 15 | 10 | 5 | 18 | 18 |
| 3 | 20 | 15 | 8 | 25 | 20 |
| 4 | 25 | 20 | 11 | 32 | 10 |
| 5 | 30 | 25 | 14 | 39 | 29 |
| 6 | 35 | 30 | 17 | 46 | 34 |




Graph 1. Packets dropped versus window size

Graph 1. Packets dropped versus window size, represents Table I.

TABLE II.
DELAY V/S PACKETS DROPPED AND ACKNOWLEDGEMENT FOR WINDOW SIZE 25 AND 20

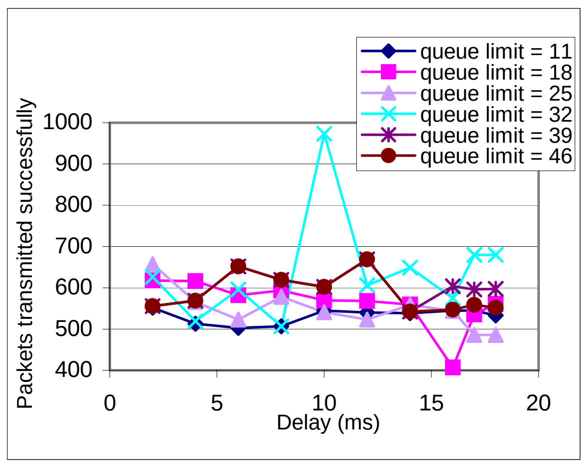
| DELAY IN ms | *QL = 11 | | *QL = 18 | | *QL = 25 | |
|---|---|---|---|---|---|---|
| | Packets dropped | Ack | Packets dropped | Ack | Packets dropped | Ack |
| 2 | 45 | 552 | 33 | 618 | 32 | 657 |
| 4 | 60 | 513 | 36 | 616 | 35 | 566 |
| 6 | 43 | 503 | 30 | 582 | 31 | 523 |
| 8 | 43 | 507 | 33 | 594 | 29 | 578 |
| 10 | 32 | 545 | 27 | 570 | 28 | 541 |
| 12 | 38 | 540 | 34 | 568 | 28 | 523 |
| 14 | 30 | 539 | 27 | 560 | 26 | 559 |
| 16 | 39 | 545 | 32 | 407 | 27 | 544 |
| 17 | 28 | 545 | 28 | 536 | 24 | 486 |
| 18 | 28 | 533 | 27 | 561 | 24 | 486 |
| *QL= Queue Limit, Ack = Acknowledgement | | | | | | |

TABLE III
DELAY V/S PACKETS DROPPED AND ACKNOWLEDGEMENT FOR WINDOW SIZE 25 AND 20

| DELAY IN ms | *QL = 32 | | *QL = 39 | | *QL= 46 | |
|---|---|---|---|---|---|---|
| | Packets dropped | Ack | Packets dropped | Ack | Packets dropped | Ack |
| 2 | 33 | 626 | 27 | 556 | 27 | 556 |
| 4 | 35 | 519 | 26 | 569 | 26 | 569 |
| 6 | 29 | 596 | 23 | 652 | 23 | 652 |
| 8 | 29 | 507 | 23 | 619 | 23 | 619 |
| 10 | 9 | 973 | 25 | 602 | 24 | 602 |
| 12 | 17 | 606 | 23 | 669 | 23 | 669 |
| 14 | 18 | 649 | 17 | 543 | 17 | 543 |
| 16 | 15 | 575 | 18 | 604 | 18 | 547 |
| 17 | 18 | 680 | 17 | 596 | 17 | 559 |
| 18 | 18 | 680 | 17 | 598 | 17 | 552 |
| *QL= Queue Limit, Ack = Acknowledgement | | | | | | |

Table II and III summarizes that by keeping the window size of TCP1 and TCP2 constant, keeping queue limit fix for a set of delay from 2ms to 18ms, the number of acknowledgements and packets dropped is obtained.

Graph 2. Delay v/s Acknowledgement represents Table II and III for varying delay and queue limit, the number of actual packets transmitted.

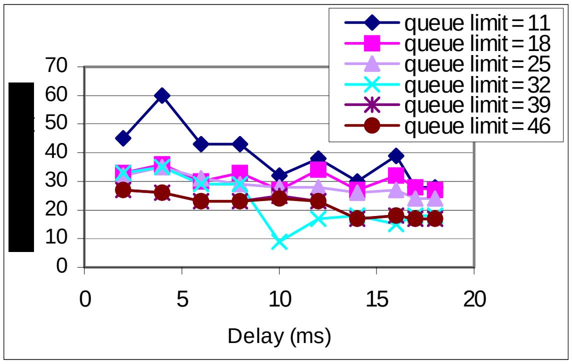



Graph 2. Delay v/s Acknowledgement for Window Size 25 and 20

Graph 3.Delay v/s Packets dropped represents Table II and III for varying delay and queue limit, the number of packets dropped.

It is observed from Figure2. Delay v/s Acknowledgement that for a delay of 10ms and queue limit 32, the number of packets transmitted, successfully, is maximum.

From Graph 3. Delay v/s Packets dropped it is observed that for a delay of 10ms and queue limit 32, the number of packets dropped is minimum.




Graph 3. Delay v/s Packets Dropped for Window Size 25 and 20

Thus we are getting improved channel throughput by optimizing delay, queue limit and window size in TCP/IP network.

## III. Conclusion

In an integrated environment, end-to-end quality of service is required. As TCP does not rely on any explicit feedback from the network to detect congestion and ATM being a connection oriented service provides QOS, Thus by optimizing parameters the packet loss should be reduced to increase channel throughput .We are getting increase in channel throughput by optimizing delay, queue limit and window size in TCP/IP network.

## IV. References

[1] R.N. Shorten, D.J. Leith, J. Foy, and R. Kilduff, “Analysis and design of Congestion Control in Synchronized Communication Networks”, June 20, 2003, Miami, Florida.

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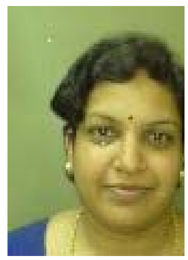
## V. Biographies



**Roopam Gupta** was born on October 17 ,1970. She graduated from the Government Engineering College, Bhopal in 1992 and Post graduated from Maulana Azad College of Technology, Bhopal in 1998.. Her Ph.D was awarded in April 2007. She is presently working as Reader in University Institute of Technology, Rajeev Gandhi Technical University. Her special fields of interest includes networking and wireless communication.