A STUDY ON THE PERFORMANCE OF WIRELESS-CUM-WIRED NETWORK USING DIFFERENT TCP FLAVORS

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ABSTRACT
The Transmission Control Protocol (TCP) which was designed primarily for reliable wire line networks has packet loss which is mainly because of network congestion. The TCP congestion avoidance mechanisms attribute packet loss to congestion in the network. Subsequently, congestion avoidance mechanisms are triggered. These measures are targeted to reduce the load on intermediate links. In the wireless scenario, however, packet loss is primarily because of bit errors and hand overs, which gets mistaken as an indication of network congestion. The resulting measures taken by TCP lead to degradation of throughput. In this paper, the hybrid network so designed consists of two base stations, BS0 and BS1 respectively in cluster 1 and 2. It consists of four wired nodes and six mobile nodes in each cluster. The wireless nodes are interconnected by a wire line which serves as the backbone network. The comparative result of performance for different network scenarios both for Drop Tail and RED are studied using TCP Reno, TCP Vegas and TCP New Reno. The Comparison of performance both in RED and Drop Tail is studied using different TCP flavors. The comparative study on performance under different error rates, link capacity and mobility rates are presented in this paper. The packets are sent from different nodes at different times. The two level hierarchical routing is used to reduce the routing table and calculation time. The window size management is observed for all the TCP connections. Transmission control protocol(TCP) plays an important role in designing a network's performance. Its use in wireless networks has exposed several inadequacies in its operation

KEYWORDS: *Drop Tail, Link Capacity, Queue Size, TCP Flavors, Window Size, Wireless Backbone, NS-2, Throughput.*

I. INTRODUCTION

(Vinton G. Cerf and Bob Kahn, 1974) [1], A Protocol for Packet Network Interconnection", an internetworking protocol for sharing resources using packet-switching among the nodes. A central control component of this model was the Transmission Control Program that incorporated both connection-oriented links and datagram services between hosts. The monolithic Transmission Control Program was later divided into a modular architecture consisting of the Transmission Control Protocol at the connection-oriented layer and the Internet Protocol at the internetworking (datagram) layer. The model became known informally as TCP/IP, although formally it was henceforth called the Internet Protocol Suite. A transfer function (also known as the system function or network function (Bernd Girod, et. al., 2001))[2] is a mathematical representation, in terms of spatial or temporal frequency, of the relation between the input and output of a linear time-invariant system. With optical imaging devices, for example, it is the Fourier transform of the point spread function (hence a function of spatial frequency) i.e. the intensity distribution caused by a point object in the field of view. TCP provides a communication service at an intermediate level between an application program and the Internet Protocol (IP). That is, when an application program desires to send a large chunk of data across the Internet using IP, instead of breaking the data into IP-sized pieces and issuing a series of IP requests, the software can issue a single request to TCP and let TCP handle the IP details. IP works by exchanging pieces of information called packets. A packet is a sequence of octets and consists of a header followed by a body. The header describes the packet's destination and, optionally, the routers to use for forwarding until it arrives at its destination. The body contains the data IP is transmitting.

II. LITERATURE SURVEY

Due to network congestion, traffic load balancing, or other unpredictable network behavior, IP packets can be lost, duplicated, or delivered out of order. TCP detects these problems, requests retransmission of lost data, rearranges out-of-order data, and even helps minimize network congestion to reduce the occurrence of the other problems. Once the TCP receiver has reassembled the sequence of octets originally transmitted, it passes them to the application program. Thus, TCP abstracts the application's communication from the underlying networking details. TCP is utilized extensively by many of the Internet's most popular applications, including the World Wide Web (WWW), E-mail, File Transfer Protocol, Secure Shell, peer-to-peer file sharing, and some streaming media applications. TCP is optimized for accurate delivery rather than timely delivery, and therefore, TCP sometimes incurs relatively long delays (in the order of seconds) while waiting for out-of-order messages or retransmissions of lost messages. It is not particularly suitable for real-time applications such as Voice over IP. For such applications, protocols like the Real-time Transport Protocol (RTP) running over the User Datagram Protocol (UDP) are usually recommended instead (Comer, Douglas E. 2006)[3].

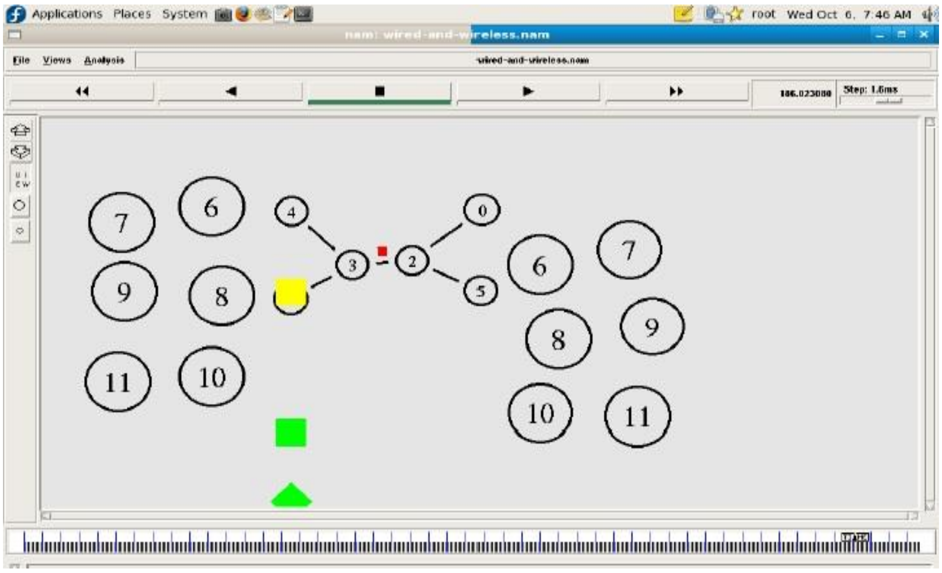

Figure: 1 (NAM Showing Packet drops)

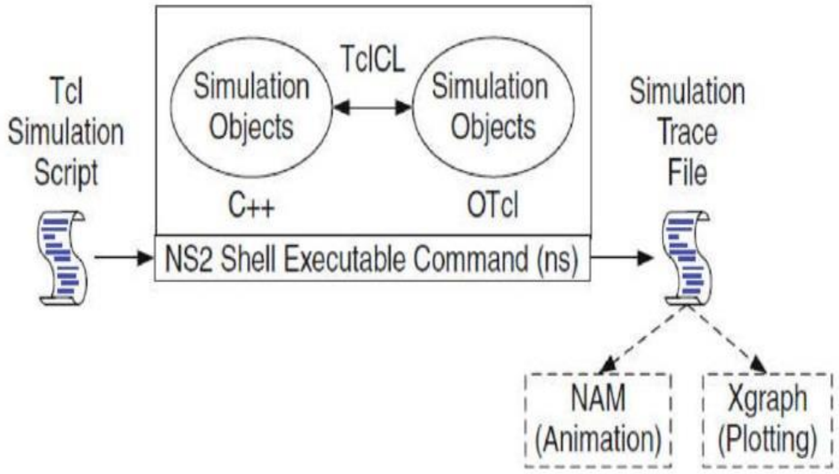

Figure: 2 (NS-2 Simulation)

III. ELABORATIONS

Below is the part of the trace file generated. The 1st field gives the queue type which can be either a (Average) or Q (Current). The 2nd field gives the simulation time i.e. the time at which the queue is generated. The 3rd field gives the size of queue in bytes.

a 15.594 5.18702
Q 15.594 1040
a 15.6262 5.16115
Q 15.6262 0
a 15.6279 5.13541
a 15.6366 5.1098
4.3. RED QUEUE 62
a 15.6383 10.2713
Q 15.6383 1040
a 15.647 15.4071
a 15.6487 25.7043
Q 15.6487 2080

Table: 1 (Average Throughput of TCP New Reno in RED)

Node No.	For No Mobility	For Mobility of $1ms^{-1}$	For Mobility of $5ms^{-1}$
	Average Throughput (Mbs^{-1})		
9	0.38984374999	0.366015625000	0.367773437500
1	0.740489130434	0.731657608695	0.730978260869
0	0.04609375000	0.05442708333	0.05885416666
8	0.14249999999	0.21249999999	0.21625000000
10	0.241443452380	0.21763392857	0.213913690476

Table: 2 (Average Throughput of TCP Reno in RED)

Node No.	For No Mobility	For Mobility of $1ms^{-1}$	For Mobility of $5ms^{-1}$
	Average Throughput (Mbs^{-1})		
9	0.368750000	0.3443359375000	0.37363281250
1	0.731884057971	0.732110507246376	0.73211050724
0	0.06406249999	0.0549479166666	0.06197916666
8	0.15250000000	0.20718751122	0.21781249999
10	0.222098214285	0.245163690476190	0.19605654761

IV. SIMULATIONS AND RESULTS

Below is the part of the trace file generated. The 1st field gives the queue type which can be either a (Average) or Q (Current). The 2nd field gives the simulation time i.e. the time at which the queue is generated. The 3rd field gives the size of queue in bytes.

a	15.594	5.18702
Q	15.594	1040
a	15.6262	5.16115
Q	15.6262	0
a	15.6279	5.13541
a	15.6366	5.1098
4.3. RED QUEUE 62		
a	15.6383	10.2713
Q	15.6383	1040
a	15.647	15.4071
a	15.6487	25.7043
Q	15.6487	2080

Discussion and Analysis: Node 1 has the maximum average throughput in all the mobility cases and node 8 has the minimum average throughput in all the mobility cases. Since the node 1 receives packets from wired node 0 which forms the wired to wired network. The node 8 receives packets from node 1. Even the node 0 starts sending to node 1 too earlier than node 1 to node 8. So it has maximum average throughput. From table B) node 1 has maximum and minimum is yielded by node 8 for no mobility and node 10 when there is a mobility of 5m/s.

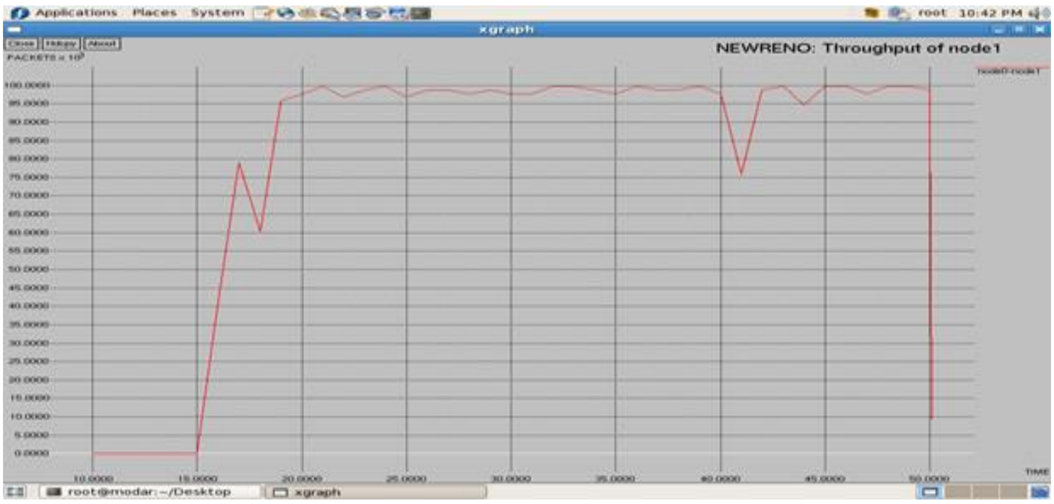

Figure: 3: Throughput of TCP New Reno in RED for wired node 1 when there is no node mobility

Discussion and Analysis: Here the packets are sent from wired node 0 to wired node 1 and they are interconnected by the backbone network. The node 0 starts sending packets to node 1 at time 15.5sec and it is found that there are packet drop at time 18sec (approx.) and at time 41sec (approx.).

Figure: 4: Throughput of TCP New Reno in RED for wireless node 9 when there is no node mobility

Discussion and Analysis: Here the packets are sent from wireless node 11 to wireless node 9 interconnected by a wired backbone network. The node 11 starts sending packets at time 10sec. Initially it has a higher throughput and later drops down because of network traffic in the backbone network formed by the wired nodes 2 and 3.

V. CONCLUSIONS

From the figures of congestion window size shown, it is known that the window size for all the TCP connections change randomly throughout the simulation time as the window size expands or compresses with the buffer size. The size reaches level 0 when the receiving buffer is full. The TCP connected at first has the higher throughput and that connected in the last has least throughput because of the congestion and packet drop. The packet arrival and departure rate shows a linear fashion at later time of simulation because initially there is no packets arrived correctly because of congestion and error in the packets. From the figures of throughput shown, it is fully assured that the wired nodes has higher throughput than the mobile nodes which is because of the wired nodes capability of having less or no packet errors. The comparison of throughputs for RED and Drop tail shows RED is dominant because in RED, the packets before sending are discarded or removed from the window buffer before it is full. Studying all the comparisons, I have found that, the throughput (performance) is best when the link capacity is in the range of 3Mb - 4Mb, the error rate is 10% and the TCP flavor is TCP New Reno. From the plots of graph for throughputs, it is seen that the mobility of mobile nodes has an impact on it.

VI. FUTURE WORK

In future, the number of nodes can be increased and study its throughput, window size, no. of packets dropped and their queue size. Different TCP flavors can be implemented and study their feasibilities for implementation in a network.

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