Enhancement of TCP Performance over MANET

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Abstract - *Mobile ad hoc networks (MANET) plays an important role in today's communication. The Ad hoc networks are a new wireless networking paradigm for mobile hosts. To facilitate communication within the network a routing protocol is used to discover routes between nodes. The goal of the routing protocol is to have an efficient route establishment between a pair of nodes, so that messages can be delivered in a timely manner. Bandwidth and power constraints are the important factors to be considered in current wireless network because multi-hop ad-hoc wireless relies on each node in the network to act as a router and packet forwarder .TCP is used in communication between different applications programmers like file transfer, database management etc. in different system. In this paper we have discussed the existing TCP/IP protocol and some existing extension of this protocol TCP-F, TCP-BUS, ACTP and internet control message protocol. Finally we have given some simulated result using OPNET to find the performance of the throughput using the TCP model.*

***Key Words*: MANETs, TCP, ATCP, Delay, Throughput**

1. INTRODUCTION

Like traditional mobile wireless networks, Ad hoc networks do not rely on any fixed infrastructure. It represents complex distributed system that comprises wireless mobile nodes that can freely and dynamically self-organize into arbitrary and temporary, "ad-hoc" network topologies, allowing people and devices to seamlessly internetwork in areas with no pre-existing communication infrastructure. These nodes can be arbitrarily located and are free to move randomly at a given time, thus allowing network topology and interconnections between nodes to change rapidly and unpredictably [1, 2]. The transport layer is responsible for end-to-end connection establishment, end-to-end data packet delivery, congestion control, and flow control [3, 4]. The real time protocol RTP is a genetic real time transport protocol used in conjunction with UDP. Real time transport control protocol (RTCP) is a protocol used to provide feedback on bandwidth, congestion, and delay to RTP.TCP is a reliable, end to end, connection-oriented transport layer protocol. .A comperes ion of UDP and TCP protocol packet format are shown.

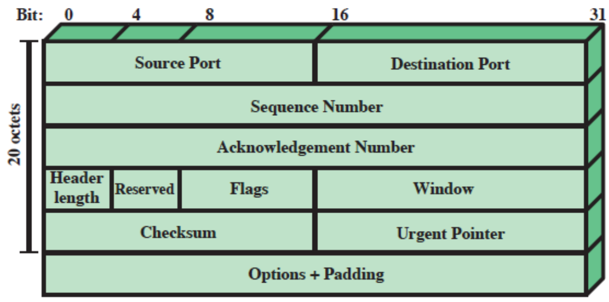

Fig1.TCP header

Wireless systems operate with the aid of a centralised supporting structure such as an access point. This access
In this paper we design a protocol that has the following characteristics. *Improve TCP Performance for Connections set up in ad hoc Wireless Networks.* TCP performance is affected by the problems of high BER and disconnections due to route precomputation or partition. In each of these cases, the TCP sender mistakenly invokes congestion control [5]. The appropriate behaviour in these cases ought to be the following.
High BER: Simply retransmit lost packets without shrinking the congestion window.

Delays due to Route re-computation: Sender should stop transmitting and resume when a new route has been found.

Maintain End-to-End TCP Semantics. We believe that it is critical to maintain end-to-end TCP semantics in order to ensure that applications do not crash.
ATCP treats loss due to packet loss and loss due to congestion. ATCP ensures that the congestion window is recomputed after every new route re-computation. In TCP-F upon receiving the RFN, the source suspends all the packet transmissions and freezes its state, including the Retransmission time out Interval and the congestion window. Eventually the intermediate node that has previously forwarded the RFN learns of a new route to the destination. In TCP-BUS the message is propagated to the source and stops transmission, after receiving an explicit route disconnection message (ERDN) Packet transmission is resumes after a partial path has been re-established[6] .In the subsequence section we have developed a system model

using OPNET environment to evaluate performance of the TCP model.

2. SYSTEM MODEL

In Fig.2 we have developed a simple client server OPNET model to evaluate performance of TCP protocol over ATCP in wireless networks when packet losses occur due to congestion. The simulated network consists of a mobile client, a file transfer protocol (FTP) server connected with wired router by a 10 Mbps link and in ad hoc network the network design is used using the unconnected version of network design, and a wireless router/ server of Wireless LAN. A 50 MB file is transferred using the FTP application from the server to the mobile client via wireless server.

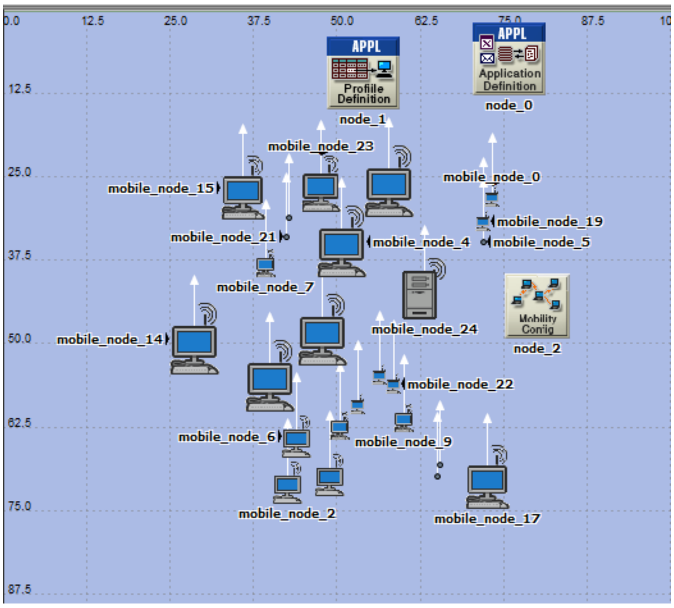

Fig.2 SYSTEM MODEL

We have simulated a network using 24 numbers of nodes in OPNET platform using rapid configuration and within an area of 100m x 100m.

Number of mobile nodes	24
Field size	100m X 100m
Simulation Time	1 hour
Maximum Segmented size	2224 bytes
TCP Packet size	536 bytes
Maximum congestion window	8
Maximum receiver window	8

Network technology simulation parameter

Performance Metrics for the OPNET platform

i)Throughput

The average rate at which the data packet is delivered successfully from one node to another over a communication network is known as throughput. The throughput is usually measured in bits per second (bits/sec). A throughput with a higher value is more often an absolute choice in every network. Mathematically, throughput can be defined by the following formula.

Throughput= (number of delivered packet * packet size)/total duration of simulation

ii) End-to-End Delay

The end-to-end delay is the time needed to traverse from the source node to the destination node in a network. End-to-end delay assesses the ability of the routing protocols in terms of use- efficiency of the network resources.

3. SIMULATED RESULT

A hop to hop delay was introduced by simply delaying ip input by some amount of time at each hop and we measured the time taken to transfer a one-MB file when using plain TCP and when using ATCP.

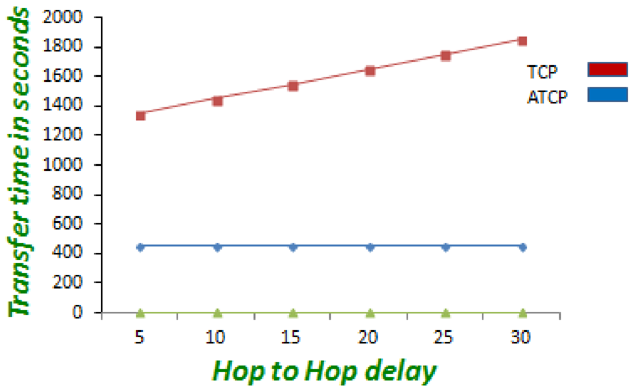

Fig.3

In Fig.3 we plot the transfer time (in seconds) on the axis and the mean hop-by-hop delay on the axis. It is interesting to note that the time taken by TCP to transfer the file increases almost linearly from 900 to 1900 s with increasing hop-by-hop delays. On the other hand, the time taken by ATCP is almost constant at approximately 425 s. It is instructive to perform a rough computation to explain the 425 s transfer time for ATCP. At a BER of, we have an end-to-end probability of packet loss of approximately 3.2% (100 byte packets).

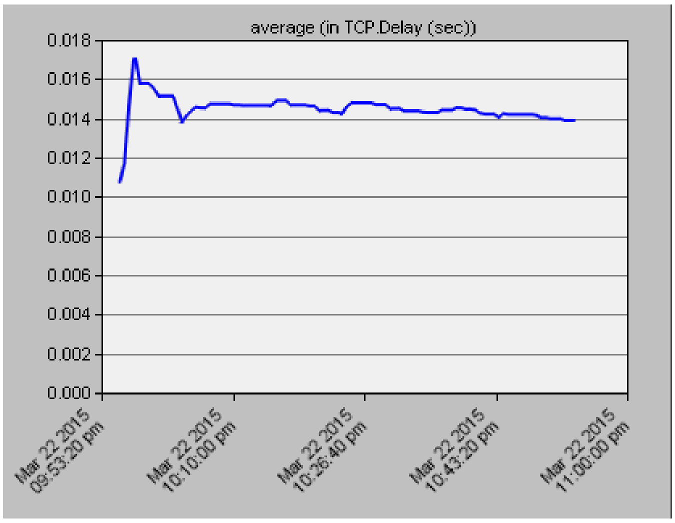

Fig.4

The Fig.4 shows end-to-end delay which is the time needed to traverse from the source node to the destination node in a network. End-to-end delay assesses the ability of the routing protocols in terms of use-efficiency of the network resources.

The Fig.5 shows the average rate at which the data packet is delivered successfully from one node to another over a communication network.

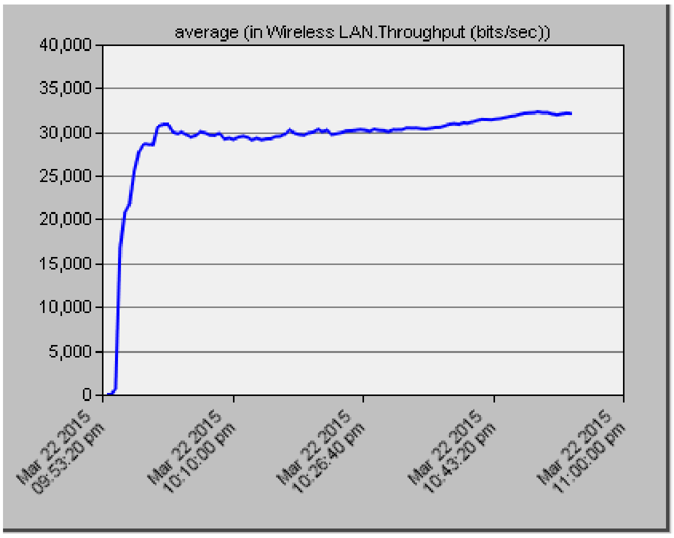

Fig.5

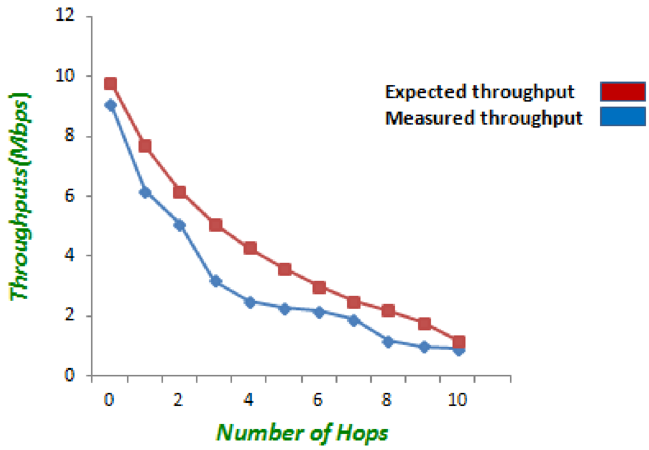

Fig.6

The Fig.6shows the TCP throughput decreases with increasing hop counts, since the first host is said to inject more data than other mobile hosts. Thus the throughput is highest at the sending node. As the number of hops is further increased, the TCP throughput decreases exponentially.

4. CONCLUSIONS

In our approach, we have maintained compatibility of the extended existing TCP model with the standard TCP/IP suite. Therefore, to implement our solution, a thin layer called ATCP (*ad hoc* TCP) between IP and TCP that listens to the network state information provided by the explicit congestion notification messages have been inserted. A comparative study of TCP and ATCP model where we find that hop to hop delay for ATC model is almost constant as well as better than the existing TCP model. The end to end delay and throughput are also obtained using OPNET simulator.

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