A Novel Technique For TCP Throughtput is Revealing in Mobile Adhoc Network (MANET)

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ABSTRACT

The operation of TCP in mobile and wireless communications has been an important issue in recent research years, The impressive growth experience in the area of novel telecommunications during the past decade. TCP/IP is the dominating end-to-end transport layer protocol which provides reliable and secure data Packet transfer together with some other protocols in the protocol stack. and in this research paper proposed a approach in both TCP and UDP base protocols in which we contains an adaptive rate control based technique from destination node to source node. Here destination node copies the estimated rate from the intermediate nodes, the observation is forwarded to the source node through an SACK packet and also adjusted the packet sending rate based on the estimated rate from intermediate node. and not waiting for ACK from Source node. Our proposed technique will be better for packet delivery ratio and also controls the congestion with improvement of throughput.

Keywords

ACK packet, TCP and UDP, MANETs, Wireless, intermediate nodes, destination, TCP performance, UDP performance, congestion controls.

1. INTRODUCTION

Current I.T. trends operating to provide easy and simple measures intended for efficient, reliable and error free communication. The mobile technology is becoming an integral part as it is accessible almost everywhere in the globe. Mobile computing has been in the past few many years forming a new computing environment the fact that mobile computing is constrained by poor resources, highly dynamic variable connectivity and restricted energy sources, the design of stable and efficient mobile information systems has been greatly complicated.

The standardized operation of TCP is not well aligned with the peculiarities of cellular environments. Terminal movement across cell boundaries, leading to handover, is misinterpreted by common TCP implementations as sign of congestion within the fixed network. To handle such congestion, TCP unnecessarily slows down transmission by reducing window sizes, and performing retransmissions, if relevant need arises.

Popularity of the internet services for applications like file transfer, web surfing, e-mail etc. are increasing rapidly. And hence, TCP [1] which is the dominating end-to-end protocol on the internet today carrying more than 90% of the total traffic. It provides a secure and reliable connection between two hosts in a multi-network environment appeared in numerous clones.

Ad-hoc networks are multi-hop wireless networks that can operate without the services of an established backbone infrastructure. The mobile-stations that form the ad-hoc network perform the additional role of routers. Since each station in the network is potentially mobile, the topology of an ad-hoc network can be highly dynamic. Such networks have traditionally been considered to have applications in the military and disaster relief environments. Recent applications in regular wireless packet data environments [2] along with capacity, energy, and range arguments for the use of such networks in tandem with the existing cellular infrastructure [3, 4, 5, 6] have increased the significance of this class of networks.

Over the past decade, a tremendous amount of research has focused on developing network protocols for ad-hoc networks [7, 8]. While the IEEE 802.11 multiple access protocol is primarily considered for the medium access control (MAC) layer, robust and simple protocols such as dynamic source routing (DSR) and ad-hoc on-demand distance vector (AODV) have emerged as the primary mechanisms at the routing layer. With the research at the MAC and routing layers gaining maturity, some researchers have lately shifted focus to the transport layer performance in ad-hoc networks [9, 10, 11]. Since TCP (Transmission Control Protocol) is by far the most used transport protocol in the current Internet, studying TCP's performance over ad-hoc networks is of obvious interest. Recent work in this area has investigated the impact of adhoc network characteristics on TCP's performance and have proposed schemes that help TCP overcome the negative impact of such characteristics as random wireless loss and mobility. While we discuss related work in detail later in the paper, the primary mechanism proposed to handle mobility in related works involves sending an explicit link failure notification (ELFN) to the source from the link failure point. The source, upon receiving the ELFN freezes TCP's timers and state, re-computes a new route to the destination, and either releases the timers and state, or re-starts them from their respective initial values. The set of applications for MANETs is diverse, ranging from small, static networks that are constrained by power sources, to large-scale, mobile, highly dynamic networks. The design of network protocols for these networks is a complex issue. Regardless of the application, MANETs need efficient distributed algorithms to determine network organization, link scheduling, and routing. Military networks are designed to maintain a low probability of intercept and/or a low probability of detection. Hence, nodes prefer to radiate as little power as necessary and transmit as infrequently as possible, thus decreasing the probability of detection or interception. A lapse in any of these requirements may degrade the performance and dependability of the network.

The nature of ad hoc networks poses a great challenge to system security designers due to the Wireless network is more susceptible to attacks ranging from passive eavesdropping to active interfering. The lack of an online CA or Trusted Third Party adds the difficulty to deploy security mechanisms, Mobile devices tend to have limited power consumption and computation capabilities which makes it more vulnerable to Denial of Service attacks and incapable to execute computation-heavy algorithms like public key algorithms.

In MANETs, there are more probabilities for trusted node being compromised and then being used by adversary to launch attacks on networks, in another word, we need to consider both insider attacks and outsider attacks in mobile ad hoc networks, in which insider attacks are more difficult to deal with Node mobility enforces frequent networking reconfiguration which creates more chances for attacks, for example, it is difficult to distinguish between stale routing information and faked routing information.

Without knowing exactly who you are talking with, it is worthless to protect your data from being read or altered. Once authentication is achieved in MANET, confidentiality is a matter of encrypting the session using whatever key material the communicating party agree on. Note that these security services may be provided singly or in combination. In this paper, we propose a improving performance of Adhoc Networks.

2. RELATED WORK

In mobile ad hoc network, [12] congestion is one of the most important restrictions that deteriorate the performance of the whole network. Multi-path routing can balance the load better than the single path routing in ad hoc networks, thereby reducing the congestion by dividing the traffic in several paths. In their paper author presents a new approach Multipath Load Balancing and Rate Based Congestion Control (MLBRBCC) based on rate control mechanism for avoiding congestion in network communication flows. The proposed approach contains an adaptive rate control based technique in which the destination node copies the estimated rate from the intermediate nodes and the feedback is forwarded to the sender through an acknowledgement packet. Since the sending rate is adjusted based on the estimated rate, this technique is better than the traditional congestion control technique.

Their Simulation results for a technique of Multipath Load Balancing and Rate Based Congestion Control (MLBRBCC) is presented. Source node forwards the data packet to the destination node through the intermediate nodes. Upon reception of the data packet, the channel utilization percentage and queue length are estimated at each intermediate node along the destination. Based on these values congestion status and estimated rate are calculated and transmitted towards the destination. By checking the updated values from the intermediate nodes, the destination node determines the estimated rate and it is transmitted as a feedback to the sender. The sender performs rate control based on the estimated rate in the feedback packet.

An ad-hoc network [13] is a collection of wireless terminals that are able to dynamically form a temporary network without any aid from fixed infrastructure or centralized administration. Mobility and the absence of any fixed infrastructure make wireless Mobile Ad-hoc Networks (MANETs) very attractive for time-critical applications. In this paper, they deal with TCP congestion control for Multiple traffic in MANETs. For network simulations, they used ns-3 network simulator considering Ad hoc On-Demand Distance Vector (AODV) and Optimized Link State Routing (OLSR) routing protocols. They present MANET performance considering random waypoint mobility model for different number of nodes by sending multiple traffic in the network. They found that coupling congestion control mechanisms between multiple flows has problems in some cases.

Most of the work for MANET has been done in simulation,

as in general, a simulator can give a quick and inexpensive understanding of protocols and algorithms [14]. So far, there are many simulation results on the performance of MANET, e.g. in terms of end-to-end throughput, round trip time and packet loss. It is known that to fully utilize the capacity of a network, a TCP flow should set its CWL to the BDP of the current path [10]. A path's BDP is defined as the product of the bottleneck bandwidth of the forward path and the packet transmission delay in a round trip. The CWL should never exceed the path's BDP in order to avoid network congestion. In a IEEE 802.11 based MANET, there is interference caused by the MAC layer due to maximum spatial reuse and that by TCP data and ACK along forward and return paths. Because of this the upper bound of BDP is further tightened as kN, where N is the RTHC of the path, and k is a reduction factor due to transmission interference at the MAC layer I-ADTCP tries to remain in congestion avoidance phase at all times by detecting and reacting to incipient congestion. This serves to keep the network uncongested, and reduces the number of congestion related packet losses. Compared to a small fixed CWL setting (1 or 2 packets) as in [15, 16] although dynamically setting the CWL in I-ADTCP has better performance in most cases, but it may not outperform a small fixed CWL in every scenario. This is because the adaptive upper bound may be too high for certain scenarios in a dynamic MANET. Nevertheless, it is generally applicable to any path in MANET, and its performance is usually better than that of a small fixed CWL setting. For IADTCP new field for hop count is required in IP header to deliver the RTHC. For the simulations Optimal value of the CWL derived is specifically for IEEE 802.11 MAC layer protocol.

3. PROPOSED TECHNIQUE

The source node forwards the data packet to the destination through protocol which always selects the shortest path between source and destination, but the shortest path is the easiest broken because of the limited wireless transmission range or the high relative velocity between the intermediate nodes located at the end of the transmission range. Routes failure is caused by the break of the most fragile path here mobility introduced AODV protocol which can prevent link break, increase system packet delivery ratio and reduce times of route request.

In our algorithm the source node is forwards the data packet to the destination node through the intermediate nodes. On reception of the data packet at the intermediate node, queue length are estimated, percentage of channel utilization and node is verified for congestion control. This process is repeating at every intermediate node, and finally the data packet is reaches to the destination node.

We contains an adaptive rate control based technique from destination node to source node here destination node copies the estimated rate from the intermediate nodes, the observation is forwarded to the source node through an SACK packet and also adjusted the packet sending rate based on the estimated rate from intermediate node and not waiting for ACK from Source node. After the reception of the data packet, the destination node checks for the rate information in the packets IP header fields. The sender performs rate control according to the estimated rate obtained from the destination.

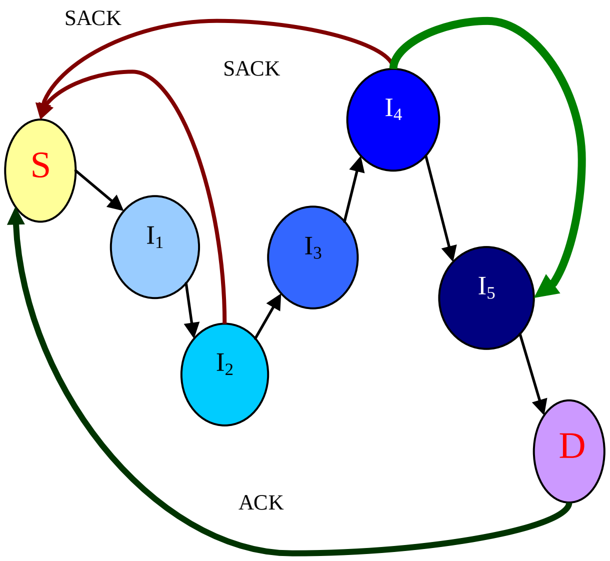

Fig-1 Working Method of our Algorithm

3.1.Route Failure Prediction (RFP)

When a node predicts the a link failure on the progression of signal strengths of packet receptions from the concerned neighbor. The history of signal strengths enables nodes is Maintain to dynamically profile and the speed at which neighbors are moving away by observing the slope of the progression. Some time the *look-ahead time* threshold to trigger a prediction is set to be sufficiently large enough to compute an alternate route before the current path fails, and connections remain unaffected by mobility induced route failures.

When the source receives a failure message of predicted route it issues a new route request, but continues to use the current path until either a new route is computed, and the other option is the current route fails and the normal route error is received. Route requests are suppressed based on the same thresholds used to predict route failures. Hence, a route that is close to failure will not be chosen during any route computation process.

4. PERFORMANCE EVALUATION

We have changed the number of successful data packet transmitted and measured the performance in terms of the number of total packets sent. If the numbers of successful data packet transmitted are limited, the intermediate transmission path of the wireless link becomes unreliable and hence there is huge probability of packet losses and timeouts. As a result, the total number of packets sent is less then successful data packet transmitted. For an optimum number of intermediate nodes, packet transmission is good It is observed from Fig. 1 that the performance is a little bit better. increased number of successful data packet transmitted as it uses modified quick start procedure under this packet loss and timeout conditions. But if the number of successful data packet transmitted increases more, the optimum value of congestion and nodal delay also increases and hence the total number of packets sent falls again.

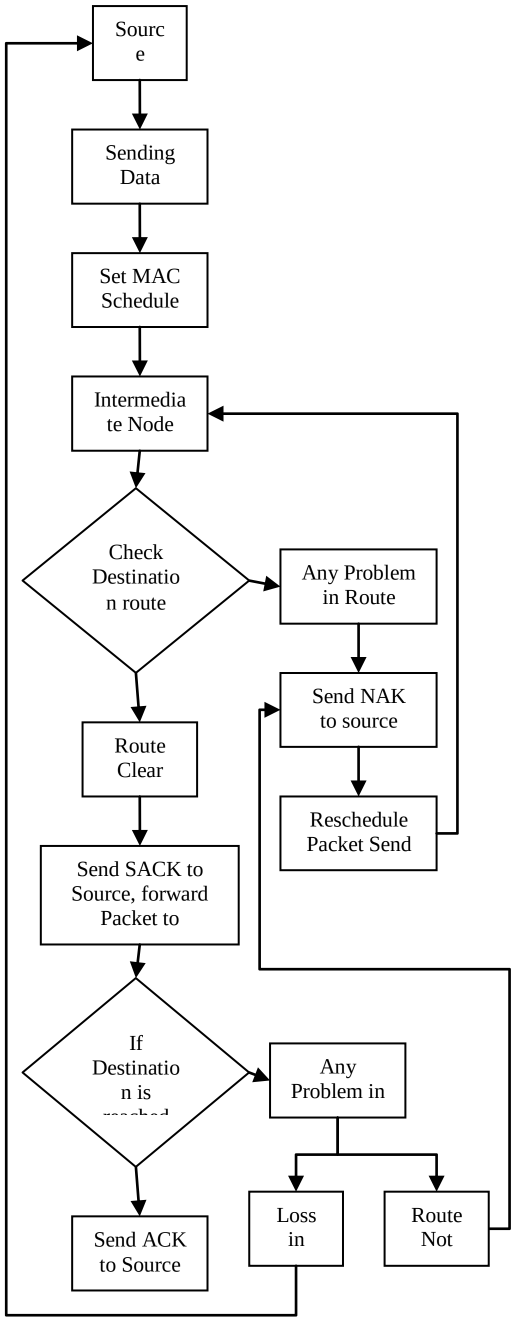

Fig-2 Proposed Algorithm

We have changed the transmitting speed of the data packet sending and measured the total number of packet drops and percentage of packet drops. With the increase of this speed, probability of timeout increases as it performs handoff and wrong estimation of Round Trip Time (RTT) that the number of packet drops increases although the performance is not uniform. But the minimum performance is better than other existing approaches because of its improved functional criteria in case of timeout. The reason behind this behavior depends on the mobile ad-hoc network's topology pattern in simulation like nodes initial and final positions and their parameters.

3.2.Throughput

For network metrics, we collect link utilization as the aggregate link throughput. Throughput is sometimes different from the Throughput, due to Throughput consists solely of useful transmitted traffic, where throughput also include retransmitted traffic. So the collects tool application level end-to-end Throughput is no matter what the transport protocol is employed.

For traffic in long-lived FTP, it measures the transmitted traffic during some intervals in bits per second. For traffic in short-lived web, the Pack Mime HTTP model collects request/response Throughput and response time to measure web traffic performance. Their performance is affected by delay jitter, packet loss rate, and packet delay as well as Throughput. So their Throughput is measured in transmitted packet rate excluding delayed packets and lost packets in excess of a predefined delay threshold.

3.3. Loss Rate

The measure of bottleneck queue loss rate to obtain network statistics. We do not collect loss rates for web traffic and FTP because they are less affected by this metric. For interactive high packet loss rates and streaming traffic, result in the failure of the receiver to decode the packet. In this tool, they are measured during specified intervals. The received packet is considered lost if its delay is beyond a predefined threshold.

4. CONCLUSION

In this paper try to improve the performance over ad-hoc networks have focused only on a subset of the factors affecting TCP and UDP performance. In this research paper we contains an adaptive rate control based technique from destination node to source node here we improve the performance of TCP and UDP from copies of the estimated rate for destination node to the intermediate nodes, the SACK is forwarded to the source node through and also adjusted the packet sending rate based on the estimated rate from intermediate node and not waiting for ACK from Source node.

As short-term solution is also to the lack of effective kernel space transport protocols for high bandwidth-delay product networks. To reach same data transfer rate, UDP needs slightly less CPU time than TCP, and cause slightly less end system delay. And some time we predicting route failures before they occur, reducing the number of route failures and minimizing the latency for route error propagation. So from all this we improves on the throughput performance of a default protocol stack by 80-98%.

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