A CROSS LAYER FRAMEWORK FOR CONGESTION CONTROL USING RECEIVER SIDE FEEDBACK SCHEMES OVER MANETS

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ABSTRACT

Wireless Ad hoc networks are short-lived wireless networks because they are composed to fulfil a certain goal and refrain to exist after fulfilling this goal. Wireless Mobile stations might promptly join or leave the Ad hoc network at any instance, thus Ad hoc networks have a dynamic framework. In most Wireless networking applications, the protocols are categories into distinct modules to compose a protocol stack. Each layer accomplishes benefit of the services bring by the layer directly below it, and also grant service to the layer exactly above it. The source destination transmission is restraint between neighbouring layers with a least possible set of primitives. The layering mechanism facilitates the construction and deployment and contributes the possibility of alternative layer implementations. The characteristics of wireless Ad hoc networks contradict from infrastructure based networks in many ways. Wireless networks have minimum medium scope and maximum bit error rates. Because of the forthright coupling among the physical layer and the upper layers, the conventional protocol stack is not satisfactory for infra-structureless networks. Cross-layer scheme is an effective research pace to enhance infra-structureless or wireless network performance, where information is transferred automatically between different protocol layers. in this paper, the effect of Robust Random Early Detection (RRED) active queue management technique with different TCP variant - TCP-LP, TCP-Cubic, TCP-Westwood and TCP-Compound under varying congested network density is examined on to check the improvement and performance of DSDV under the FTP traffic. The effect of network size on the TCP variants with and without RRED was studied. Experimental studies show that TCP LP and TCP Westwood along with RRED technique perform much better than the others.

Keyword : TCP, DSDV, RRED, MANETs.

1. INTRODUCTION

As the popularity of mobile devices and wireless networks significantly increased, wireless mobile networks has become popular and active field of communication and networks over the years. MANET is the new advance innovation which permits clients to communicate without any physical infrastructure regardless of their position, that's why it is sometime stated as an "infrastructure-less" network. A MANET comprises of various mobile nodes which are connected through wireless links and each movable node acts not only as a host but also as a router to establish a route. The route between the nodes in the network can communicate with several different paths. An ad-hoc network is a self-configuring and adaptive network. It allows the nodes/devices to maintain path by adding and removing the nodes to and from the network. Due to node mobility, the network topology changes rapidly. Due to the major characteristic of MANETs i.e. vigorous topology and lack of centralized management security, MANETs are vulnerable to attacks. Mobile ad-hoc networks offer unique versatility for certain environments and certain applications. Since no fixed infrastructure, including base stations, is prerequisite, they can be created and used anytime, anywhere. Indeed, since all nodes are allowed to be mobile, the composition of such networks is necessarily time varying. Addition and deletion of nodes occur only by interactions with other nodes; no other agency is involved. Such perceived advantages elicited immediate interest in the early days among military, and rescue agencies in the use of such networks, especially under disorganized or hostile environments, including isolated scenes of natural disaster and armed conflict.

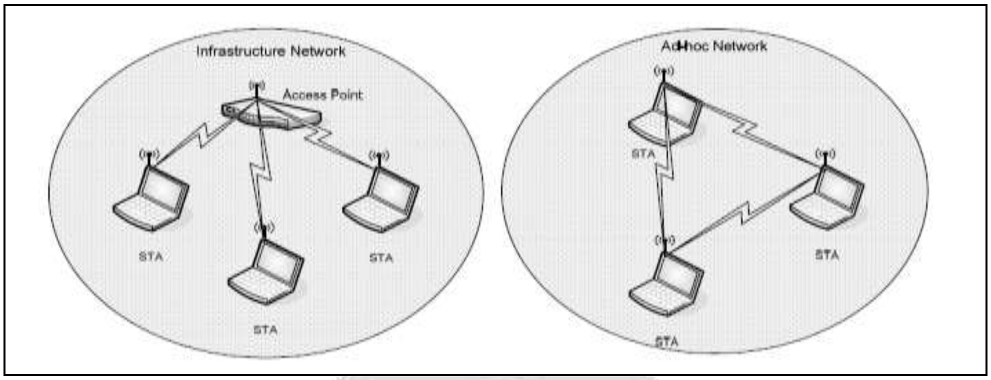

Figure 1 : Mobile Ad hoc Networks.

A. CROSS LAYER DESIGN

In most networking software, the protocols are divided into several modules to form a protocol stack. Each layer makes use of the services provided by the layer directly below it, and also provides service to the layer directly above it. The communication is limited between adjacent layers with a minimum set of primitives. The layering principle simplifies the design and implementation, and provides the possibility of alternative layer implementations. Because of the direct coupling between the physical layer and the upper layers, the traditional protocol stack is not sufficient for wireless networks. Cross-layer design methodology is an active research area to improve wireless network performance, where the information is exchanged between different protocol layers dynamically. Cross-layering is an innovative form of protocol interaction, placed beside strict-layer interactions, which make optimizations possible.

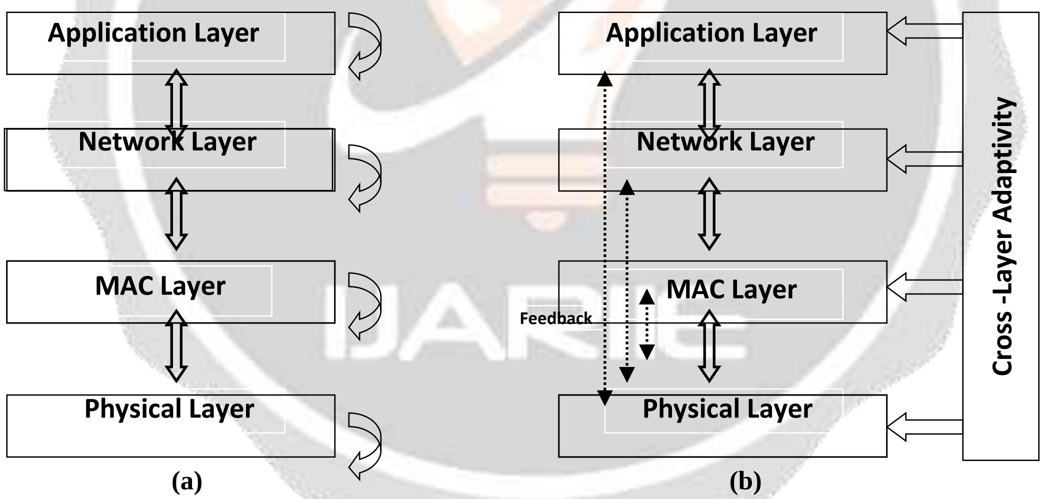

Figure 2. (a) ISO/OSI Model (b) Cross-Layer Model

The differences between the cross-layer model and the layered one are shown in Figure 2. The sharing of information enables each layer to have a global picture of the constraints and characteristics of the network. Moreover, the network protocols are jointly designed and integrated in a hierarchical framework. There are four different approaches of cross-layer design, classifying each approach according to the possible violations of the layered architecture:

- Creation of new interfaces between the levels- New interfaces between adjacent and non-adjacent layers are introduced to enable information sharing at runtime.
- Merging of adjacent layers- Two or more layers are merged to one inseparable superlayer which runs an optimization algorithm and jointly takes care of all the former layer's tasks.
- Completely new abstractions - No layered approach is used in this.
- Vertical calibration across layers- Layer-specific parameters are read and manipulated across all layers.

Cross-layer interactions between layers can be from bottom-to-top or top-to-bottom. In bottom-to-top interaction, higher layers are notified with details related to the network they operate on. Top-to-bottom approach allows lower layers to access information available with the upper layers. There are three main motivations supporting the adoption of cross-layer design in protocol design for MANETs: the need by protocols to be adaptive to network dynamics, to support the requirements specified by the applications and to tackle the energy and security constraints.

B. CROSS LAYER OPTIMIZATIONS

Figure 3 is an examples of cross-layer solutions involving physical, MAC, network and transport layers In this example, cross-layer feedbacks are used to enable state information flow from upper to lower layers or vice versa, while the traditional layered structure is preserved. Physical layer, MAC layer and routing layer together contend for the network resource. The physical layer affects MAC and routing decisions by its transmission power and rate. The MAC layer is responsible for scheduling and allocating the wireless channel, which finally will determine the available bandwidth of the transmitter and the packet delay. This bandwidth and packet delay can also affect the decision at the routing layer to select the link. The routing layer chooses the wireless links to relay the packets to the destination. The routing decision will change the contention level at the MAC layer, and accordingly the physical layer parameters.

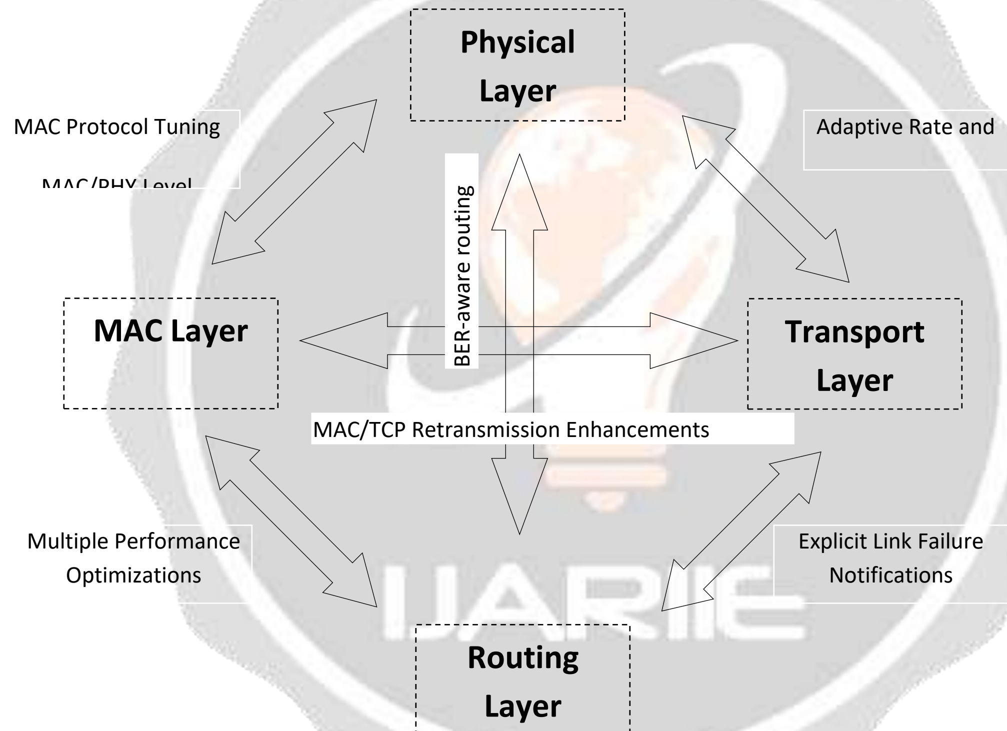

Figure 3 Cross-layer optimizations involving physical, MAC, routing & transport layers

- Physical and MAC cross-layer feedback- In a wireless network, each device has a transmission radius and an interference radius. The relation between the transmission and interference radius depends on the underlying physical layer, and affects the contention level perceived at MAC level.
- Physical and network cross-layer feedback shows the impact of the physical layer on the performance of five different routing protocols for MANETs.
- Physical and transport cross-layer feedback- Power control can often influence the transmission rate of mobile nodes.

C. TRANSMISSION CONTROL PROTOCOL (TCP)

TCP [7,8] is a connection oriented point-to-point protocol. It is a means for building a reliable communications stream on the top of the unreliable Internet Protocol (IP). TCP is the protocol that supports nearly all Internet applications. TCP is used by a large number of IP applications, such as email, Web services, and TELNET. As a connection-oriented protocol, TCP ensures that data is transferred reliably from a source to a destination. Reliability in transmission involves the use of some form of handshake between the sender and receiver. Also, sequence numbers can be used to ensure in-sequence delivery of segments and help to identify lost or corrupted segments. Retransmission can be used to resend lost or corrupted segments. Hence, a retransmission timer is needed to determine when to initiate a resend. For TCP, an adaptive retransmission mechanism is employed to accommodate the varying delays encountered in the Internet environment. When the load offered to the network is more than its capacity to handle, congestion builds up. Congestion can be dealt with by employing a principle borrowed from physics - the law of conservation of packets. The idea is to refrain from injecting a new packet into the network until an old one leaves. TCP attempts to achieve this goal by dynamically manipulating the window size. Figure 4, show how TCP manage network congestion.

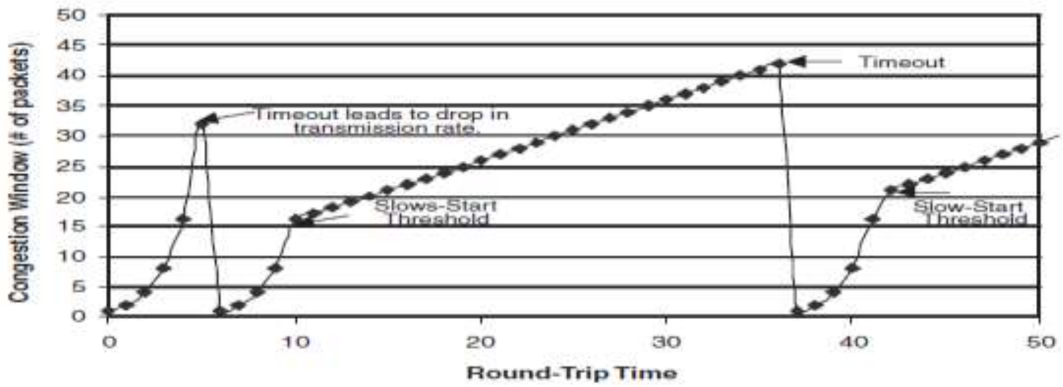

Figure 4. TCP congestion control [32].

D. TCP Timer Management

TCP uses multiple timers (at least conceptually) to do its work. The most important of these is the **retransmission timer**. When a segment is sent, a retransmission timer is started. If the segment is acknowledged before the timer expires, the timer is stopped. If, on the other hand, the timer goes off before the acknowledgement comes in, the segment is retransmitted. In the latter case, the expected delay is highly predictable (i.e., has a low variance), so the timer can be set to go off just slightly after the acknowledgement is expected, as shown in Fig. 5. Since acknowledgements are rarely delayed in the data link layer (due to lack of congestion), the absence of an acknowledgement at the expected time generally means either the frame or the acknowledgement has been lost.

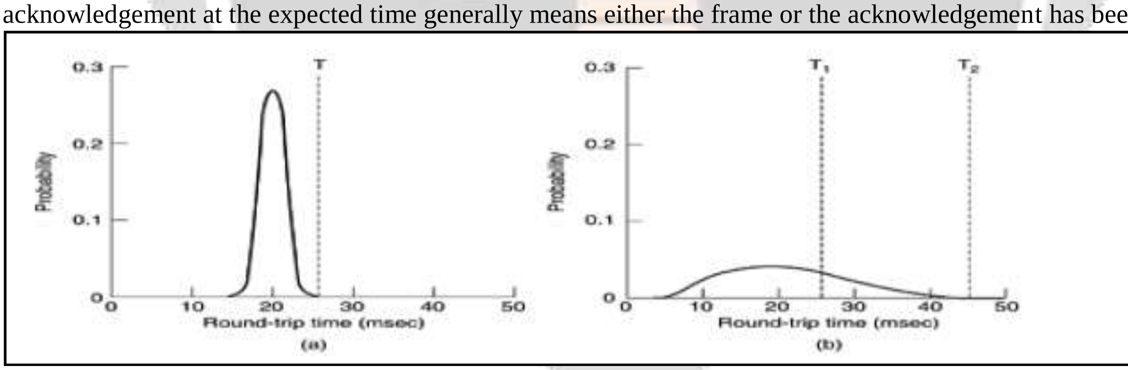

Figure 5: Delay timer [10].

TCP is faced with a radically different environment. The probability density function for the time it takes for a TCP acknowledgement to come back looks more like Fig. 5(b) than Fig. 5(a). Determining the round-trip time to the destination is tricky.

Even when it is known, deciding on the timeout interval is also difficult. If the timeout is set too short, say, *T* 1 in Fig. 5(b), unnecessary retransmissions will occur, clogging the Internet with useless packets. If it is set too long, (e.g., *T* 2), performance will suffer due to the long retransmission delay whenever a packet is lost. Furthermore, the mean and variance of the acknowledgement arrival distribution can change rapidly within a few seconds as congestion builds up or is resolved. The solution is to use a highly dynamic algorithm that constantly adjusts the timeout interval, based on continuous measurements of network performance. The algorithm generally used by TCP is due to Jacobson (1988) and works as follows. For each connection, TCP maintains a variable, *RTT* that is the best current estimate of the round-trip time to the destination in question. When a segment is sent, a timer is started, both to see how long the acknowledgement takes and to trigger a retransmission if it takes too long. If the acknowledgement gets back before the timer expires, TCP measures how long the acknowledgement took, say, *M*. It then updates *RTT* according to t

$$RTT = \propto RTT + (1 - \propto) M \ldots\ldots\ldots\ldots\ldots\ldots\ldots\ldots\ldots \text{(i)}$$

Where α is a smoothing factor that determines how much weight is given to the old value.

Typically α = 7/8

Even given a good value of RTT, choosing a suitable retransmission timeout is a nontrivial matter. Normally, TCP uses βRTT, but the trick is choosing β. In the initial implementations, β was always 2, but experience showed that a constant value was inflexible because it failed to respond when the variance went up [6, 7].

II. RELATED WORK

Muhammad Aamir et al. (2013), propose a scheme of buffer management for packet queues for fixed and mobile hosts over wireless ad hoc environment. For a host, the packet queue is maintained in such a way that an equal buffer space is allocated to each neighbouring source and an allowable extension is also available to each neighbour to avoid any underutilization of resources. The allocation is made in the buffer of a centrally communicating MANET host and it is based on number of packets received in the queue at host's buffer to utilize the buffer space efficiently without any monopolization of some surrounding source. According to this scheme for achieve efficient queuing in the buffer of a centrally communicating host called *QMN* through an active queue management strategy by assigning dynamic buffer space to all neighbouring hosts in proportion to the number of packets received from neighbours and hence controlling packet drop probabilities. The authors simulated this scheme for packet loss ratios and transmission efficiencies in 50-host, 150-host, and 250-host scenarios and compared its performance with Drop Tail and PAQMAN schemes and evaluates the performance of proposed scheme in conjunction with DiffServ implementation of QoS packet markings for VoIP traffic in terms of throughput, packet end-to-end delay, and jitter statistics and found it better as compared to Drop Tail and PAQMAN schemes.

Sanjeev Patel (2013), proposed a model to calculate dropping probability and packet loss for Active Queue Management and shown a comparative analysis of the loss delay product as a new parameter of performance measure obtained from simulation on ns2 for different AQM algorithms. congestion caused due to many unavoidable events like retransmission and increased RTT due to queuing delay. There are a number of mechanisms that have been proposed for IP layer protocols to maintain high throughput and low delay in the network in the absence of feedback from the network. The author has attempted to analyse the AQM algorithm using delay, jitter, throughput, loss rate as measurement parameters. The statistical concept of Random Drop is technique in which a packet randomly selected from all traffic passing through the gateway belongs to a particular connection with a probability matching the connection's proportion to the traffic. Dropping of random packets from connections leads to reduce the total steady state traffic of the gateway. From experiment results, author concludes that RED performs better than other AQM algorithms at low bandwidth.

F. Furqan Doan et al. (2013), propose a mechanism namely WiMAX Fair Intelligent Congestion Control (WFICC) to avoid congestion at the base station. WFICC ensures that the traffic is scheduled in such a way that the base station output buffer operates at a target operating point, without violating the QoS requirements of connections. A detailed simulation study is performed in ns-2 to evaluate the effectiveness of proposed algorithm to meet the QoS requirements of different Class of Services (CoSs). The results have shown that the proposed WFICC algorithm enables the base station to avoid congestion and ensures the provision of QoS of different Class of Services (CoSs) in terms of throughput, fairness and packet delay.

S. Soundararajan et al. (2012), proposed new approach Multipath Load Balancing and Rate Based Congestion Control (MLBRBCC) based on rate control mechanism for avoiding congestion that contains an adaptive rate control based technique in which the destination host copies the estimated rate from the intermediate hosts and the feedback is forwarded to the sender through an acknowledgement packet. Here authors compare the proposed algorithm against explicit rate based congestion control (XRCC) for different performance metrics such as average end-to-end delay, average packet delivery ratio, drops and throughput. In Proposed MLBRBCC, the source host forwards the data packet to the destination through the intermediate hosts. On reception of the data packet at the intermediate host, percentage of channel utilization and queue length are estimated and host is verified for congestion status. After the reception of the data packet, the destination host checks for the rate information in the packets IP header fields. Along with other essential fields, estimated rate is copied to an acknowledgement packet and sent as a feedback to the sender. The sender performs rate control according to the estimated rate obtained from the destination. From Simulation results, the authors conclude that Proposed MLBRBCC has higher better than XRCC.

P. Arivubrakan et al. (2012), focuses on the analysis the performance of AODV and DSR routing protocols under varying range of the transmission in terms of distance and simulation time using Network Simulator (ns2). The performance of AODV & DSR has been analysed with varying transmission range with a distance of 50m, 75m, 100m,125m and 150m under CBR connection of simulation time at 3ms and 5ms using ns2. Data transmitted by a host is received by all the hosts within its communication range. The authors compare theses protocols and found that overall performance of DSR is better than AODV routing protocols at 125m range. Also, they found that the performance of the routing protocol could be enhanced in higher transmission range.

Ipsita Panda (2012), gave overview about various routing protocol for QoS parameter in MANETs. As different applications have different requirements, the services required by them and the associated QoS parameters differ from application to application. For example, in case of multimedia applications time, bandwidth requirement, power requirement, probability of packet loss, the variation in latency (jitter), Route acquisition Delay, Communication Overhead, Scalability are the key QoS parameters, whereas military applications have stringent security requirements. For applications such as emergency search and rescue operations, availability of network is the key QoS parameter.

Makoto Ikeda et al. (2012), evaluate the performance of two routing protocols namely AODV (reactive) and OLSR (proactive) using performance metrics such as CWND (congestion window) and throughput for single and multiple TCP traffic flow over Mobile adhoc network using NS-3 Simulator. TCP has built-in support for congestion control. Congestion control ensures that TCP does not pump data at a rate higher than what the network can handle. Here authors used the TCP-Newreno as TCP congestion control algorithm for congestion avoidance. The authors consider different parameters such as random waypoint mobility model with a randomly chosen speed, uniformly distributed Chosen speed , log-distance path loss model and constant speed delay model; to create scenarios for simulation. From simulation based evaluation, authors found that the number of hosts affects the performance of the network, because of the communication coverage and for AODV the flows have fairness of communication whereas for OLSR, one of the flows is aggressive towards the others.

S. Rajeswari et al. (2012), presented a simulation-based performance evaluation and comparison of three queuing techniques namely First-In-First-Out , *Random early detection* and Weighted Fair Queuing which is implemented against the AEERG protocol for different number of hosts, packet size and pause time . In FIFO queuing (known as first come-first-serve (FCFS) queuing), all packets are placed in a single queue and then processing will begin in the same order on which they arrived. *RED* also known as threshold based queuing discipline, is an active queue management algorithm. It is used for a congestion avoidance algorithm. This algorithm plays an important role by the way of not admitting full queues for processing, reducing the packet delay and loss. It monitors the average queue size and average number of dropped packets based on statistical probabilities. It statistically starts of dropping the packets from flows before it reaches its threshold value. WFQ is a combination of PQ and FQ algorithms. As in the FQ method, all queues are served so that there is no bandwidth starvation, but some queues have more weight in a sense that they receive more service. In other words, a weight is given to each queue to assign different priorities to the queues. From simulation based comparison, the authors noticed that using RED has greatly improved all the performance measures especially with FIFO. The reason is that RED monitors the average queue size and randomly drops packets when congestion is detected.

Taneja and Kush (2012), discussed that the wireless communication links in this network are highly error prone and can go down frequently due to mobility of hosts, less infrastructure and interference. Therefore, routing in MANET is a critical task due to highly dynamic environment. In recent years, several routing protocols have been proposed for mobile ad hoc networks and prominent among them are DSR, AODV and TORA. Authors provides an overview of these protocols by presenting their characteristics, functionality, benefits and limitations and then makes their comparative analysis so to analyze their performance. The results after analysis have reflected in form of parameters selected with respect to low mobility and lower traffic. It has been observed that the performance of all protocols studied was almost stable in sparse medium with low traffic. TORA performs much better in packet delivery owing to selection of better routes using acyclic graph. AODV keeps on improving with denser mediums and at faster speeds. AODV is still better in Route updation and maintenance process. It has been further concluded that due to the dynamically changing topology and infrastructure less, decentralized characteristics, security and power awareness is hard to achieve in mobile ad hoc networks.

Parbu and Subramani (2012), discussed about AODV, DSR and TORA routing protocols. In this authors present overview, characteristics, functionality, benefits and limitations of routing protocols and makes their comparative analysis, so to analyze their performance. The objective is to make observations about how the performance of these protocols can be improved. In this, the performance analysis of various on-demand/reactive routing protocols (DSR, AODV, and TORA) on the basis of routing overhead, end-to-end delay, path optimality performance metrics are consered. From the result conclusion, this comes TORA outperforms in all the given cases.

Parveen Goyel (2012), did simulation study of AODV, OLSR, FSR and LAR using qualnet 4.0 simulator for large scale scenario. This paper presents the scope and performance analysis of these routing protocols under variable pause time and host density. This discusses the efficiency of the above protocols and metrics used are throughput, average end to end delay, average jitter effect. The results show that LAR performs best and OLSR perform worst in all situations.

III. **PROPOSED METHODOLOGY**

This simulation process considered a wireless network of network size consisting of 50 hosts which are placed within a 1500m x 1500m area. FTP traffic is generated among the hosts. The simulation runs for 150 Seconds. Table I shows the important simulation parameters used in the simulation process.

Table: Important Simulation Parameters

Parameter	**Value**
Simulation area	1600m x 500m
Simulation time	200 Sec
Simulation area	1500m x 1500m
Antenna	Omni antenna
No. of hosts	80
TCP –Variants	TCP-LP, TCP-cubic, TCP-Compound and TCP-Westwood.
Routing protocols	DSDV
Traffic	FTP
TCP segment size	1024

- **Simulation Results and discussion**

Simulations are performed for two different proactive routing protocols namely DSDV in a multi hop adhoc network environment. The impact of network size on the performance of above said DSDV protocols under three different TCP variants against RRED is shown with the help of simulation graphs in terms of packet delivery ratio, and throughput. The figure 6 shows the impact of pause time on the throughput for TCP-LP, TCP-Cubic, TCP-Westwood and TCP-Compound against DSDV routing protocols.

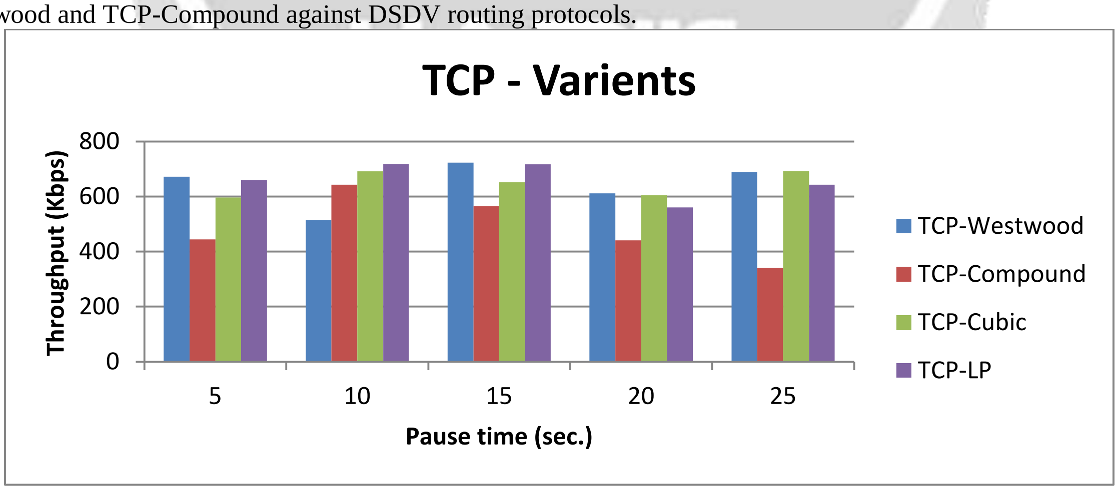

Figure 6: Throughput versus pause time for different TCP-variants.

Figure 7 to 10 shows the throughput when the pause time is varied between 20 to 100 against Robust Random Early Detection mechanism . It is observed that the throughput of TCP-LP under QoS management machanism RRED gives better throughput than other TCP-Variants. The throughput is representative of number of bits received per second.

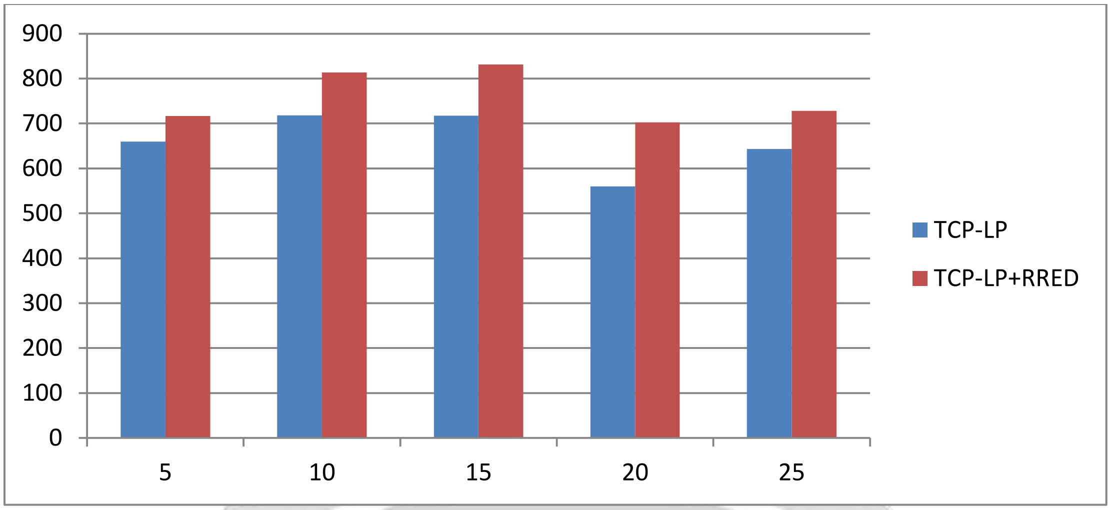

Figure 7. Throughput versus pause time for TCP-LP against RRED.

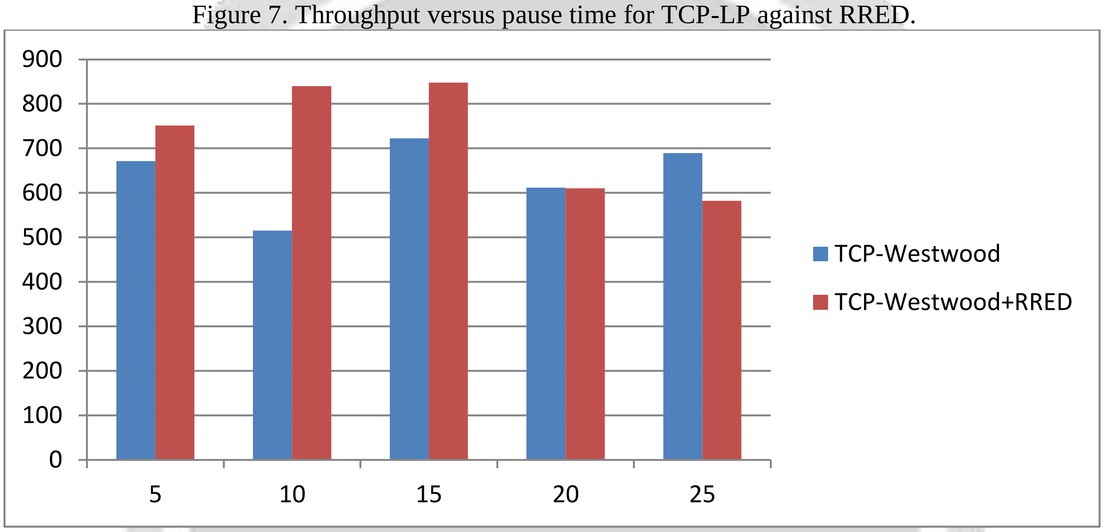

Figure 8. Throughput versus pause time for TCP-Westwood against RRED.

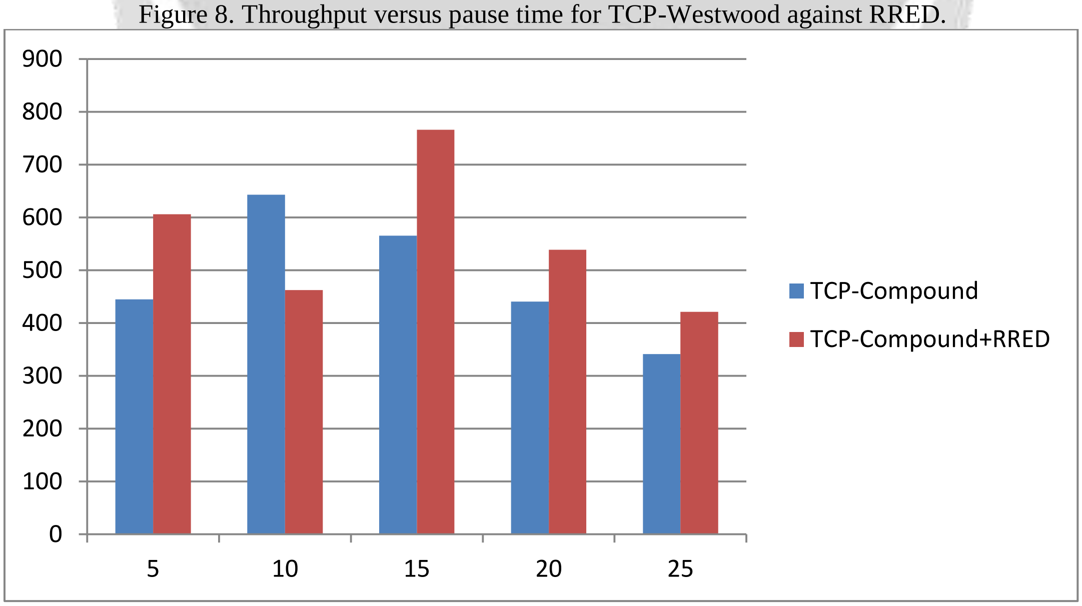

Figure 9. Throughput versus pause time for TCP-Compound against RRED.

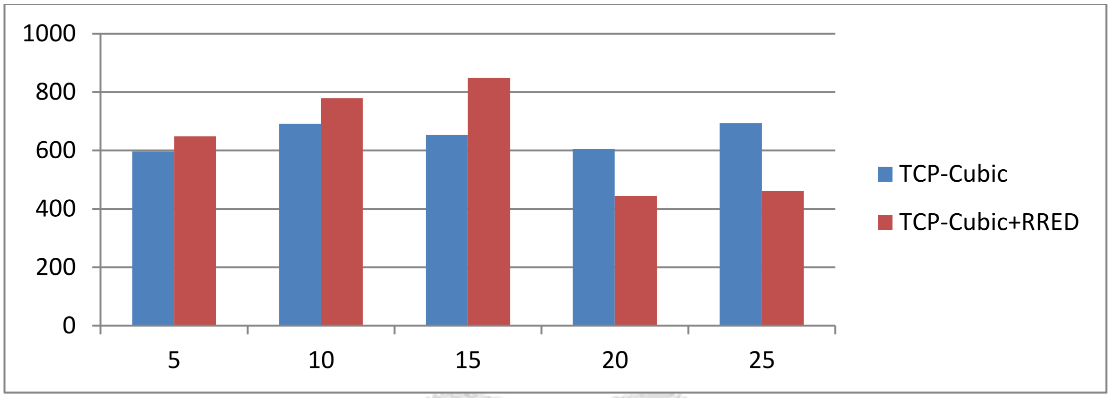

Figure 10. Throughput versus pause time for TCP-Cubic against RRED.

Figure 11 shows the packet delivery ratio for three TCP variants namely TCP-LP, TCP-Cubic, TCP-Westwood and TCP-Compound against DSDV routing protocols when pause time is varied. Simulation results shows TCP-Cubic and TCP-Westwood gives higher performace when pause time is smaller. It is observed that the packet delivery ratio of TCP Westwood is better than both TCP-LP and TCP-Westwood .

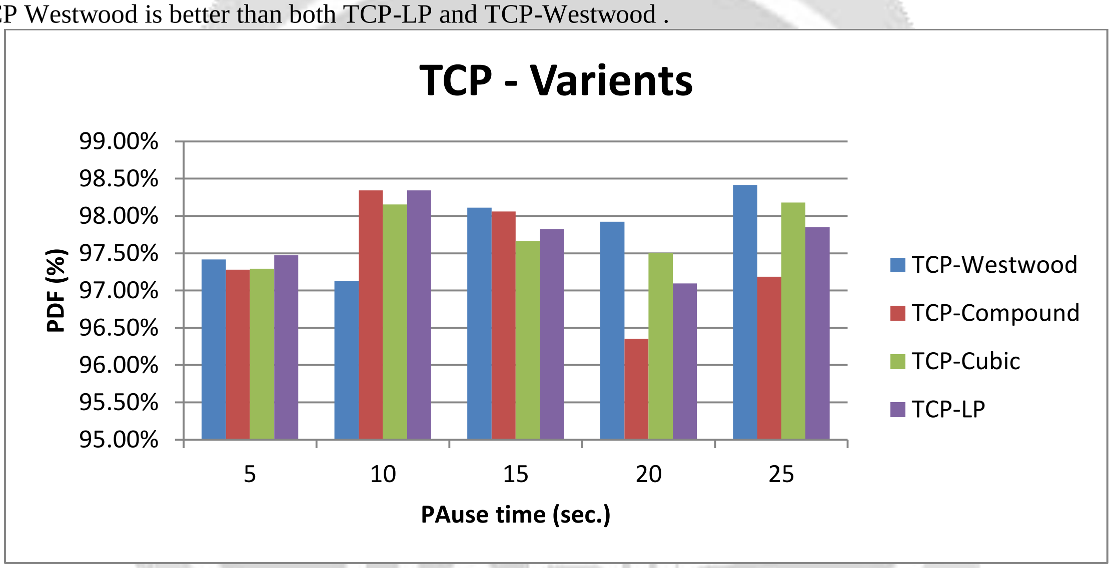

Figure 11. Packet delivery ratio versus pause time for different TCP-variants.

Figure 12 to 15 shows the packet delivery ratio when the pause time is varied between 5 to 25 against Robust Random Early Detection mechanism . It is observed that the packet delivery ratio of TCP-Westwood under QoS management machanism RRED gives better performance than other TCP-Variants.

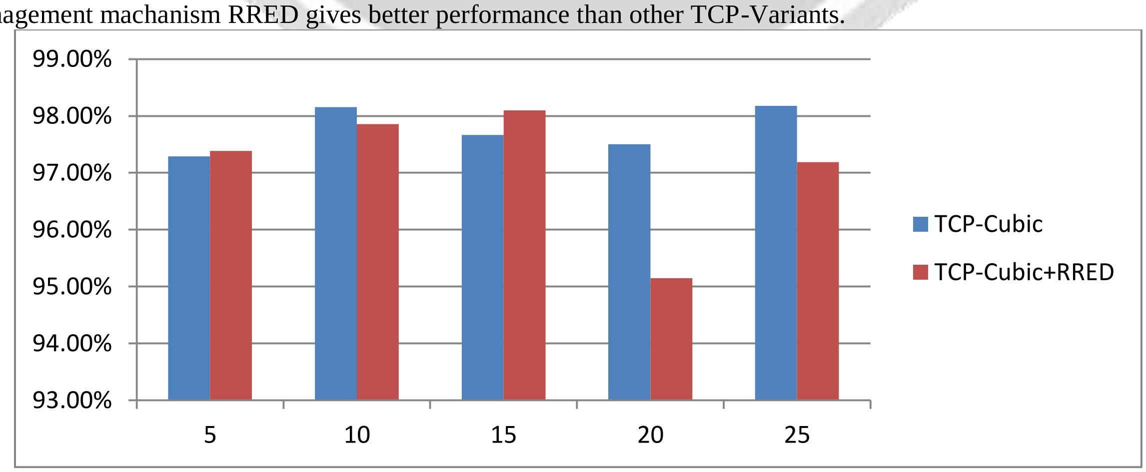

Figure 12. Packet delivery ratio versus pause time for TCP-Cubic against RRED.

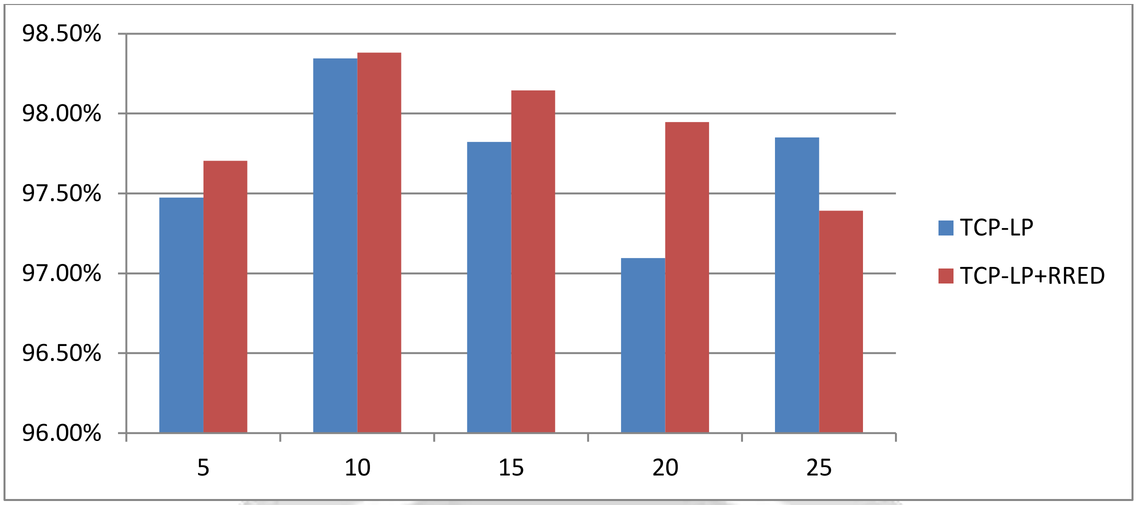

Figure 13. Packet delivery ratio versus pause time for TCP-LP against RRED.

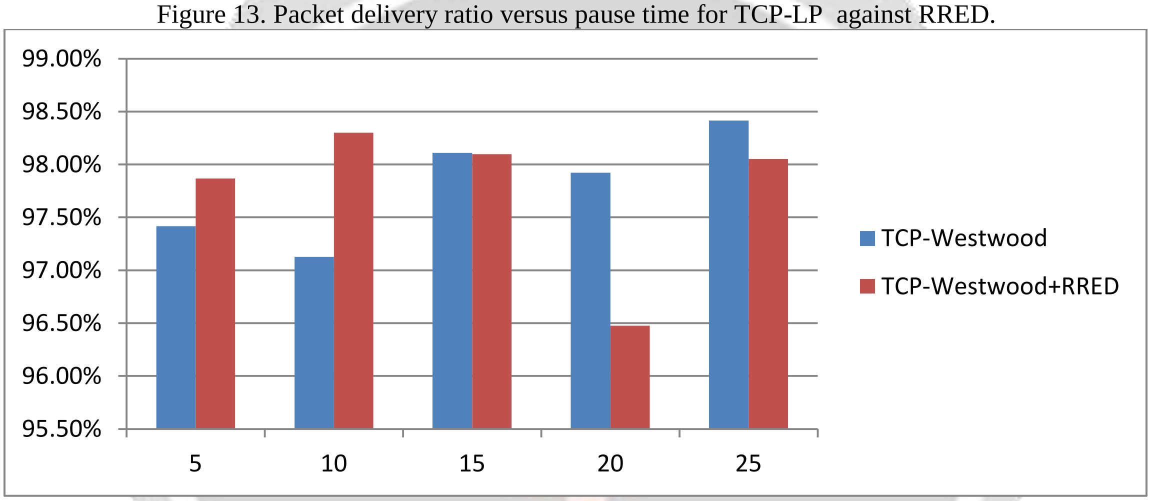

Figure 14. Packet delivery ratio versus pause time for TCP-Westwood against RRED.

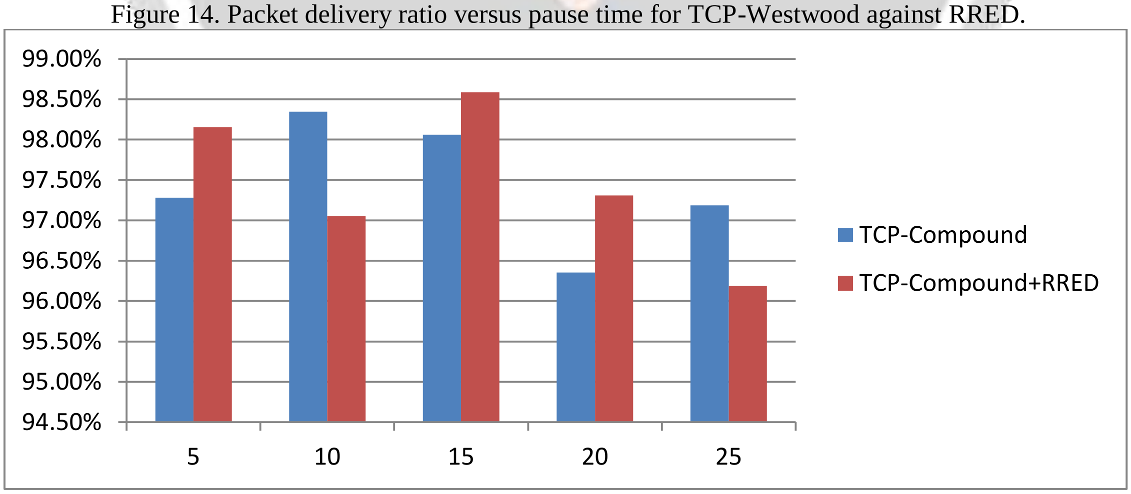

Figure 15. Packet delivery ratio versus number of nodess for TCP-Compound against RRED.

VI. CONCLUSION AND FUTURE WORK

It is a well known fact that TCP can experience significant performance degradation during hand-off, if multiple packet droppings, packet re-ordering or exorbitant hand-off delays occur. We have shown that the reaction on packet droppings and re-ordering is very much related to the implemented TCP version. Different TCP versions react with different types of behavior. In addition, from the perspective of transport layer, we believe that TCP will be on top of the routing protocols for reliable data transmission. Since TCP has its variants, namely TCP-LP, TCP-Cubic, TCP-Westwood and TCP-Compound, we performed the comparison of TCP-LP, TCP-Cubic, TCP-Westwood, and TCP-Compound with and without QoS management technique robust random early detection scheme against DSDV routing protocol under different pause time scenarios. Through simulation, we noted that TCP throughput decreases significantly when host movement cases link failures, due to TCP's inability to recognize the difference between link failure and congestion. From the view of throughput, average delay and packet delivery ratio, TCP SACK and TCP-Westwood are the best congestion control scheme out of selected TCP variants and RRED significantly improve their performance. From this analysis, we found that TCP-Westwood against DSDV routing protocol along with QoS management scheme namely RRED significant improve the perform than other TCP variants in case of increasing Random Packet Loss as well as in case of increasing mobility.

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