# Survey and Simulation based Performance Analysis of TCP-Variants in terms of Throughput, Delay and drop Packets over MANETs

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**Abstract**—Transmission Control Protocol (TCP) is a reliable which maintains end-to-end senentics and it is one of the core of the internet protocol (IP) suite responsible for transmission of internet traffic and very efficient for wired networks. However, experimental analysis and research showed that, TCP's congeation control algorithm performs poorly over Mobile Ad-Hoc Networks with degraded throughputs, fairness services and energy consumption. In this paper we present the reviews and simulated result based comparision of TCP-Variants in terms of throughput, delay and packet drop. Our result based analysis showed that, traditional TCP congestion control protocol fails in MANET whereas Proactive and BWE based TCP-variants gives improved performance in terms of throughput, delay and packet drop. Hence there is motivation for RTT and BWE based investigation and research desirable for adapting TCP over Mobile Ad-Hoc Networks.

**Index Terms**— TCP, Congestion Window, Congestion Control, Congestion Avidence, MANET, Bandwidth Estimate (BWE) , Round Trip Time(RTT), Network simulator, throughput, delay, packet drop.

## 1 Introduction

THE MANET is considered as promising communication network in situations where rapid deployment and self-configuration are essential. In ad hoc networks, nodes are allowed to communicate with each other without any existing infrastructure [3]. Here every node should also play the role of a router. This kind of networking can be applied to scenarios like conference room, disaster management, and battle field communication and places where deployment of infrastructure is either difficult or costly.

The phenomenal growth experienced by the Internet over the last decade has been supported by a wide variety of evolving mechanisms to meet the requirements of emerging, demanding applications. The basic TCP/IP protocol suite has been instrumental in developing today's Internet. In particular, TCP has been successful due to its robustness in reacting dynamically to changing network traffic conditions and providing reliability on an end-to-end basis. This wide acceptance has driven the development of many TCP applications, motivating the extension of this protocol to wireless networks. These networks pose some critical challenges to TCP since it was not originally designed to work in such complex environments, where the level of bit error rate (BER) is not negligible due to the physical medium. High mobility may further degrade the end-to-end performance because TCP reduces its transmission rate whenever it perceives a dropped packet.

This paper focused on studuing the impact of mobile ad hoc environment characteristics on the throughput, delay and packet drop with eight major TCP variants (Tahoe, Reno, New Reno, Westwood, WestwoodNR, Vegas, Sack, Fack) over basic validation network model such as congestion, link failure, signal loss and interference as well as chain and grid network scenario.

The reminder of the paper is organized as follows: brif explation with digram about MANET characteristics in section 2, section 3 presents TCP congestion control algorithms along with congestion window trace diagrams, section 4 explains eight TCP variant with its Psudocode and problems, section 5 gives brif information about tools and technique used to generate results, section 6 presents basic validation model , simulation screnario's and obtained results, finally concluded with the opinion of possible future work.

## 2 Manet Characteristics

The main MANET characteristics that can cause packet loss are: mobility [9] [10], wireless channel, and power constraints.

### 2.1 Mobility:-

All devices in MANET are free to move, so an on-going connection should be kept alive to support mobility. However, once the host moves the topology will change and this may lead to one of the following situations:

#### 2.1.1 Route Failure

They are frequent events due to mobility in MANET. And it needs to re-establish a new transmission route to adjust to the topological changes. Once route failure occurs, it may accompany with a frequent of route change and failure, packet reordering in the sender and the receiver side, not to forget some packets will drop in the way.

#### 2.1.2 Network Partition

A network partition occurs when a node in a MANET moves away from the network thereby causing an isolation of some part of the network by breaking it into two isolated parts. These isolated parts are called partitions.

### 2.2 Wireless Channel:-

Mobile nodes use wireless channel as a medium to send and receive data. However, it is well known that wireless channel is weak, unreliable and unprotected from outside signals. In other words, wireless channel prone the following complexities:

#### 2.2.1 High Bit Error Rate

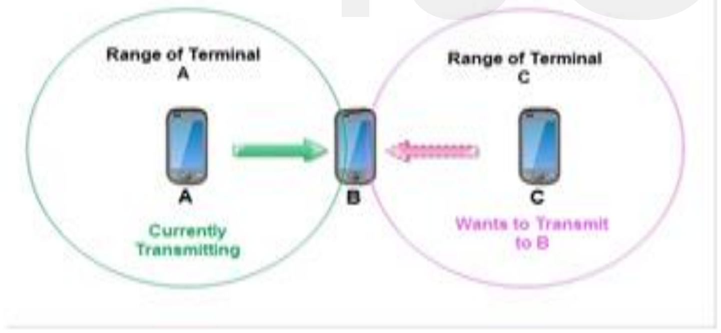
The use of wireless channel is vulnerable to errors due to signal attenuation, interference, obstacles and multipath fading. These errors may generate packet loss within a very short duration or receive corrupted packet at the receiver.

#### 2.1.2 Contention

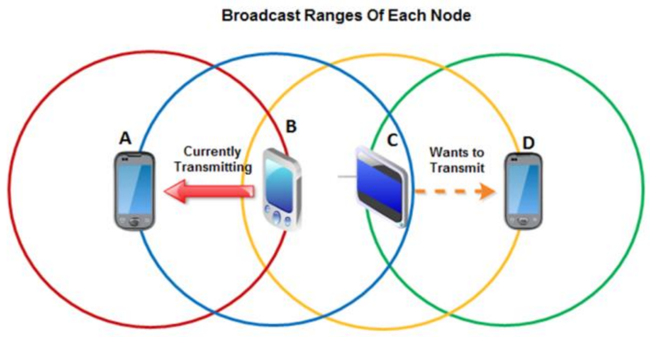
The use of shared wireless channel limits the ability of a node to send packets. Two types of contention:-
(i) Interflow contention which refers to the contention experienced by a node due to transmission by nearby flow.
(ii) Intraflow contention which refers to the contention within same node due to the forward data transmissions and reverses ACK transmission. Generally, contention may produce packet loss and delay.

#### 2.1.3 Hidden & Exposed Terminal

A typical hidden terminal situation is depicted in Fig. 3. Stations A and C have a frame to transmit to station B. Station A cannot detect C's transmission because it is outside the transmission range of C. Station C (resp. A) Is therefore "hidden" to station A (resp. C). Since the transmission areas of A and C are not disjoint, there will be packet collisions at B. These collisions make the transmission from A and C toward B.



Hidden Station Problem



Esposed Station Problem

**Fig. 1 Hidden and Exposed Station Problem**

We show a typical scenario where the exposed terminal problem occurs. Let us assume that node A and C are within B's transmission range, and A is outside C's transmission range. Let us also assume that B is transmitting to A, and C has a frame to be transmitted to D. According to the carrier sense mechanism, C senses a busy channel because of B's transmission. Therefore, station C will refrain from transmitting to D, although this transmission would not cause interference at A. The exposed station problem may thus result in a reduction of channel utilization.

### 2.3 Power Constraints:-

Mobile node is generally a small device with a limited power supply and processing power. However, each node acts as a host and a router simultaneously because not all mobile nodes communicate directly with each other and this requires an additional energy. This imposes a route change or network partition when node energy is low.

## 3 TCP Congestion Control Algorithm

The congestion control algorithms which are used in TCP procol such as slow start , congestion avoidance , fast retransmit and fast recovery [1] [2] [12]are explained with congestion Window trace diagram below.

### 3.1 Slow Start

Slow-start is one of the algorithms that TCP uses to control congestion inside the network. It is also known as the exponential growth phase. Slow start [2] is conducted in the beginning of every TCP connection and its main purpose is to find the maximum available bandwidth at which it can send data without ca using the network to be congested. The Slow-Start, improperly called like this, actually increases exponentially the size of the congestion window. The purpose of Slow-Start is to fill as soon as possible a transmission channel [3]. During the exponential growth phase, slow-start works by increasing the TCP congestion window each time the acknowledgment is received. It increases the window size by the number of segments acknowledged. This happens until either an acknowledgment is not received for some segment or a predetermined threshold value is reached. If a loss event occurs, TCP assumes that it is due to network congestion and takes steps to reduce the offered load on the network. Once the threshold has been reached, TCP enters the linear growth (congestion avoidance) phase. At this point, the window is increased by 1 segment for each RTT. This happens until a loss event occurs [1].

*Slow Start Algorithm:*

```
Initial: CWND = 1;

For (each packet Acked)

CWND++;

Until (congestion event/ CWND>SSTreshHold)
```

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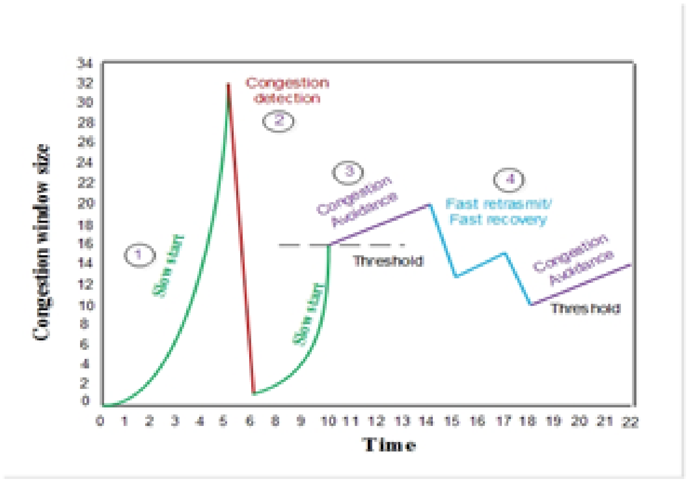

Fig. 2 TCP Congestion Window Trace

## 3.2 Congestion Avoidence

If the receiver window is large enough, the slow start mechanism described in the previous routers in between the hosts will start discarding packets [2]. TCP interprets packet loss as a sign of congestion, and when this happens TCP invokes the Congestion Avoidance mechanism. This variable, SSTreshHold, is the slow start threshold which TCP uses to determine if slow start or congestion avoidance is to be conducted.

Congestion Avoidance Algorithm:

```
/* slow start is over and CWND>SSTreshHold */
every Ack:
CWND = CWND + (1/CWND)
Until (Timeout or 3 DUPACKs)
```

## 3.3 Fast Retransmit

If an out-of-order segment is received TCP generates a so called duplicate acknowledgment. This duplicate acknowledgment is sent immediately from the receiver to the sender indicating that a segment arrived out-of-order, and which segment that was supposed to be received. Since it is not possible to know whether the duplicate acknowledgment was caused by a lost segment or just reordering of segments, the sender waits for three duplicate acknowledgments before retransmitting the segment. If this limit would have been lower, this would increase the chance of reordered segments causing duplicates to be created, and transmitted needlessly. The advantage of this mechanism is that TCP does not have to wait for the retransmission timer to expire. It simply assumes that three duplicate acknowledgments is a good indicator of a lost segment [2].
If the congestion was indicated by duplicate acknowledgements, the TCP sender goes into the Fast Retransmit mode to retransmit what appears to be lost packet without waiting for the retransmission timer to expire. Then, the sender sets the SSTreshHold to half of the current congestion window and the new congestion window to the new SSTreshHold plus the number of received duplicate acknowledgements and enters into the Fast Recovery phase [12].

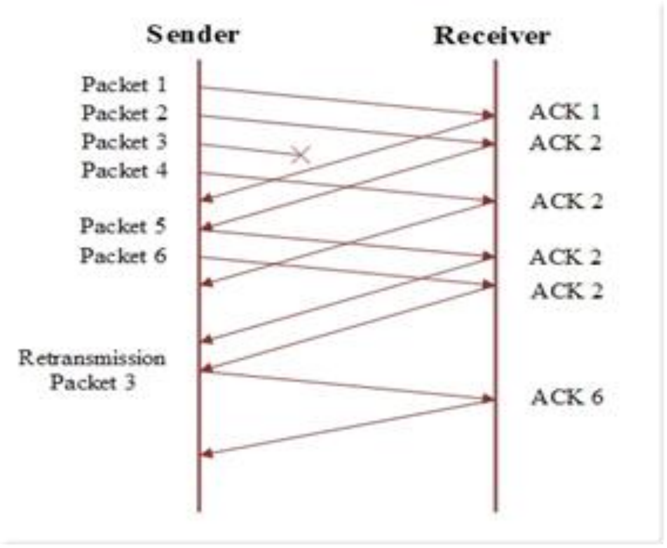

Fig. 3 Fast retransmit

## 3.3 Fast Recovery

During Fast Recovery, the TCP New Reno distinguishes between a "partial" ACK and a "full" ACK. A full ACK acknowledges all segments that were outstanding at the start of fast recovery, while a partial ACK acknowledges some but not all of this outstanding data. After fast retransmit is conduct congestion avoidance and not slow start is performed. This behavior is called Fast Recovery [2] [12]. Fast recovery is an algorithm allows for higher throughput under congestion, especially when using large congestion windows. Receiving three duplicate acknowledgments tells TCP more than the expiration of the retransmission timer. Since the receiving, TCP only can generate duplicate acknowledgments when it is receiving other segments it is an indication that data still flows between the different hosts, and that the congestion is not that severe. By using this approach, skipping the slow start, the TCP does not reduce the transfer rate unnecessarily much.

Fast Recovery Algorithm:

```
/* after fast retransmit; do not enter slow start */
SSTreshHold = CWND / 2;
CWND = SSTreshHold + 3;
each DACK received;
CWND ++;
Send new packet if allow;
After receiving an Ack:
if partial Ack;
Stay in fast recovery;
Retransmit next lost packet (one packet per RTT);
if Full Ack;
CWND = SSTreshHold;
Exit fast recovery;
Invoke Congestion Avoidance Algorithm;
When Timeout:
SSTreshHold = CWND /2;
CWND = 1;
Invoke Slow Start Algorithm;
```

## 4 TCP VARIANTS

### 4.1 TCP Tahoe

Early TCP implementations followed a go-back- model using cumulative positive acknowledgment and requiring a retransmit timer expiration to re-send data lost during transport [6] In the process of trying to improve the original TCP three traffic management mechanisms slow start, congestion avoidance and fast retransmit were introduced to the original TCP and the new TCP version was named as TCP Tahoe.TCP Tahoe is the TCP variant developed by Jacobson in 1988. It uses Additive Increase Multiplicative Decrease (AIMD) algorithm to adjust window size. It means that increases the congestion window by one for successful packet delivery and reduces the window to half of its actual size in case of data loss or any delay only when it receives the first negative acknowledge. In case of timeout event, it reduces congestion window to 1 MSS [7].

- TCP Tahoe uses packet loss probability to adjust the congestion window size.
- During Slow Start phase, TCP Tahoe increases window size exponentially i.e. for every acknowledgement received, it sends two packets.
- During Congestion Avoidance, it increases the window size by one packet per Round Trip Time (RTT) so as to avoid congestion.
- In case of packet loss, it reduces the window size to one and enters in Slow Start phase.

**Psudocode for Tahoe:**

**Initially:**
```
CWND = 1;
SSTreshHoldHold = AWS; /* when TCP begins for the first time*/
```

**New ack received:**
```
If (CWND < SSTreshHold)
/* Slow Start*/
CWND = CWND + 1;
Else
/* Congestion Avoidance */
CWND = CWND + 1/CWND;
```

**Timeout:**
```
/* Multiplicative decrease */
SSTreshHold = CWND/2;
CWND = 1;
```

**Problems**

Due to automatic set back to slow start mode of operation with initial congestion window of one every time packet loss is detected, TCP Tahoe does not prevent the communication link from going empty. Hence this may have high cost in high bandwidth product link.

### 4.2 TCP Reno

TCP Reno retains the basic principle of Tahoe, such as slow starts and the coarse grain retransmit timer [12]. However it adds some intelligence over it so that lost packets are detected earlier and the pipeline is not emptied every time a packet is lost. Reno requires that we receive immediate acknowledgement whenever a segment is received. The logic behind this is that whenever we receive a duplicate acknowledgment, then his duplicate acknowledgment could have been received if the next segment in sequence expected, has been delayed in the network and the segments reached there out of order or else that the packet is lost. If we receive a number of duplicate acknowledgements then that means that sufficient time have passed and even if the segment had taken a longer path, it should have gotten to the receiver by now. There is a very high probability that it was lost. So Reno suggests Fast Re-Transmit. Whenever it receives 3 duplicate ACK's, take it as a sign that the segment was lost, so re-transmit the segment without waiting for timeout. Thus it manages to re-transmit the segment with the pipe almost full. Another modification that Reno makes is in that after a packet loss, it does not reduce the congestion window to 1. Since this empties the pipe. It enters into an algorithm which is called Fast-Recovery.

**Psudocode for Reno:**

**Step 1: Initially**
```
    0<CWND<= min (4*mss, max (2*mss, 4380 bytes))
    SSThreshholdHold = max (CWND/2, 2*MSS)
```
**Step 2: Slow Start Algorithm (Exponential Increases)**
```
    if (receive acks && CWND SSThreshholdHold)
    CWND = CWND+1;
```
**Step 3: Congestion Avoidance Algorithm (Additive increase)**
```
    if (receive ACKs) {
    if (CWND > SSThreshholdHold)
    CWND = CWND + ( segsize * segsize / CWND);
    else
    CWND = CWND + 1}
```

**Step 4: Fast Retransmission**
```
    if (congestion) {
    if (Receive same Acks 3 time or retransmission time out) {
    SSThreshholdHold = CWND/2;
    Retransmit lossed packet
```

**Problems:**

Reno performs very well when the packet losses are small. But when we have multiple packet losses in one window then it doesn't perform too well and its performance is almost the same as Tahoe under conditions of high packet loss. The reason is that it can only detect a single packet loss. If there is multiple packet drops then the first info about the packet loss comes when sender receive the duplicate ACK's. But the information about the second packet which was lost will come only after the ACK for the retransmitted first segment reaches the sender after one RTT. Another problem is that if the widow is very small when the loss occurs then sender would never receive enough duplicate acknowledgements for a fast retransmit and have to wait for a coarse grained timeout. Thus it cannot effectively detect multiple packet losses.

### 4.3 TCP New Reno

The experimental version of TCP Reno is known as TCP New Reno [8]. It is slightly different than TCP Reno in fast recovery algorithm. New Reno is more competent than Reno when multiple packets losses occur. New Reno and Reno both correspond to go through fast retransmit when multiple duplicate packets received, but it does not come out from fast recovery phase until all outstanding data was not acknowledged . It implies that in New Reno, partial ACK do not take TCP out of fast recovery but they are treated as an indicator that the packet in the sequence space has been lost, and should be retransmitted. Therefore, when multiple packets are lost from a single window of data, at this time New Reno can improve without retransmission time out. The retransmitting rate is one packet loss per round trip time until all of the lost packets from that window have been transmitted. It exist in fast recovery till all the data is injected into network, and still waiting for an acknowledgement that fast recovery was initiated.

The critical issue in TCP New Reno [8] is that it is capable of handling multiple packet losses in a single window. It is limited to detecting and resending only one packet loss per round -trip-time. This insufficiency becomes more distinct as the delay-bandwidth becomes greater.

However, still there are situations when stalls can occur if packets are lost in successive windows, like all of the previous versions of TCP New Reno which infer that all lost packets are due to congestion and it may therefore unnecessarily cut the congestion window size when errors occur.There are [5] some steps of congestion control for New Reno transmission control protocol.

**Psudocode for New Reno:**

```
Step 1: Initially
    0<CWND<= min (4*mss, max (2*mss, 4380 bytes))
    SSThreshholdHold = max (CWND/2, 2*MSS)
Step 2: Slow Start Algorithm (Exponential Increases)
    if (receive acks && CWND SSThreshholdHold)
    CWND = CWND+1;
Step 3: Congestion Avoidance Algorithm (Additive increase)
    if (receive ACKs) {
    if (CWND > SSThreshholdHold)
    CWND = CWND + (seg_size * seg_size / CWND);
    else
    CWND = CWND + 1}
Step 4: Congestion Detection Algorithm (Multiplicative Decrease): Fast Retransmission and Fast Recovery
    if (congestion) {
    if (Receive same Acks 3 time or retransmission time out) {
    SSThreshholdHold = CWND/2;
    if (Retransmission time out) {
    CWND = initial;
    exit and call Slow Start step;
    else /* Receive same Acks 3 time*/
    CWND = SSThreshholdHold;
    exit and call congestion avoidance step}}}
```

**Problems:**

Problem with New Reno is that the constraint of retransmitting at most one loss packet per RTT results in substantial delay in retransmitting the later dropped packets in the window. Thus the available bandwidth is not effectively utilized.

### 4.4 TCP Selective Acknowledgment (Sack)

The Selective Acknowledgment (SACK) mechanism is an extension to Transmission Control Protocol's (TCP) ACK mechanism, allows a data receiver to explicitly acknowledge arrived out-of-order data to a data sender. When using SACKs, a TCP data sender need not retransmit SACKed data during the loss recovery period. Previous research showed that SACKs improve TCP throughput when multiple losses occur within the same window. The success of SACK-based loss recovery algorithm is proportional to the SACK information received from the data receiver [9].

With selective acknowledgments, the data receiver can inform the sender about all segments that have arrived successfully, so the sender need retransmit only the segments that have actually been lost [6]. TCP with 'Selective Acknowledgments' is an extension of TCP Reno and it works around the problems face by TCP Reno and TCP New-Reno, namely detection of multiple lost packets, and re-transmission of more than one lost packet per RTT. SACK retains the Slow-Start and Fast Re-Transmit parts of Reno. It also has the coarse grained timeout of Tahoe to fall back on, in case a packet loss is not detected by the modified algorithm. TCP-Sack requires that segments not be acknowledged cumulatively but should be acknowledged selectively. Thus each ACK has a block which describes which segments are being acknowledged. Thus the sender has a picture of which segments have been acknowledged and which are still outstanding. Whenever the sender enters fast recovery, it initializes a variable pipe which is an estimate of how much data is outstanding in the network, and it also set CWND to half the current size. Every time it receives an ACK it reduces the pipe by 1 and every time it retransmits a segment it increments it by 1. Whenever the pipe goes smaller than the CWD window it checks which segments are not received and send them. If there are no such segments outstanding then it sends a new packet. Thus more than one lost segment can be sent in one RTT.

**Problems:**

The biggest problem with SACK is that currently selective acknowledgements are not provided by the receiver to implement SACK we'll need to implement selective acknowledgment which is not a very easy task and Requires modification to the acknowledgement procedures at both sender and receiver sides. Energy consumption is higher then new reno due to additional overheads.

### 4.5 TCP Westwood

TCP Westwood proposes an end-to-end bandwidth estimation algorithm based on TCP Reno. TCP Westwood implements slow start and congestion avoidance phases as TCP Reno, but instead of halving the congestion window size as in TCP Reno when congestion happens, TCP Westwood adaptively estimates the available bandwidth and sets the congestion window size and slow start threshold accordingly to improve the link utilization. In TCP Westwood, packet loss is indicated by the reception of 3 duplicated acknowledgements (DUPACKs) or timeout expiration. When 3 DUPACKs are received, TCP Westwood sets SSTreshHold and CWND as follows:

**Psudocode for Westwood:**

```
if (n DUPACKs are received)
if (CWND>SSThreshhold) /* congestion avoid. */
SSThreshhold = BWE*RTTmin;
CWND = SSThreshhold;
endif
if (CWND<SSThreshhold) /*slow start */
SSThreshhold= BWE*RTTmin
if (CWND > SSThreshhold)
CWND = SSThreshhold
endif
endif
endif
```

Where the seg_size is the length of the TCP segments and RTTmin is the minimum RTT experienced. BWE is the estimated available bandwidth. It is assumed in TCP Westwood that when 3 DUPACKs are received in the congestion avoidance phase, the available bandwidth is fully utilized. So the values SSTreshHold and CWND should reflect the estimated bandwidth (BE).

If a packet loss is indicated by timeout expiration, TCP Westwood sets SSTreshHold and CWND as follows:

```
if (coarse timeout expires)
if (CWND>SSThreshhold) /* congestion avoid. */
SSThreshhold = BWE*RTTmin;
if (SSThreshhold < 2)
SSThreshhold = 2;
CWND = 1;
else
CWND = BWE*RTTmin;
endif
endif
if (CWND<SSThreshhold) /* slow start */
SSThreshhold = BWE*RTTmin;
if (SSThreshhold < 2)
 SSThreshhold = 2;
CWND = 1;
else
CWND = BWE*RTTmin;
endif
endif
endif
```

This sets the CWND to 1 and SSTreshHold to BWE after the timeout event and then the TCP Reno behavior continues. In TCP Westwood, the setting of SSTreshHold and CWND is based on the bandwidth estimation, which is obtained by measuring the rate of the acknowledgments and collecting the information of the amount of packets delivered to the receiver in the ACK. Samples of bandwidth are computed as the amount of packet delivered divided by the inter-arrival time between two ACKs. Those sample bandwidth estimates are then filtered to achieve an accurate and fair estimation. TCP Westwood modifies the Additive Increase and Multiplicative Decrease (AIMD) in TCP Reno and adaptively sets the transmission rates to remove the oscillatory behaviour of TCP Reno and to maximize the link utilizations. But this also causes TCP Westwood to degrade the performance of TCP Reno connections when they coexist in the network [12].

**Problems:**

Perform poorly if it estimates incorrect bandwidth because of unpredictability in the behaviour of the bandwidth estimation scheme used in TCP Westwood. Changes in the inter-arrival times of the acknowledgements cause improvement or worsening of the throughput in rather unpredictable ways. Additionally, the sensitivity of TCP Westwood AckdInterval is variable.

## 4.6 TCP WestwoodNR

In TCP WestwoodNR [13] the sender continuously computes the connection Bandwidth Estimate (BWE) which is defined as the share of bottleneck bandwidth used by the connection. Thus, BWE is equal to the rate at which data is delivered to the TCP receiver. The estimate is based on the rate at which ACKs are received and on their payload. After a packet loss indication, (i.e, reception of 3 duplicate ACKs, or timeout expiration). The sender resets the congestion window and the slow start threshold based on BWE. More precisely, CWND=BWE x RTT. To understand the rationale of TCP-WNR, note that BWE varies from flow to flow sharing the same bottleneck; it corresponds to the rate actually achieved by each individual flow. Thus, it is a FEASIBLE (i.e. achievable) rate by definition. Consequently, the collection of all the BWE rates, as estimated by the connections sharing the same bottleneck, is a FEASIBLE set. When the bottleneck becomes saturated and packets are dropped, TCP-WNR selects a set of congestion windows that correspond exactly to the measured BWE rates and thus reproduce the current individual throughputs. The solution is feasible, but it is not guaranteed per se to be “fair share.” An additional property of this scheme, described in Section III, drives the rates to the same equilibrium point and makes it “fair share” under uniform propagation delays and single bottleneck assumptions. Another important element of this procedure is the RTT estimation. RTT is required to compute the window that supports the estimated rate BWE. Ideally, the RTT should be measured when the bottleneck is empty. In practice, it is set equal to the overall minimum round trip delay (RTTmin) measured so far on that connection (based on continuous monitoring of ACK RTTs). A packet loss is indicated by (a) the reception of3 DUPACKs, or (b) a coarse timeout expiration. In case the loss indication is 3 DUPACKs, TCP-WestwoodNR sets CWND and SSThreshhold as follows:

**Psudocode for WestwoodNR:**

```
If (3 DUPACKs are received)
SSThreshhold = (BWE * RTTmin) / seg_size;
if (CWND > SSThreshhold) /* congestion avoid. */
CWND = SSThreshhold;
endif
```

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*endif*
In case a packet loss is indicated by timeout expiration, CWND and SSThreshhold are set as follows:

```
if (coarse timeout expires)
CWND = 1;
SSThreshhold = (BWE * RTTmin) / seg_size;
if (SSThreshhold < 2)
SSThreshhold = 2;
endif;
endif
```

**Problems:**
This variant cann't distinguishes between buffer overflow and random losses, Performs poorly if it estimates incorrect Bandwidth. The performance of TCP WestwoodNR is strongly impacted when the number of lost ACKs increases, with the increase of BER value.

### 4.7 TCP Vegas

Bandwidth Estimation scheme used by TCP Vegas is more efficient than other TCP variants. This scheme makes bandwidth estimation by using the difference between the expected flow rates and the actual flow rates. TCP Vegas was introduced in 1994 as an alternative to TCP Reno and its implementation and tests showed that it achieves better throughput than TCP Reno. TCP Vegas' bandwidth estimation differs from that in TCP Reno. Unlike TCP Reno, which uses packet loss as the indication of network congestion, TCP Vegas uses the difference between the estimated throughput and the measured throughput as the measure of congestion. TCP Vegas records the smallest measured round trip time as BaseRTT and computes the available bandwidth as:

$$ExpectedBandwidth = WindowSize / BaseRTT$$

Where as the Window Size is the current window size. During the packet transmission the round trip time (RTT) of packets are recorded. The actual throughput is calculated as:

$$Actual\ Bandwidth = WindowSize / RTT$$

The difference between the Expected Bandwidth and Actual Bandwidth is used to adjust the Window Size.

$$Diff = Expected\ Bandwidth - Actual\ Bandwidth$$

Two values α and β ( $0 \le \alpha < \beta$ ) are defined as the thresholds. If $Diff < \alpha$ , the window size is increased during the next RTT; If $Diff > \beta$ , then the window size is decreased during the next RTT. Otherwise, the window size is unchanged. The goal of TCP Vegas is to keep a certain number of packets or bytes in the queues of the network [1]. If the actual throughput is smaller than the expected throughput, TCP Vegas takes this as indication of network congestion, and if the actual throughput is very close to the expected throughput, it is suggested that the available bandwidth is not fully utilized, so TCP Vegas increases the window size.

This mechanism used in TCP Vegas to estimate the available bandwidth does not purposely cause any packet loss. Hence the oscillatory behavior is removed and a better throughput is achieved. Retransmission mechanism used by TCP-Vegas is more efficient as compared to TCP-Reno as it retransmits the corresponding packet as soon as it receives a single duplicate ACK and does not wait for three ACKs. TCP-Vegas as compared to TCP-Reno is more accurate and is less aggressive, thus it does not reduce its CWND unnecessarily.

**Problems:**
It has problems when packets do not follow the same route and when large delays are present. When routes change for a certain TCP Vegas flow, the BaseRTT recorded from the previous route is no longer accurate; this affects the accuracy of ActualBandwidth and subsequently influences the performance of TCP Vegas. TCP Vegas could become unstable when there is large network delay for a flow; later established connections cannot get a fair share of the bandwidth, and when they coexist with TCP Reno connections, TCP Reno connections use most of the bandwidth.

### 4.8 TCP Fack

The development in TCP SACK with Forward Acknowledgement is identified as TCP FACK [1]. The utilization of TCP FACK is almost identical to SACK but it establishes a little enhancement evaluated to it. It uses SACK option to better estimate the amount of data in transit. TCP FACK introduces a better way to halve the window when congestion is detected. When CWND is immediately halved, the sender stops transmitting for a while and then resumes when enough data has left the network. In this one RTT can be avoided when the window is gradually decreased .When congestion occurs; the window should be halved according to the multiplicative decrease of the correct CWND. Since the sender identifies congestion at least one RTT after it happened, if during that RTT it was in slow start mode, then the current CWND will be almost double than CWND when congestion occurred. Therefore, in this case, CWND is first halved to estimate the correct CWND that should BWE further decreased.

## 5 Tools and Technique

### 5.1 NS2 (Network simulator version 2)

NS2 is a discrete event simulator targeted at networking research. Provides substantial support for simulation of TCP, routing, and multicast protocols over wired and wireless (local and satellite) networks. It is primarily UNIX based.

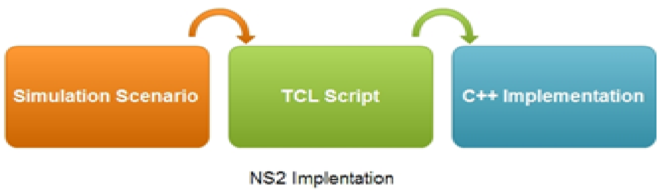

**Fig. 4 NS2 Implementation**

In this paper validation of basic network model and simulation of different network scenario's done using ns2, because it is scientifically verified and recommended by research community worldwide.

### 5.2 Gnu plot

Gnu plot is a command-driven interactive function plotting program. It can be used to plot functions and data points in both two- and three-dimensional plots in many different formats. It is designed primarily for the visual display of scientific data.
It is software for making 2D and 3D graphs from data and functions. Gnu plot supports lots of output formats, including drivers for many printers, (La) TeX, (x) fig, Postscript, and much more. It is frequently used for publication-quality graphics as well as education. The program runs on all major computers and operating systems such as GNU/Linux, UNIX, Microsoft Windows, Mac OS, and others. Gnuplot can produce output directly on screen, or in many formats of graphics files, including Portable Network Graphics (PNG), Encapsulated PostScript (EPS), Scalable Vector Graphics (SVG), JPEG and many others. The program can be used both interactively and in batch mode using scripts. The program is well supported and documented.
Gnuplot is used as the plotting engine of GNU Octave, Maxima and gretl, and it can be used from various scripting languages, including Perl, Python, Java, Ruby, Ch, and Smalltalk.

### 5.3 Edraw Max

Edraw Max, vector-based diagramming software, allows users to create flowcharts, organizational charts, workflow diagrams, software diagrams, network diagrams and more. Edraw's ease of use makes it particularly suitable for users who need to create professional quality drawings quickly and simply, without having to invest time in learning to use a complex application.
It enables students, teachers and business professionals to reliably create and publish kinds of diagrams to represent any ideas so we have used Edraw Max for drowing the various diagrams.

### 5.4 AWK scripting language

The AWK utility is an interpreted programming language typically used as a data extraction and reporting tool. It is a standard feature of most Unix-like operating systems.
AWK is a language for processing text files. A file is treated as a sequence of records, and by default each line is a record. Each line is broken up into a sequence of fields, so we can think of the first word in a line as the first field, the second word as the second field, and so on. An AWK program is of a sequence of pattern-action statements. AWK reads the input a line at a time. A line is scanned for each pattern in the program, and for each pattern that matches, the associated action is executed. In our simulation we have used AWK programs to process and obtain result fromtrace and NAM file.

## 6 SIMULATION SCENARIO AND OBTAINED RESULTS

### 6.1 Basic Validation Model

#### 6.1.1 Network Congestion Scenario

In this scenario, we create a congested node at the middle of the five-node topology by generating three TCP data traffic flows that must pass by this intermediate node to reach the other communicating end. One should also note that, different levels of data

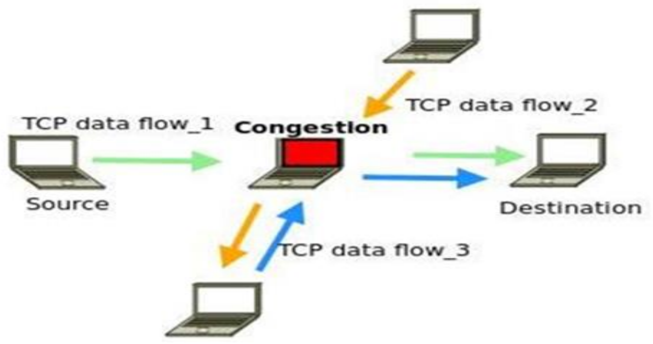

**Fig. 5 Congestion Network Scenario**

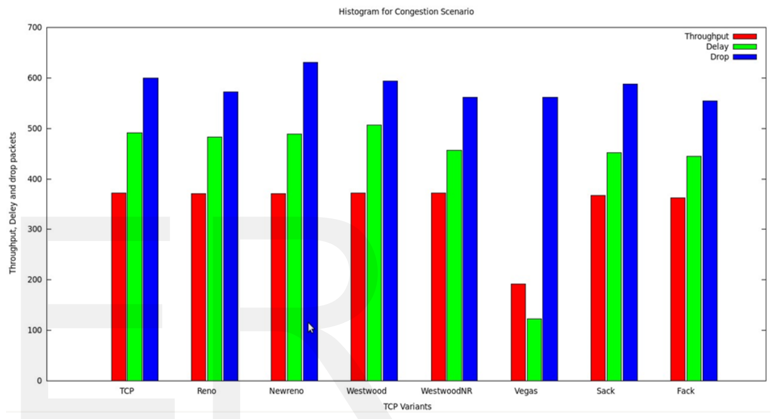

**Fig. 5 Histogram of throughput, delay and packet drop of Different tcp variants in congestion network scenario**

TABLE 1 Values of Congestion Scenario

| Variant | Throughput | Delay | Drop Packet |
|---|---|---|---|
| Tahoe | 371.54 | 491.632 | 600 |
| Reno | 371.25 | 482.392 | 572 |
| NewReno | 370.96 | 489.4 | 631 |
| Westwood | 371.70 | 506.764 | 594 |
| WestwoodNR | 371.70 | 456.783 | 562 |
| Vegas | 192.35 | 122.535 | 562 |
| Sack | 367.49 | 451.994 | 588 |
| Fack | 362.81 | 445.309 | 555 |

Congestion can be generated by controlling the number of TCP data flows crossing this particular network node at a certain time. The congested network is tested using Tcl script with different TCP variants as traffic agent and ploted histogram which is showing throughput, delay and drop packets for each variant.

### 6.1.2 Link failure Scenario

In this model we force TCP to change its communication path way by moving one intermediate node which acts as router out of signal reception range between the communicating end points. In addition, it implies routes with different number of hops. Thus, each time TCP changes the communication route, the characteristics of the path between the communicating nodes changes. It is obvious that the choice and the establishment delay of the new route will be dependent on the implemented ad hoc routing protocol. Packet losses and delay changes will also be implied by the link loss and the new chosen route. We notice that the effect of such networks nodes' mobility can be represented by the link failure scenario described above as it is the most direct consequence of mobility.

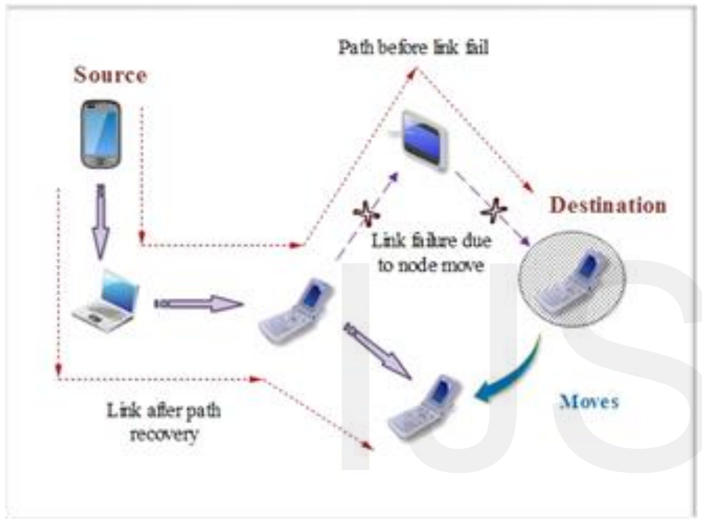

**Fig. 5 Link failure Scenario in Ad-Hoc Network**

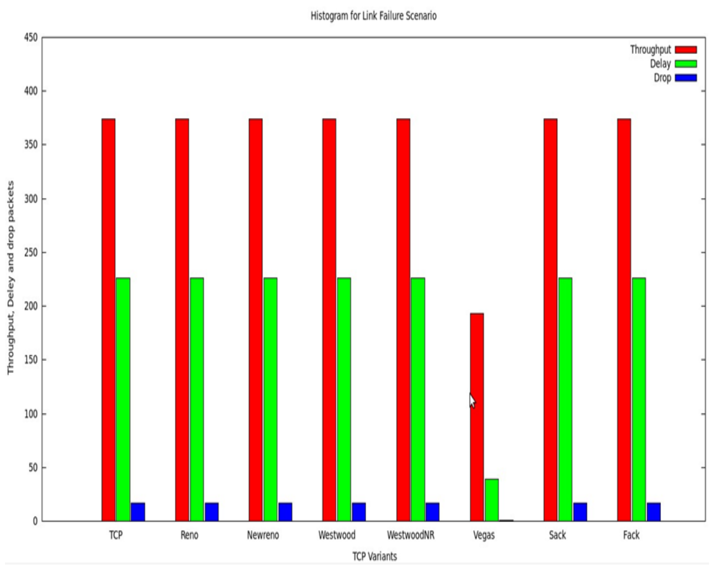

**Fig. 6 Histogram of throughput, delay and packet drop of Different tcp variants in Link failure Network Scenario**

TABLE 2 Values of Link Failure scenario

| Variant | Throughput | Delay | Drop Packet |
|---|---|---|---|
| TCP | 374.09 | 226.1 | 17 |
| Reno | 374.09 | 226.1 | 17 |
| New Reno | 374.09 | 226.1 | 17 |
| Westwood | 374.09 | 226.1 | 17 |
| WestwoodNR | 374.09 | 226.1 | 17 |
| Vegas | 193.36 | 39.093 | 1 |
| Sack | 374.09 | 226.1 | 17 |
| Fack | 374.09 | 226.1 | 17 |

### 6.1.3 Signal Loss Scenario in Network

This scenario illustrates the situation where the wireless signal is not stable. The communicating nodes loose the connection due to signal loss and resume the communication when the signal comes back. Signal losses are generated by moving one of the intermediate nodes out of the radio range of its connection neighbors. This scenario created using three nodes end node acts as sender and receiver and intermediate node as router. Transmission of ftp traffic source flow through an intermediate node, Intermediate node moves away for few second and so signal loss occurs between source and destination, after few second intermediate node moves at original place and again retransmission starts.

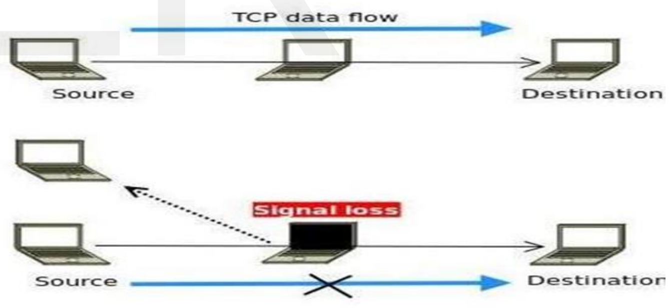

**Fig. 7 Signal loss Scenario in Ad-Hoc Network**

TABLE 3 Value of signal loss scenario

| Variant | Throughput | Delay | Drop Packet |
|---|---|---|---|
| Tahoe | 126.00 | 242.009 | 42 |
| Reno | 127.52 | 229.244 | 42 |
| NewReno | 48.99 | 242.241 | 27 |
| Westwood | 124.90 | 243.186 | 45 |
| WestwoodNR | 125.61 | 232.455 | 41 |
| Vegas | 93.94 | 55.8684 | 20 |
| Sack | 126.20 | 229.576 | 43 |
| Fack | 146.16 | 217.699 | 40 |

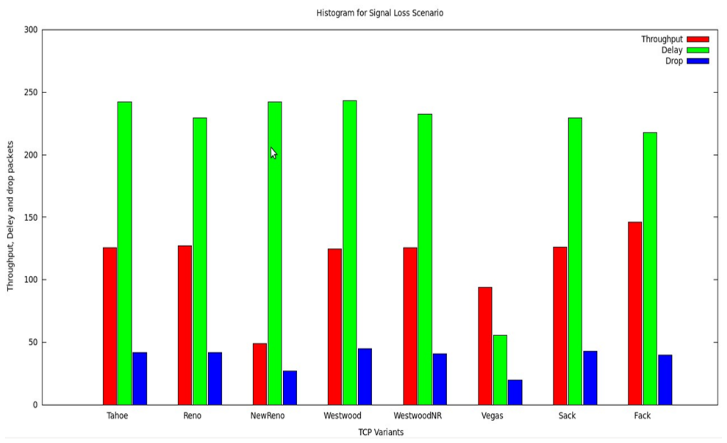

**Fig. 6 Histogram of throughput, delay and packet drop of Different tcp variants in Signal Loss Network Scenario**

### 6.1.4 Interference Scenario in Ad-hoc Network

In this scenario, two TCP connections are established in parallel.The main TCP connection is disturbed by the interferences generated by the second TCP connection. Indeed, the node acting as forwarder for the main TCP connection is placed within the interference range of the second TCP connection sender. So, this situation creates interference and thus data packet losses. Interference scenario in wireless environment created using two traffic sources; Transmission of second traffic source will interfere to the first traffic source.

TABLE 4 Values Interference Scenario

| Variant | Throughput | Delay | Drop Packet |
|---|---|---|---|
| Tahoe | 736.51 | 236.228 | 1861 |
| Reno | 736.51 | 236.228 | 1861 |
| New Reno | 736.51 | 236.228 | 1861 |
| Westwood | 736.51 | 236.228 | 1861 |
| WestwoodNR | 736.51 | 236.228 | 1861 |
| Vegas | 380.24 | 38.713 | 1905 |
| Sack | 736.51 | 236.228 | 1861 |
| Fack | 736.51 | 236.228 | 1861 |

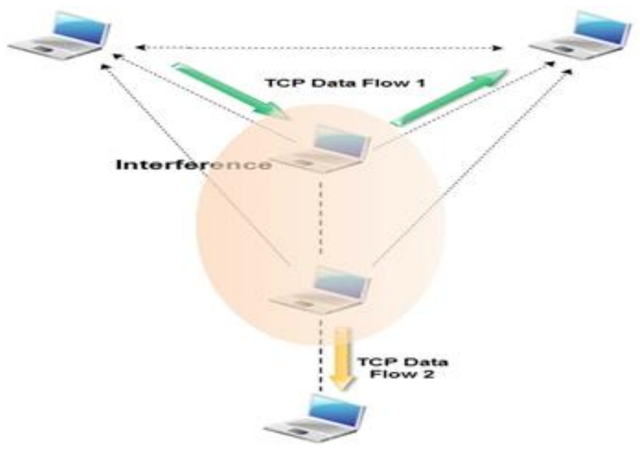

**Fig. 7 Intetrference Scenario in Ad-Hoc Network**

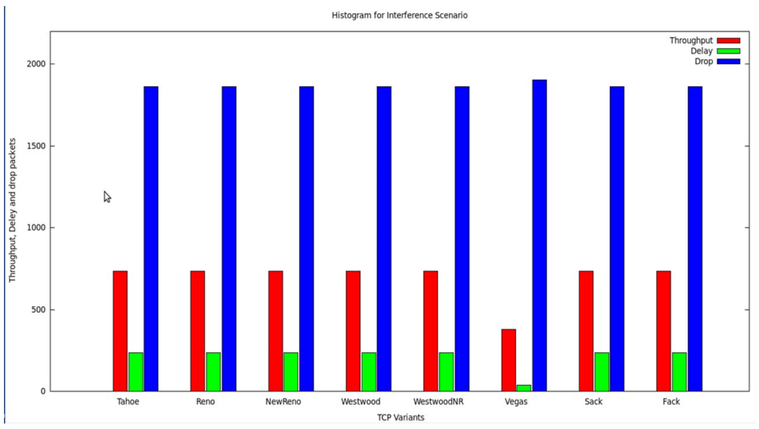

**Fig. 8 Histogram of throughput, delay and packet drop of Different tcp variants in Interference Network Scenario**

TABLE 5 Simulation Parameters Values

| Parameter | Values |
|---|---|
| Channel Type | Wireless channel |
| Radio Propagation Model | Two Ray ground |
| Queue type | Droptail/PriQue |
| Max. packet(buffer size) | 50 |
| Network interface | Wirelessphy |
| MAC Protocol | 802.11 |
| Data Rate | 1 Mbps |
| Transmission Radius | 250 |
| Interference Radius | 550 |
| Packet size | 1000 bytes |
| Routing protocol | AODV, DSDV |
| Simulation Time | 150 s |
| Value x | 700 |
| Value y | 500 |
| Agent trace | ON |
| Mac trace | OFF |
| Router trace | ON |
| Movement trace | ON |

## 6.1 Main Simulation Scenarios

### 6.2.1 Chain multi-hop Network

The chain network created with five hopes and six nodes to conduct simulation experiments.

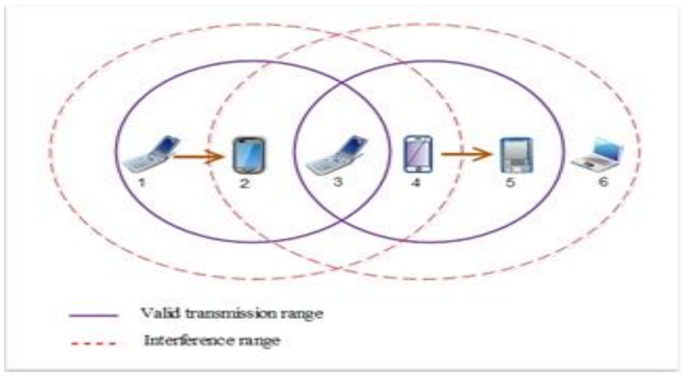

**Fig. 9 Chain topology in multihop Ad-Hoc Network**

The network consists of variable length chain of static nodes, placed at a distance of 200m from one another. FTP traffic is transferred between the first and last node of the chain shown in (Fig. 9). During the simulation different TCP variant attached as agent at the transport layer for the same scenario. One FTP connection kept active at a time and sequential TCP connection are initiated and terminated.

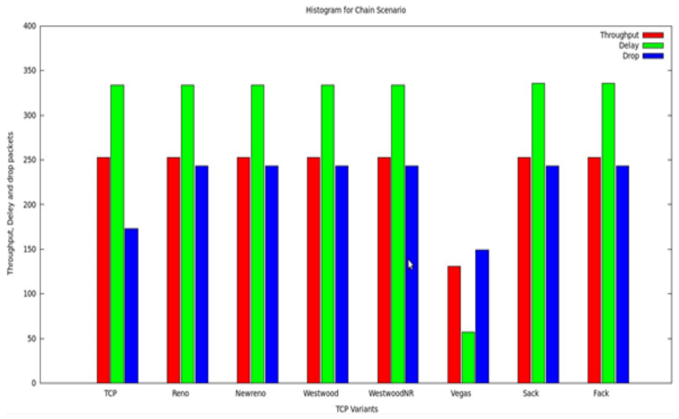

**Fig. 10 Histogram of throughput, delay and packet drop of Different tcp variants in Chain Topology**

TABLE 6 Values for Chain Topology

| Variant | Throughput | Delay | Drop Packet |
|---|---|---|---|
| Tahoe | 253.03 | 333.966 | 173 |
| Reno | 253.03 | 333.966 | 243 |
| New Reno | 253.03 | 333.966 | 243 |
| Westwood | 253.03 | 333.966 | 243 |
| WestwoodNR | 253.03 | 333.966 | 243 |
| Vegas | 130.86 | 57.637 | 149 |
| Sack | 253.03 | 333.966 | 243 |
| Fack | 253.03 | 333.966 | 243 |

## 6.2.2 Grid multi-hop Ad-Hoc Network

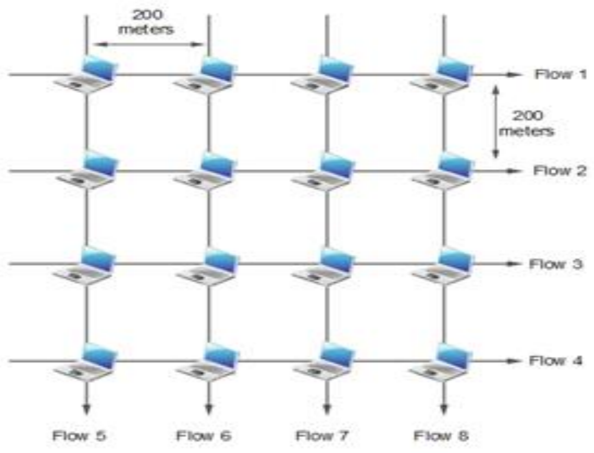

**Fig. 11 Grid topology in multihop Ad-Hoc Network**

Fig. 11 shows a static grid Ad-hoc network as experiment topologycal network with *16* nodes. In a stationary environment without mobility, the distance between two adjacent nodes is set to be 200 m, and the transmission and interference radii are set to 250 and 550 m, respectively. In each row, a TCP connection is assumed to set up from the left end node to the right end, and similarly, in each column, a TCP connection is assumed to set up from the top end node to the bottom end node.

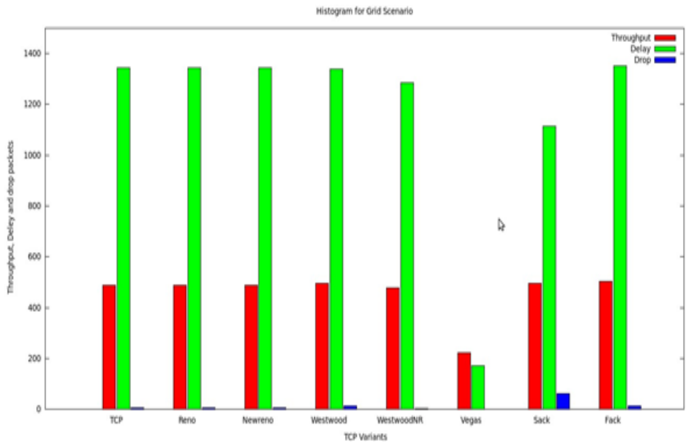

**Fig. 12 Histogram of throughput, delay and packet drop of Different tcp variants in Chain Topology**

TABLE 7 Values for Grid Topology

| Variant | Throughput | Delay | Drop Packet |
|---|---|---|---|
| Tahoe | 488.35 | 1345.12 | 5 |
| Reno | 488.35 | 1345.12 | 5 |
| NewReno | 488.35 | 1345.12 | 5 |
| Westwood | 495.59 | 1339.02 | 12 |
| WestwoodNR | 478.16 | 1285.47 | 4 |
| Vegas | 222.78 | 171.355 | 0 |
| Sack | 497.10 | 1113.29 | 63 |
| Fack | 504.30 | 1350.66 | 14 |

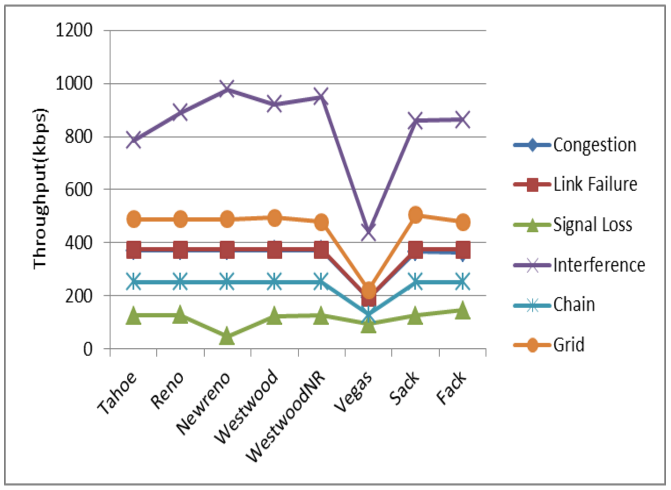

**Fig. 13 Throughput plot fop TCP variants in all Network scenario**

## 7 Conclusion

Analysis of various TCP variants has been carried out based on the problems and proposed enhancements to find out pros and cons of each variant. It reveals that there is no single variant which can handle all problems such as congestion, link failure, signal loss, interference. Also it neither gives optimized throughput nor performs efficiently over chain and grid network scenarios. Most of these variants are best in particular scenarios whereas the performance is lagging in some other MANET scenarios. Referring to the ns2 based simulated results, it is observed that proactive and bandwidth estimation based variants shows better handling to the mobility and frequent link failures whereas many of the TCP-variants show poor handling to changing network conditions. Thus it can be concluded that; the area is still open for research and the TCP-variants which are proactive and based on bandwidth estimation are more effective. Hence these parameters need to be taken into consideration for further investigation to obtain unique, reliable, robust, efficient TCP-variant and need to be further explored to make it suitable for adapting over mobile ad hoc networks.

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