International Journal of
Emerging Research in Management &Technology
ISSN: 2278-9359 (Volume-4, Issue-2)

Research Article **February 2015**

A Survey on Versions of TCP over WiMAX

Gurvinder Singh Dhaliwal
Research Scholar, CCC, College of Engg., Landran Road, Mohali, India

Gagandeep Singh
Asst. Prof., CEC, College of Engg., Landran Road, Mohali, India

Sumit Kumar
Asst. Prof., CEC, College of Engg., Landran Road, Mohali, India

Abstract—

Worldwide Interoperability for Microwave Access (WiMAX) is essentially a next-generation wireless technology that enhances broadband wireless access. (WiMAX) is a standards-based wireless technology for providing high-speed, last-mile broadband connectivity to homes and businesses and for mobile wireless networks. WiMAX is similar to Wi-Fi but offers larger bandwidth, stronger encryption, and improved performance over longer distances by connecting between receiving stations that are not in the line of sight. WiMAX uses Orthogonal Frequency Division Modulation (OFDM) technology, which has a lower power consumption rate. WiMAX can be used for a number of applications, including last-mile broadband connections, hotspots and cellular backhaul, and high-speed enterprise connectivity for business. It supports broadband services such as VoIP or video. WiMAX is also a possibility for backhaul technology in municipal Wi-Fi networks. WiMAX or 802.16 is definitely a hot topic and has a fair list of industry supporters. The paper focuses on the different congestion control mechanisms implemented by the Transmission Control Protocol (TCP). Reliable transport protocols such as TCP are tuned to perform well in traditional networks where packet losses occur mostly because of congestion. However, networks with wireless and other lossy links also suffer from significant losses due to bit errors and handoffs. TCP responds to all losses by invoking congestion control and avoidance algorithms, resulting in degraded end-to-end performance in wireless and lossy systems. In this Paper, we compare several schemes designed to improve the performance of TCP in such networks.

Keywords: WiMAX, IEEE 802.16, TCP, C-TCP, TCP-CUBIC.

I. INTRODUCTION

The term *wireless broadband* generally refers to high-speed (minimally, several hundred kilobits per second) data transmissions occurring within an infrastructure of more or less fixed points, including both stationary subscriber terminals and service provider base stations. This is distinct from mobile data transmissions where the subscriber can expect to access the network while in transit and where only the network operator's base stations occupy fixed locations. Broadband wireless, as it is today, is properly a competitor to optical fiber, hybrid fiber coax (the physical infrastructure of most cable television plants), DSL, and, to a much lesser extent, broadband satellite. The demand for broadband connectivity from urban homes is growingrapidly, but this cannot be met effectively by existing wired technologies.WiMAX has the potential to providewidespread Internet access that can usher in economic growth, better educationand health care, and improved entertainment services. WiMAX can be describedas a framework for the evolution of wireless "broadband" rather than a static implementationof wireless technologies. Due to the trend toward mobile applications,WiMAX has a promising future. Low networkinvestment costs and non-line-of-sight operation over licensed or non-licensedradio spectrum make WiMAX an attractive technology. The first chapter provides a overview of different wireless networks followed by a complete introduction to the WiMAX and its specifications.

1. WIRELESS NETWORKS

Since the final decades of the twentieth century, data networks have known steadily growing success. After the installation of fixed Internet networks in many places all over the planet and their now large expansion, the need is now becoming more important for wireless access. There is no doubt that by the end of the first decade of the twentieth century, high-speed wireless data access will be largely deployed worldwide. A large number of wireless transmission technologies exist, other systems still being under design. These technologies can be distributed over different network families, based on a network scale. In figure 1, a now-classical representation is shown of wireless network categories, with the most famous technologies for each type of network.

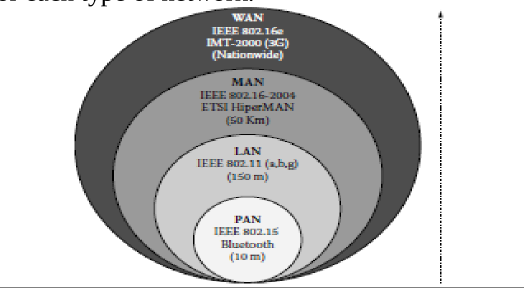

Figure1 :Illustration of network types

- **PAN :** A *Personal Area Network* (PAN) is a (generally wireless) data network used for communication among data devices close to one person. The scope of a PAN is then of the order of a few metres, generally assumed to be less than 10m, although some WPAN technologies may have a greater reach. Examples of WPAN technologies are Bluetooth, UWB and Zigbee.
- **LAN :** A *Local Area Network* (LAN) is a data network used for communication among data devices: computer, telephones, printer and personal digital assistants (PDAs). This network covers a relatively small area, like a home, an office or a small campus (or part of a campus). The scope of a LAN is of the order of 100 metres. The most presently used LANs are Ethernet (fixed LAN) and WiFi (Wireless LAN, or WLAN).

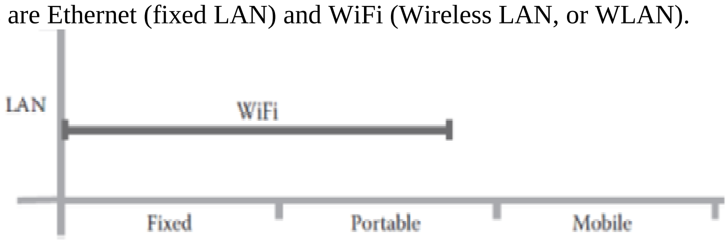

Figure 2 :Local Area Network (mapped against usage models and access modes).

- **MAN :** A *Metropolitan Area Network* (MAN) is a data network that may cover up to several kilometres, typically a large campus or a city. For instance, a university may have a MAN that joins together many of its LANs situated around the site, each LAN being of the order of half a square kilometre. Then from this MAN the university could have several links to other MANs that make up a WAN. Examples of MAN technologies are FDDI (Fibre-Distributed Data Interface), DQDB (Distributed Queue Dual Bus) and Ethernet-based MAN. Fixed WiMAX can be considered as a Wireless MAN (WMAN).

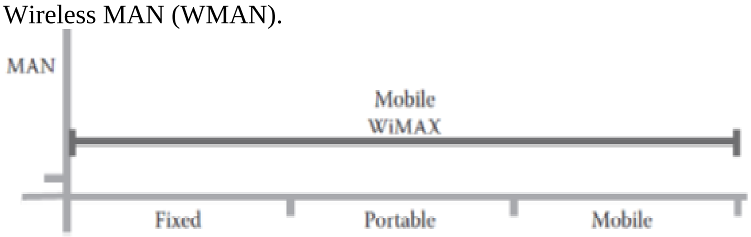

Figure 3 :Metropolitan Area Network (mapped against usage models and access modes).

- **WAN :** A *Wide Area Network* (WAN) is a data network covering a wide geographical area, as big as the Planet. WANs are based on the connection of LANs, allowing users in one location to communicate with users in other locations. Typically, a WAN consists of a number of interconnected switching nodes. These connections are made using leased lines and circuit-switched and packet-switched methods. The most (by far) presently used WAN is the Internet network. Other examples are 3G and mobile WiMAX networks, which are Wireless WANs. The WANs often have much smaller data rates than LANs (consider, for example, the Internet and Ethernet).

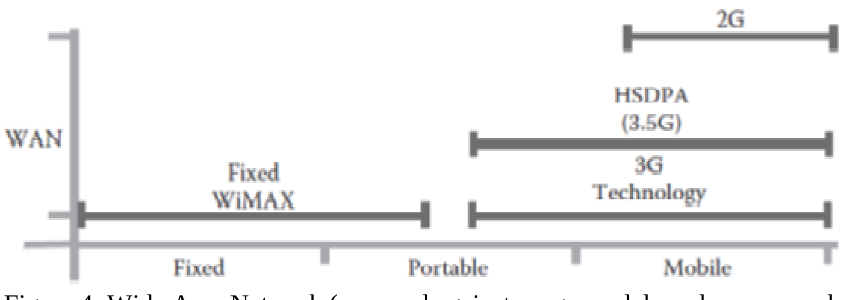

Figure 4 :Wide Area Network (mapped against usage models and access modes).

Obviously, the traditional mechanisms such as wifi, Bluetooth etc., to access network are no longer suitable and cannot meet the newly upcoming requirement. Fortunately, the Broadband Wireless Access (BWA) technology such as WiMAX, which can meet people's need to access the Internet conveniently, becomes more popular in recent years.

2. WIMAX-INTRODUCTION

The IEEE 802.16 group has started to produce recommendations for a relatively long period. The evolution of the wireless physical layers is seen in the different versions, the same way it can be noticed in IEEE 802.11 standard. That is why we can see a first physical layer implementing plesiochronous digital hierarchy (PDH) like data rates with a line of sight restrictive condition. Few years later, with the familiarization to OFDM, a new version has come up with "line of sight" restriction removed but with lower throughput.We did not see any IEEE 802.16 equipment in the first editions of the standard, not because the lack of products, but because of the unclear legislation in that area together with the wide deployment of fixed asymmetric digital subscriber line (ADSL) wired lines.

WiMAX will boost today's fragmented broadband wireless access market and mobileWiMAX promises to offer a solution to closing the existing digital divide. WiMAX can address the fixed wireless access and portable Internetmarket, complementing other broadband wireless technologies. Government initiatives to reduce the digital divide are making gains for broadband wireless countries such as Australia, South Korea, Taiwan, and the United States have programs in place today, and there has been a push by the European Commission for more flexible spectrum policies. WiMAX access can be easily integrated within both fixed and mobile architectures, enabling operators to integrate it within a single converged core network, thereby providing new capabilities for a user-centric broadband world. WiMAX addresses the following needs which may answer the question of closing the digital divide :

- Cost effective
- Offers high data rates
- Supports fixed, nomadic, and mobile applications thereby converging the fixed and mobile networks
- Easy to deploy and has flexible network architectures
- Supports interoperability with other networks
- Aimed at being the first truly a global wireless broadband network

WiMAX is a standard that is championed by the WiMAX forum which was formed in June 2001 to promote conformance to IEEE 802.16 standard. The WiMAX forum currently has more than 470 members comprising the majority of operators, component, and equipment companies in the communications ecosystem. TheWiMAX forum promotes interoperability by working closely with IEEE and other standards groups such as the European Telecommunications Standards Institute (ETSI) which have their own versions of broadband wireless. Along these lines, the WiMAX forum works closely with service providers and regulators to ensure that WiMAX forum certified systems meet customer and government requirements. The original WiMAX standard only catered for fixed and nomadic services. It was reviewed to address full mobility applications; hence, the mobile WiMAX standard, defined under the IEEE 802.16e specification was created. Mobile WiMAX supports full mobility, nomadic, and fixed systems to compete against DSL to cover isolated areas such as rural hot spots, private campus networks, and remote neighbourhoods. Mobile WiMAX is more promising to be deployed as a cellular network that offers ubiquitous broadband services to mobile users to over large geographical areas. It can be deployed as a central office bypass to avoid using existing wired infrastructure for competitive local exchange carriers and wireless Internet service provider.

3. WIMAX NETWORK ARCHITECTURE

In Figure 5, the overall network architecture of a WiMAX network. The network can be logically partitioned into three components, user terminals, ASN, and CSN. User terminals capture the data origination points, could be using the fixed, mobile, or portable WiMAX technology. All the three variations can be supported using a common air interface. ASN spans the BS and the ASN-GW. BS receives the transmitted signal, processes it, and converts into an IP packet and sends to the GWon the outgoing IP transport link.GWreceives and upon processing determines the destination on the network side and sends the packet. BS and GW are connected to each other using an IP transport. Typical implementations would have BS located in the field/coverage area and the GW will be centrally located in the switch centers. Therefore, the IP link between BS andGW forms the transport backhaul network.

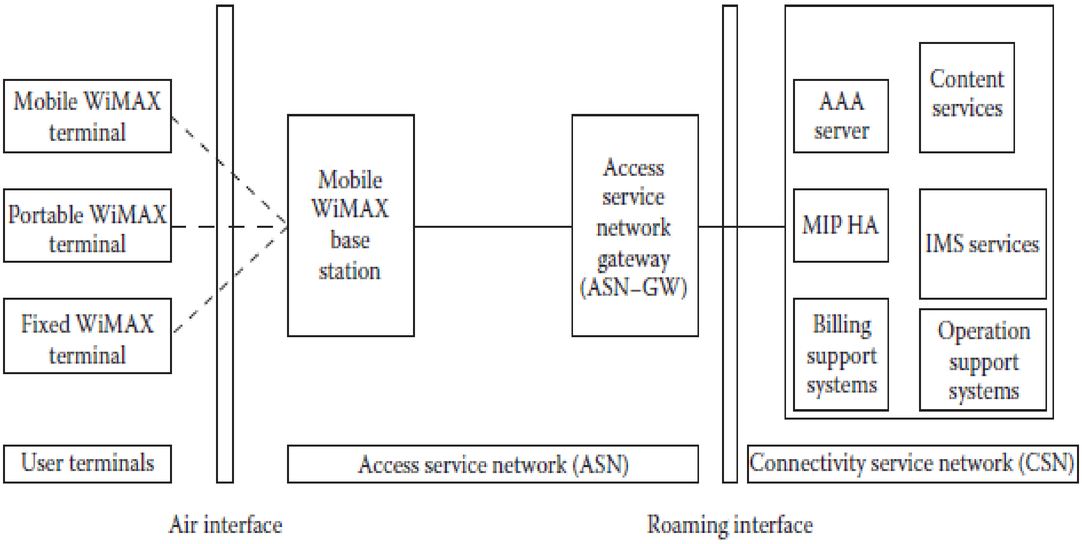

Figure 5 : Logical network architecture of a WiMAX network.

CSN contains many different commercial off-the shelf (COTS) components, which provide connectivity services to the WiMAX subscribers. Addressing, authentication, and availability (AAA) servers, mobile IP home agent (MIP HA), IP multimedia services (IMS), content services, etc. provide support for seamless services to subscribers. AAA servers ensure that a user is uniquely identified and authenticated as legitimate customer. MIP HA ensures that roaming across IP networks is handled and accurate routing of data packets is ensured. Call processing related services is provided by IMS entity. Billing and operational support systems help in managing the overall network.

In Figure 6, typical implementation of a WiMAX network in a market. For example, say a carrier plans to lay down WiMAX network in Washington D.C. market. Typically, we would have more than 100 BSs connecting to a GW location, based on the anticipated traffic, each GW location might require a cluster of servers providing the functions of the GW. Each IP transport link would be leased from the local carrier and provisioned. Based upon the cost points and required capacity, the carrier can choose to directly lease a TDM segment, Ethernet link, fiber connectivity, etc. Components of the CSN located at each switch center might also be implemented using clusters and would have enough capacity to support the entire market. Switch centers could be connected to each other using a high speed IP network running on an OC-192 (or higher) SONET ring leased from local exchange carrier. Actual network would also include connectivity to the other markets, trucking with public switched telephone network (PSTN) via the end office (EO), tandem connections with For most WiMAX networks, it is unlikely that the carriers would provision the IP transport based on the capacity of the WiMAX air interface. According to WiMAX forum, air interface built on 10 MHz channel with 2 × 2 MIMO can support peak downlink rate of 63 Mbps and peak uplink rate of 28 Mbps per sector. Assuming three sectors per BS, this would translate into close to 200 Mbps of backhaul transport for each BS. When we share the symbols 3:1 between DL and UL, it could provide data rates of 46 Mbps DL and 8 Mbps UL per sector. Even then it would require about 150 Mbps of capacity between BS and GW. Such a requirement would lead to an unmanageable backhaul cost, which might become a road block in the large-scale adoption of the WiMAX technology. Our contention is that the service providerswill only provision based on the anticipated demand. For example, they might provision just enough capacity for$N$ voice calls,M video calls, and few more Mbps for best effort. This would ensure that the initial cost of building the network is manageable, and as the users grow, more backhaul can be added to ensure acceptable QoS for the subscribers.

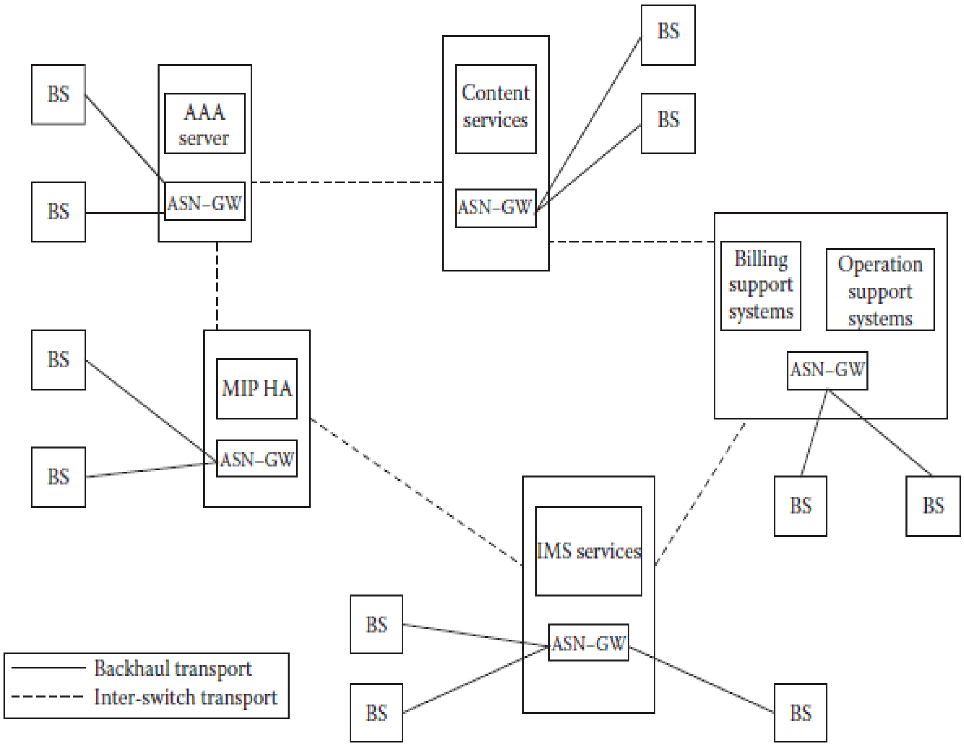

Figure 6: Physical network architecture of a WiMAX network.

4. TCP -CONGESTION CONTROL

When the load offered to any network is more than it can handle, congestion builds up. The Internet is no exception. 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 (i.e., is delivered). TCP attempts to achieve this goal by dynamically manipulating the window size. The first step in managing congestion is detecting it. In the old days, detection of congestion was difficult. A timeout caused by a lost packet could have been caused by either

(1) noise on a transmission line or
(2) packet discard at a congested router.

Telling the difference was difficult. Now a days, packet loss due to transmission errors are relatively rare because most long-haul trunks are fiber. Consequently, most transmission timeouts on the Internet are due to congestion. All the Internet TCP algorithms assume that timeouts are caused by congestion and monitor timeouts for signs of trouble the way miners watch their canaries. When a connection is established, a suitable window size has to be chosen. The receiver can specify a window based on its buffer size. If the sender sticks to this window size, problems will not occur due to buffer overflow at the receiving end, but they may still occur due to internal congestion within the network. The Internet solution is to realize that two potential problems exist—network capacity and receiver capacity—and to deal with each of them separately. To do so, each sender maintains two windows: the receiver window and a second window, the **congestion window**. Each reflects the number of bytes the sender may transmit. The number of bytes that may be sent is the minimum of the two windows. Thus, the effective window is the minimum of what the sender thinks is all right and what the receiver thinks is all right. As an illustration of how the congestion algorithm works, see Fig. 4.

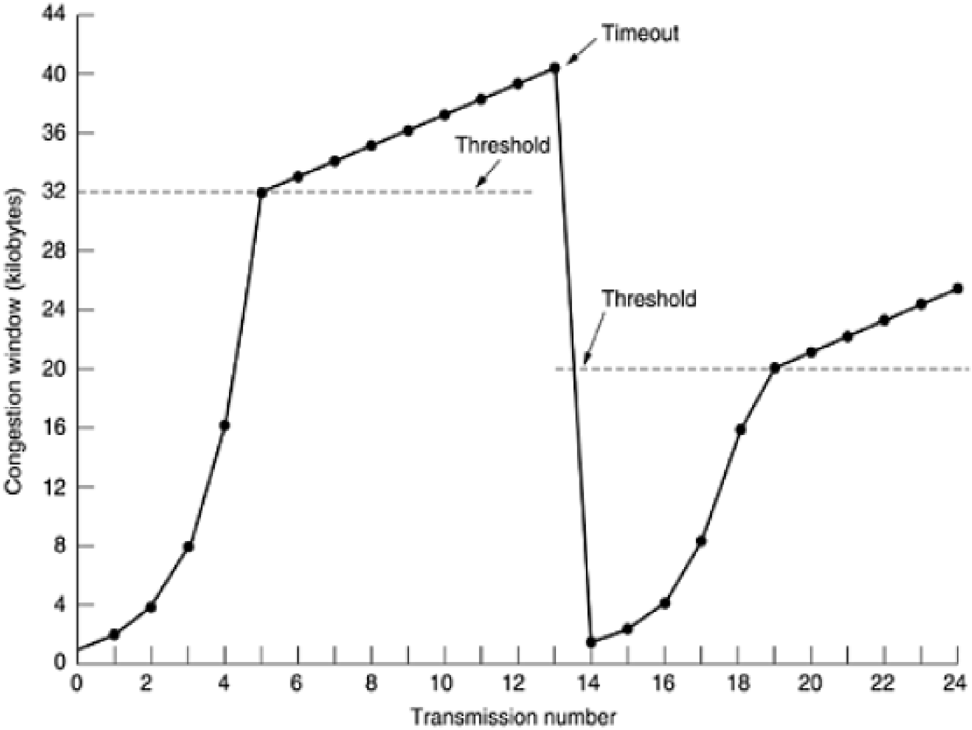

Figure 1.7: An example of the Internet congestion algorithm.

II. RELATED WORK

In [1] Vikram Mehta et.al focuses on analyzing essential QoS parameters for Wimax Network. Essential QoS parameters like delay, Jitter, Packet delivery Ratio (PLR), Packet Loss Ratio (PLR) and throughput have been calculated for 500 mobile nodes in a WiMax network. Ad Hoc on Demand Distance Vector Routing (AODV) protocol has been chosen as a routing protocol because of its ability to perform well under highly mobile and random conditions. MATLAB software version R2011 was used for creating WiMax network architecture and Regression analysis is done for each of the QoS parameter.The results help in critically analyzing QoS parameters for WIMAX Network and it has been found that an optimum value of QoS parameters is obtained with increasing number of mobile nodes for WiMax Network.

R. Wang, M. Valla, M.Y. Sanadid and Mario Gerla [2], the authors proposed an available Bandwidth Estimation TCPW-BR -a refinement of TCP Westwood that allowing the management of the Efficiency/Friendliness-to- NewReno tradeoff. TCP Westwood design adheres to the end-to-end transparency] and requires only sender side modification. The key innovation of TCPW is to use a bandwidth estimate directly to drive cwin and ssthresh. The current estimation method in TCPW is based on “bandwidth Estimation”, i.e., BE. This TCPW BE strategy provides significant throughput gains, especially the large leaky pipes. Under certain congestion circumstances, BE exceeds the fair share of a connection resulting in possible unfriendliness to TCP New Reno connections. The authors present a refinement of standard TCPW where two estimators are maintained, along with a method to identify the predominant cause of packet loss. Depending on the outcome, the appropriate estimator is used. Both estimators use information obtained from ACKs received at the sender. One estimator BE, as in the standard TCPW, considers each ACK pair separately to obtain a bandwidth sample, filters the samples into a low-pass filter and returns as a result the available bandwidth that the TCP connection is estimated to be getting from the network. The other estimator we propose in this paper, called Rate Estimator (RE), considers the amount of data acknowledged during the latest interval of time T as sample, then feeds such samples into an appropriate low-pass filter to get the estimated rate. The authors proposed a combined RE/BE estimation method—CRB—that allows to adapt to different types of losses (congestion or link errors), as well to manage the Efficiency/Friendliness tradeoff when TCPW and NewReno are simultaneously deployed in a network.

Tuan Anh Le, Choong Seon Hong, and Eui-Nam Huh [3], the authors we propose an extended version of regular TCP Westwood for multiple paths over wireless networks,called Multipath TCP Westwood (MPTCP) TCP Westwood (TCPW) uses the available bandwidth estimation technique to improve TCP performance in such environment. The performance od regular TCP is very poor in wireless networks, where packet loss often is caused by random error rather than by network congestion as in wired networks. TCP Westwood (TCPW) uses the available bandwidth estimation technique to improve TCP performance in such environment. MPTCPW congestion control is designed as a coordinated control between paths which allows load-balancing feature beween paths, fair sharing to regular TCPW at bottleneck. The

authors also conclude that MPTCPW can achieve stability, higher throughput compared with MPTCP, fairness to regular TCPW, and greater load-balancing than uncoordinated MPTCPW under various network conditions.
C. Chiang et al. [4], propose an Adaptive Split Ratio (ASR) scheme which adjusts the bandwidth ratio of DL to UL adaptively according to the current traffic profile, wireless interference, and transport layer parameters, so as to maximize the aggregate throughput of TCP based traffic. The authors study adaptive channel split ratio of uplink to downlink capacities in TDD-based IEEE 802.16 (WiMAX) wireless networks. ASR can also cooperate with the BS scheduler to throttle the TCP source when acknowledgements are transmitted infrequently, thus preventing either direction (i.e., downlink or uplink) from becoming the bottleneck. The simulation results show that our adaptive scheme outperforms static allocations in terms of higher aggregate throughput and better adaptivity to network dynamics.
L. Subedi et al. [5], compare the performance of various TCP algorithms in wireless and wired networks. Simulation results indicated that in wireless networks with signal attenuation, fading, and multipath, TCP Reno outperforms other congestion control algorithms in terms of congestion window size and file download response time. TCP Reno has larger congestion window size, shorter file download response time, and higher throughput and goodput than the remaining three TCP algorithms. In case of wired networks, TCP Reno with SACK has the best overall performance.
J. Huang et al. [6], presented a new method to improve TCP SACK'S performance over wireless links. The proposed method attempts to differentiate congestion and corruption loss by the use of tagged segments. In this method, a sender is able to discern congestion from corruption in large likelihood. In contrast to most other existing approaches employing explicit notification, proposed method is attractive in that it requires no modification at the receiver. Through simulations authors have compared the performance of proposed method with TCP SACK over both low and long delay noisy links. The results show that our method significantly improves TCP SACK over a wide range of channel conditions in both environments.
Kun Tan Jingmin *et al. proposed Compound TCP* (CTCP) [7], which incorporates a scalable delay-based component into the standard TCP congestion avoidance algorithm. This scalable delay-based component has a rapid window increase rule when the network is sensed to be under-utilized and gracefully reduces the sending rate once the bottleneck queue is built. With this delay-based component as an auto-tuning knob, *Compound TCP* can satisfy all three requirements pretty well:

1) CTCP can efficiently use the network resource and achieve high link utilization.
2) CTCP has similar or even improved RTT fairness compared to regular TCP. This is due to the delay-based component employed in the CTCP congestion avoidance algorithm. It is known that delay-based flow.
3) CTCP has good TCP-fairness. By employing the delay based component, CTCP can gracefully reduce the sending rate when the link is fully utilized. In this way, a CTCP flow will not cause more self-induced packet losses than a standard TCP flow, and therefore maintains fairness to other competing regular TCP flows[7].

III. VERSION OF TCP

TCP primary purpose is to provide a connection oriented reliable data transfer service between different applications to be able to provide these services on top of an unreliable communication system. TCP protocol has been extensively tuned to give good performance at the transport layer in the traditional wired network environment. However, TCP in its present form is not well suited for ad hoc networks where packet loss due to broken routes can result in the counterproductive invocation of TCP‘s congestion control mechanisms.

TCP Reno

TCP Reno employs the basic principle of Tahoe, such as slow starts and the congestion avoidance. 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 an algorithm called ‗Fast Re-Transmit‘. Whenever we receive 3 duplicate ACK‗s we take it as a sign that the segment was lost, so we re-transmit the segment without waiting for timeout. Thus we manage 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 we call Fast-Re-Transmit‘[2,5].

TCP New Reno

TCP New RENO is a slight modification over TCP-RENO. It is able to detect multiple packet losses and thus is much more efficient that RENO in the event of multiple packet losses. Like RENO, New-RENO also enters into fast-retransmit when it receives multiple duplicate packets, however it differs from RENO in that it does not exit fast-recovery until all the data which was out standing at the time it entered fast recovery is acknowledged.

TCP Sack

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 retransmits 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. SACK TCP requires that segments not be acknowledged cumulatively but should be acknowledged selectively. 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 [1,6].

TCP Fack

FACK or Forward Acknowledgement is a special algorithm that works on top of the SACK options, and is geared at congestion controlling. FACK algorithm uses information provided by SACK to add more precise control to the injection of data into the network during recovery – this is achieved by explicitly measuring the total number of bytes of data outstanding in the network. FACK decouples congestion control from data recovery thereby attaining more precise control over the data flow in the network.The main idea of FACK algorithm is to consider the most forward selective acknowledgement sequence number as a sign that all the previous acknowledged segments were lost. This observation allows improving recovery of losses significantly[1].

TCP Vegas

Vegas is a TCP implementation which is a modification of RENO. It builds on the fact that proactive measure to encounter congestion is much more efficient than reactive ones. It tried to get around the problem of coarse grain timeouts by suggesting an algorithm which checks for timeouts at a very efficient schedule. Also it overcomes the problem of requiring enough duplicate acknowledgements to detect a packet loss, and it also suggests a modified slow start algorithm which prevents it from congesting the network.

TCP-Westwood

TCP Westwood makes no attempt to correct the problem of non-congestion packet loss in wireless networks solely like Veno, but rather to improve the efficiency of TCP in all heterogeneous networks. It estimates the network's bandwidth by properly low-pass filtering and averaging the rate of returning acknowledgment packets per RTT. It then uses this bandwidth estimate to adjust the *ssthresh* and the *cwnd* to a value close to it when a packet loss is experienced *(adaptive decrease).* In particular, when three DUPACKs are received, both the *cwnd* and *ssthresh* are set equal to the *Estimated Bandwidth (BWE)* times the minimum measured RTT *(RTTmin)*; when a coarse timeout expires, the *ssthresh* is set as before, while the *cwnd* is set equal to one. The improvement of Westwood is a more realistic bandwidth estimation in comparison to TCP Vegas, which significantly increases TCP throughput over wireless links. TCP Westwood has also been tested in against handovers in simulated [1,3].

TCP-Compound

Compound TCP (CTCP) [7], which incorporates a scalable delay-based component into the standard TCP congestion avoidance algorithm. This scalable delay-based component has a rapid window increase rule when the network is sensed to be under-utilized and gracefully reduces the sending rate once the bottleneck queue is built. With this delay-based component as an auto-tuning knob, *Compound TCP* can satisfy all three requirements pretty well:

1) CTCP can efficiently use the network resource and achieve high link utilization.

2) CTCP has similar or even improved RTT fairness compared to regular TCP. This is due to the delay-based component employed in the CTCP congestion avoidance algorithm. It is known that delay-based flow.

3) CTCP has good TCP-fairness. By employing the delay based component, CTCP can gracefully reduce the sending rate when the link is fully utilized. In this way, a CTCP flow will not cause more self-induced packet losses than a standard TCP flow, and therefore maintains fairness to other competing regular TCP flows[7].

TCP-Cubic

CUBIC is an enhanced version of BIC: it simplifies the BIC window control and improves its TCP-friendliness and RTT-fairness. The window growth function of CUBIC is governed by a cubic function in terms of the elapsed time since the last loss event. TCP-cubic function provides a good stability and scalability. Furthermore, the real-time nature of this transport protocol keeps the window growth rate independent of RTT, which keeps the protocol TCP friendly under both short and long RTT paths..[1,2]

IV. CONCLUSION

In this article we reviewed the challenges and basic concepts behind out-of-order packet is really delivered due to multi-path route between TCP sender and receiver, which has significant performance degradation effect and provided a thorough overview of TCP Variants in routing metrics and design considerations. TCP/IP protocol was designed for wired networks which provides end to end reliable communication between nodes and assures ordered delivery of packets. It also provides flow control and error control mechanisms. As it is still a successful protocol in wired networks, losses are mainly due to congestion. But in case of ad hoc networks packet losses are due to congestion in the network and due to frequent link failures so when we adapt TCP to ad hoc networks it misinterprets the packet losses due to link failure as packet losses due to congestion and in the instance of a timeout, backing-off its retransmission timeout (RTO). This results in unnecessary reduction of transmission rate because of which throughput of the whole network degrades. Different TCP versions react with different types of behavior. Through literature, we have noted that TCP throughput decreases significantly when node movement cases link failures, due to TCP's inability to recognize the difference between link failure and congestion.

REFERENCES

[1] Vikram Mehta ,"*Performance Analysis of QoS Parameters for Wimax Networks",* International Journal of Engineering and Innovative Technology, vol.1, issue-5, May 2012.

[2] R. Wang, M. Valla, M.Y. Sanadid and Mario Gerla, "Efficiency/Friendliness Tradeoffs in TCP Westwood" Proceedings of the Seventh International Symposium on Computers and Communications (ISCC'02), IEEE, 2002, pp. 304-311.

[3] Tuan Anh Le, Choong Seon Hong, and Eui-Nam Huh, "Coordinated TCP Westwood Congestion Control for Multiple Paths over Wireless Networks" International Conference on Information Networking (ICOIN), IEEE, 2012, pp. 92-96.

[4] Chih-He Chiang, Wanjiun Liao, and Tehuang Liu," *Adaptive Downlink and Uplink Channel Split Ratio Determination for TCP-Based Best Effort Traffic in TDD-Based WiMAX Networks*", journal on selected areas in communications, IEEE, vol. 27, no. 2, 2009.

[5] L. Subedi, M.Najiminaini, and L. Trajković,*"Performance E Evaluation of TCP Tahoe, Reno,Reno with SACK, and NewReno Using OPNET Modeler",*Proceeding of International Conference on Methods and Models in Computer Science,IEEE, 2009.

[6] Jeng-Ji Huang, Jin-Fu Chang," *A New Method to Improve the Performance of TCP SACK over Wireless Links*",IEEE, 2003.

[7] Kun Tan Jingmin, Song Qian Zhang, Murari Sridharan, " A Compound TCP Approach for High-speed and Long Distance Networks" 25th IEEE International Conference on Computer Communications. Proceedings, INFOCOM, Dec. 2006**.**

