

Enhanced TCP Friendly Congestion Control Protocol

Jennifer Cecilia Rana A.

Abstract—This paper mainly aims in the improvement of the congestion control mechanism over mobile WiMAX in an adhoc environment. Here we compare the throughputs of three different TCP congestion control algorithms namely Westwood+, Veno and New Reno. This paper focused on the performance variation of three TCP congestion control algorithms caused by both handovers and random packet losses over mobile WiMAX. We have enhanced the protocol with Active Queue Management (AQM) and Random Early Detection (RED) mechanism of routing. Our ns-2 simulation results show that Enhanced Westwood+ using the handover-aware algorithm has the improved performance up to 10.2% over normal TCP Westwood+.

Index Terms—Mobile W-MAX, TCP congestion control protocols, handover

I. INTRODUCTION

Mobile WiMAX (Worldwide Interoperability Microwave Access) is a wireless system based on the IEEE 802.16e [1] standard. The 802.16-2004 is not implemented support of handovers between cells it allows only fixed and nomadic access. The handover mechanism is implemented in the newest version 802.16e. There is introduced support of soft and hard handovers.

Mobile WiMAX is considered a promising next-generation wireless technology because it has a long transmission range and supports high data rates and handover. Handover Management is the process of initiating and ensuring a seamless and lossless handover of a mobile terminal from the region covered by one base station to another base station. Unlike wired networks, random packet loss is not negligible in wireless networks. Moreover, bursty packet loss occurs during handover.

A. Scope of the System

Most of the wireless technologies have incorporated TCP/IP into their protocol stacks. However, TCP was designed for wired networks. Its sliding window and congestion avoidance mechanisms were designed to avoid routers congestion. In the past years, TCP has been extensively studied and extensions, such as BIC TCP [2] or Fast TCP [3], have been proposed to improve the congestion control mechanisms, especially in high-speed long distance networks. The TCP behavior has also been evaluated in wireless, i.e., 802.11, environment. It resulted, from this evaluation, several extensions to TCP [4]-[7] that make TCP more robust to wireless specific conditions. Indeed, while network congestion is an acceptable assumption for packet losses in many networks, wireless networks might encounter

two additional causes of packet losses [8]. The first one is the random packet loss that manifests itself through bit corruption. Such packets are discarded by the routers or the end-hosts. Second, a disconnection packet loss might occur when a mobile host completely disconnects from the wireless network. Finally, the weather conditions might also affect the signal link quality.

B. Objective and Success Criteria

For all of these reasons and considering that nowadays WiMAX is the most important and promising technology for broadband wireless access, it is very important to study TCP real behavior in one of the pre-WiMAX implementations. Furthermore, none of the studies considered TCP congestion control algorithms on mobile WiMAX which supports handovers between base stations (BS). Therefore, to simulate the effects of the handover on the algorithms over mobile WiMAX.

II. LITRATURE ANALYSIS

Modeling of mobile WiMAX has been an active area of research. However, there are only few studies [3], [4] considering TCP performance in WiMAX or mobile WiMAX. Furthermore, none of the studies considered TCP congestion control algorithms on mobile WiMAX which supports handovers between base stations (BS). Therefore, to simulate the effects of the handover on the algorithms over mobile WiMAX.

Mobile WiMAX module [5] used in our simulation has the following features. It uses the time division duplex (TDD) mode and orthogonal frequency division multiplexing (OFDM). It supports a neighbor advertisement, scanning and handover. The handover is processed as follows. According to a signal power level and a packet loss rate, a mobile node (MN) generates two events. One is link down event and the other is link going down event. When one of the events is triggered, the MN performs the channel scanning operation. If an appropriate target BS is found, the handover takes place and the MN communicates with the new target BS.

When the handover takes place, a bursty packet loss occurs. There are also relatively high packet losses in the wireless network compared to the wired network. However, New Reno algorithm considers these kinds of losses as a result of a congestion and decreases the congestion window size. Instead of the blind reduction of the window size, Westwood+ [1] and Veno [2] can identify the cause of packet loss. Westwood+ adjusts the window size by bandwidth estimation and Veno adjusts the size by using the estimated state of a connection based on TCP Vegas. The following section presents the ns-2 simulation results which show the effect of handover on the algorithms over mobile WiMAX networks.

III. RELATED ANALYSIS

A. Handoff with TCP NewReno

As is shown in Fig. 1, Fig. 2. 1 and Fig. 2. 2, TCP New Reno improves retransmission during the fast recovery phase of TCP Reno. The New Reno can send new packets at the end of the congestion window during fast recovery, high throughput is maintained during the hole-filling process, even when there are multiple holes, of multiple packets each. When this sequence number is acknowledged, TCP returns to the congestion avoidance state.

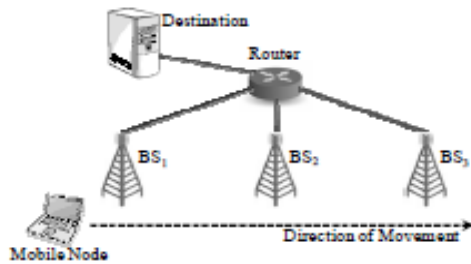


Fig. 1. Architecture.

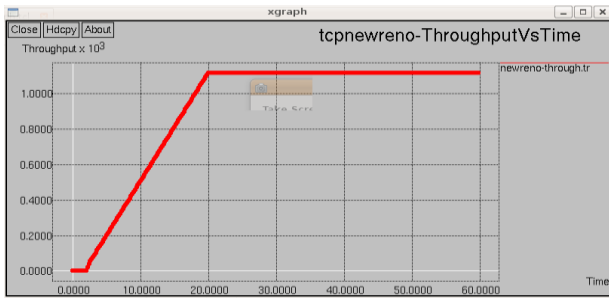


Fig. 2. 1. TCP NewReno throughput.

A problem occurs with New Reno when there are no packet losses but instead, packets are reordered by more than 3 packet sequence numbers. When this happens, New Reno mistakenly enters fast recovery, but when the reordered packet is delivered, ACK sequence-number progress occurs and from there until the end of fast recovery, every bit of sequence-number progress produces a duplicate and needless retransmission that is immediately ACKed. New Reno performs as well as SACK at low packet error rates, and substantially outperforms Reno at high error rates.

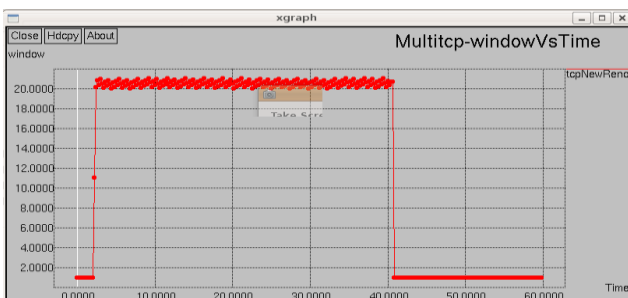


Fig. 2. 2. TCP NewReno congestion window ratio.

B. Handoff with TCP Veno

As is shown in Fig. 3. 1 and Fig. 3. 2, TCP Veno is a novel end-to-end congestion control scheme which can improve TCP performance quite significantly over heterogeneous networks, particularly when wireless links

form part of such networks [9]. The salient feature of TCP Veno is that it only needs simple modification at sender side of Reno protocol stack. Considering practical issues deployability and compatibility, Veno TCP may be quickly deployed in "hot" Mobile Internet industry.

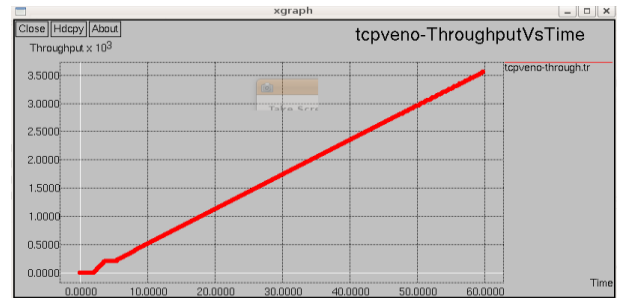


Fig. 3. 1. TCP Veno throughput.

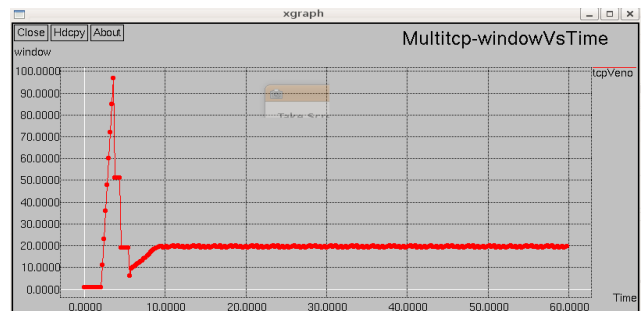


Fig. 3. 2. TCP Veno congestion window ratio.

Distinguishing between congestion loss and random loss, and providing different measures to deal with them, is a fundamental problem that remains unsolved for TCP. If packet loss is detected while the connection is in the congestive state, Veno assumes the loss is due to congestion; otherwise, it assumes the loss is random.

C. Handoff with TCP WESTWOOD+

As is shown in Fig. 4. 1 and Fig. 4. 2, TCP Westwood (TCPW), is a sender-side-only modification to TCP NewReno that is intended to better handle large bandwidth-delay product paths (large pipes), with potential packet loss due to transmission or other errors (leaky pipes), and with dynamic load (dynamic pipes). In TCPW, an "Eligible Rate" is estimated and used by the sender to update ssthresh and cwin upon loss indication, or during its "Agile Probing" phase.

Under a more appropriate criterion for friendliness, i.e. "opportunistic friendliness", TCPW is shown to have good, and controllable, friendliness. The BSs communicate with the MN over mobile WIMAX and have a 500 meter radius of coverage. The MN moves from left to right during the simulation. Therefore, in this scenario the MN performs two handovers. The traffic model of MN is a TCP constant bit rate (CBR) model. The traffic starts at 10 seconds. We repeated the simulation with New Reno, Westwood+ and Veno. To create a more realistic simulation model, we imposed a background traffic load between the router and the destination node using a UDP CBR model.

When the handover takes place, a bursty packet loss occurs. There are also relatively high packet losses in the wireless network compared to the wired network. However, New Reno algorithm considers these kinds of losses as a result of a congestion and decreases the congestion window

size. Instead of the blind reduction of the window size, Westwood+ and Veno [2] can identify the cause of packet loss. Westwood+ adjusts the window size by bandwidth estimation and Veno adjusts the size by using the estimated state of a connection based on TCP Vegas. From the Simulation congestion window sizes of TCP Westwood+ when the packet loss rate is 1% and the MN moves with the speed of 10m/s. These imply two handovers on mobile WiMAX network.

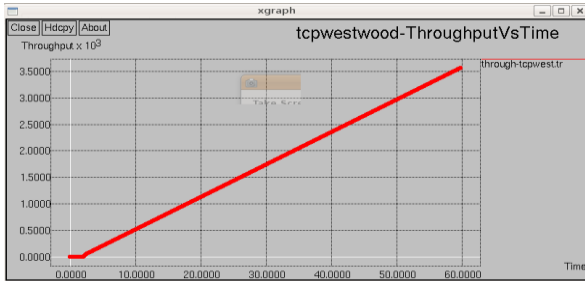


Fig. 4. 1. TCP westwood+ throughput.

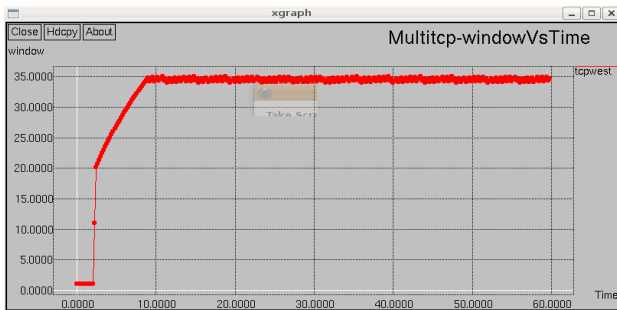


Fig. 4. 2. TCP westwood+ congestion window ratio.

IV. RESULT AND ANALYSIS

TABLE I: PERFORMANCE COMPARISON

Protocol	Packet Loss Detection	Algorithms	Efficiency	Lacks
NewReno	NO	AIMD	Fast Recovery	Congestion detection
TCP Veno	No	RTT Estimation	Wireless End To End Support	Random Packet Loss
Westwood+	Yes	AIMD + Sender Side RTT Estimation	High Throughput Parallel TCP Control	Friendliness Random packet loss

We show throughputs of the three algorithms measured in every 20 seconds (see Table I). Throughputs of all the three algorithms clearly decreased at 60 and 140 seconds. This is due to the handovers that Westwood+ recalculates the congestion window sizes starting from 1 after the handover. However, since there is no congestion, it does not need to calculate the window size starting from 1. Therefore, if we

keep the congestion window size used before handover, it can be used again to calculate the window size after the handover. A handover can be automatically detected by monitoring the attenuation of signal strength. If we apply a handover-aware algorithm which keeps the previous congestion window size when a handover occurs, we can obtain an improved result.

We measured the throughputs during the entire simulation time. Westwood+ and Veno have an improved performance up to 16.5% and 5.4% over New Reno, respectively. In addition, when we added a handover-aware algorithm to Westwood+, it has more improved performance up to 17.8% over New Reno (see Fig. 5. 1 and Fig. 5. 2).

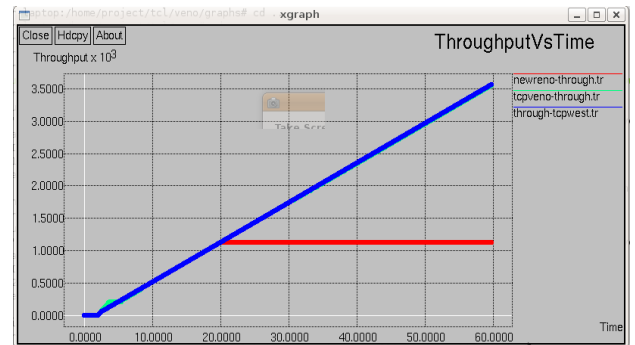


Fig. 5. 1. TCP protocols comparative throughput.

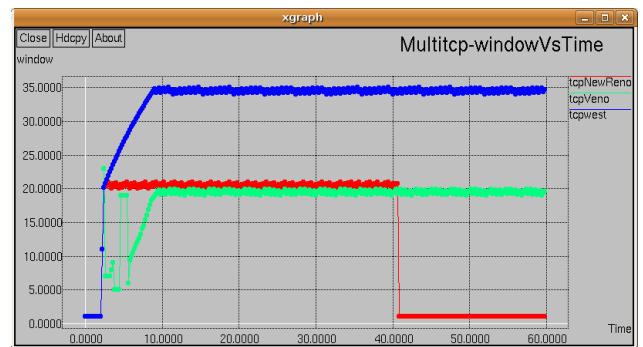


Fig. 5. 2. TCP protocols comparative congestion window ratio.

A. Proposed Enhanced TCP Friendly Congestion Control Active Queue Management and Random Early Detection

The network, in particular the routers in the network, should play an active role in its resource allocation, so as to effectively control/prevent congestion. This is known as active queue management (AQM). The essence is that an AQM router may intelligently drop packets before the queue overflows.

Among various AQM schemes, Random Early Detection (RED) is probably the most extensively studied. RED is shown to effectively tackle both the global synchronization problem and the problem of bias against bursty sources. RED shows some advantages over droptail routers but it is not perfect, mainly due to one or more of the following problems.

- RED performance is highly sensitive to its parameter settings. In RED, at least 4 parameters, namely, maximum threshold (maxth), minimum threshold (minth), maximum packet dropping probability (maxp), and weighting factor (wq), have to be properly set.
- RED performance is sensitive to the number of competing sources/flows.

- RED performance is sensitive to the packet size.
- With RED, wild queue oscillation is observed when the traffic load changes.

In are proposed TCP WestWood+ protocol which is rate based congestion control and friendly. We named our new protocol ETFCC (Enhanced TCP Friendly Congestion control). ETFCC is a congestion control mechanism designed for unicast flows operating in an Ad-hoc environment and competing with TCP traffic. It is designed for applications that use a fixed packet size, and vary their sending rate in packets per second in response to congestion. ETFCC is a receiver-based mechanism, with the calculation of the congestion control information (i.e., the loss event rate) in the data receiver rather in the data sender. This is well-suited to an application where the sender is a large server handling many concurrent connections, and the receiver has more memory and CPU cycles available for computation. Packet-dropping Behavior at WIMAX with AQM (see Fig. 6. 1 and Fig. 6. 2).

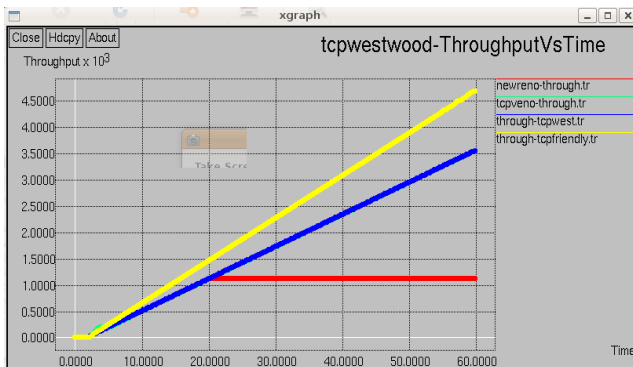


Fig. 6. 1. Proposed TCP protocols comparative throughput.

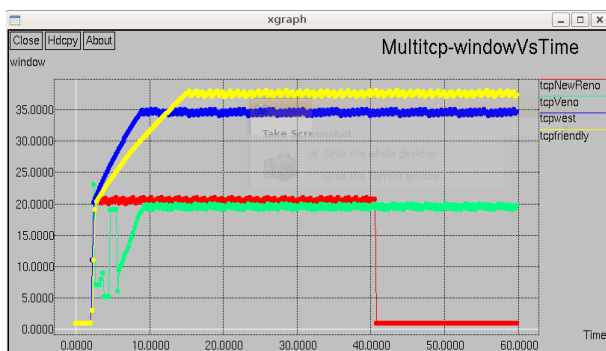


Fig. 6. 2. Proposed TCP protocols comparative congestion window ratio.

As expected, the packet-dropping behavior also can be varied by varying the Active Queue Management (AQM) mechanism in the router. When the routers use RED (Random Early Detection), there are several parameters than can affect the packet-dropping or marking behavior as a function of the packet size.

First, as with Drop-Tail, the RED queue can be in units of either packets or bytes. This can affect the packet-dropping behavior when RED is unable to control the average queue size, and the queue overflows.

Second, and orthogonally, RED can be configured to be either in packet mode or in byte mode. In packet mode, each packet has the same probability of being dropped by RED, while in byte mode, each byte has the same probability of being dropped. In packet mode, large-packet and small- packet flows receive roughly the same packet

drop rate, while in byte mode, large-packet and small-packet flows with the same throughput in bps receive roughly the same number of packet drops.

This paper compares the throughputs of three different TCP congestion control algorithms: Westwood+, Veno and New Reno over mobile WiMAX. Especially, this paper focused on the performance variation of three TCP congestion control algorithms caused by both handovers and random packet losses over mobile WiMAX. Our ns-2 simulation results show that Enhanced Westwood+ using the handover-aware algorithm has the improved performance up to 10.2% over normal tcp Westwood+. Nevertheless, this paper represents our first step towards a good comprehension of TCP within WiMAX. Our future work is to reproduce TCP behavior in Wimax with a model implemented security aspects. Further investigations and measurements are needed to achieve this goal.

V. CONCLUSION

This paper has provided the required details that the TCP Westwood + protocol when enhanced with AQM and RED can reduce the random packet loss drawback and has improved the efficiency of the protocol by 10.2%.

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Jennifer Cecilia Rana A. has completed her bachelors in Computer Science and Engineering from Jeppiaar Engineering College, Anna University in the year 2008 and masters in Information Technology with Networking specialization from VIT University, Vellore in the year 2010. She is currently working as systems engineer in Infosys Limited, Chennai.

Jennifer has been a part of the paper reviewing committee for the conferences ICCSIT 2011, ICEEE2012, ICCNDT 2012.