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Analysing the Effects of Transmission Range of Base Station on Mobile WiMAX Network Transport Layer Protocols

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Abstract: All wireless cellular networks suffer with reliability issue which is directly connected with the Transport Layer Protocols, and also selection of proper Transmission Ranges for the Base Stations. In this paper we study through extensive simulation scenarios, the effects of Transmission Range of BSs for two prominent Transport Layer Protocols- Transmission Control Protocol (TCP) and User Datagram Protocol (UDP) respectively over mobile WiMAX networks. The NIST WiMAX module is used to configure WiMAX environment in NS-2. The QoS metrics used to evaluate the performance are Packet Delivery Ratio (PDR), Throughput, Normalized Routing Overhead (NROH) and Average End to End Delay. Simulation results reveal that performance increases with increasing the transmission range of the BSs up to a certain range, after that range it degrades. Throughputs are almost similar for both TCP and UDP but TCP shows more stability and its reliability may prefer it for most of the applications.

Keywords: Mobile WiMAX, TCP, TCP New Reno, Transmission Range, Transport Layer Protocols, UDP.

I. INTRODUCTION

After experiencing a long time domination of Wire-line Broadband Networks, Broadband Wireless Access (BWA) has emerged now as a promising solution for “last mile” access technology to provide high speed Internet connections. Worldwide Interoperability for Microwave Access (WiMAX), which implements the Institute of Electrical and Electronics Engineers (IEEE) 802.16 standard for BWA, is most probably the finest version of that solution. The IEEE standard 802.16e, also known as Mobile WiMAX, standardizes WiMAX for mobile environment. Mobile WiMAX theoretically is capable of providing data transmission of up to 74 Mbps, which is well enough to fulfill today’s real time multimedia communication requirements [1], [2]. But in actual implementations, this data rate is not achieved due to the unpredictable nature of the wireless environment, the moving obstacles within the cell area, limitations of Mobile Stations’ transmission power and their speed variations. Transmission Ranges of the Base Stations have some direct effects on the performance of any wireless networks. Higher Transmission Range may result in better performance in scenarios where most of the nodes are moving with high speeds. Since network capacity is directly connected with Transmission Range, there is also a certain limit for this. It is obviously a critical job to select an optimum Transmission Range for a certain cell. Another important factor that a network vendor must consider is the amount of reliability that it needs to offer to the subscribers. Reliability is the sector dealt by the fourth layer in the OSI network model. Some fourth layer protocols, such as User Datagram Protocol (UDP) does provide very little features for reliability to make the data transmission faster. Some others protocols, such as Transmission Control Protocol (TCP), provides salient features to ensure the network reliability, but compromises data speed. The mobile WiMAX network vendors need to make the right choice concerning Transport Layer Protocols depending on their reliability criteria that they have planned to provide their subscribers.

II. AN OVERVIEW OF MOBILE WIMAX, TCP AND UDP

A. *An Overview of Mobile WiMAX*

Mobile WiMAX or IEEE 802.16e-2005 offers a true broadband connection that supports multiple usage scenarios, including fixed, portable and mobile access, using the same network infrastructure. Some of the salient features of Mobile WiMAX are as follows:

1) *High Peak Data Rates:*

WiMAX supports very high peak data rates. The peak PHY data rate can be as high as 74 Mbps when operating at 20 MHz, and 25 Mbps when operating at 10 MHz frequency [2].

2) *Support for Quality of Service (QoS):*

The support for quality of service is a fundamental component of WiMAX MAC layer design. QoS is achieved using connection oriented MAC architecture where all uplink and downlink traffics are controlled by the serving BSs **[2]**. The QoS parameters include traffic

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priority, maximum delay, tolerated jitter, ARQ technique, scheduling algorithm, etc.

3) Adaptive Modulation and Coding facilities:

WiMAX supports a number of modulation and coding schemes such as BPSK1/2, QPSK1/2, QPSK3/4, 16QAM1/2, 16QAM3/4, 64QAM1/2, 64QAM2/3, etc [2]. Here m/n stands for convolution codes where m data bits are coded to generate an n bit code, m≤n. These modulation schemes are allowed to be changed based on channel condition to maximize the throughput in Mobile WiMAX.

4) Supports Mobility:

As the name implies, Mobile WiMAX definitely defines a framework to support mobility. It does the mobility management with the help of two basic mechanisms. In particular, it defines mechanisms for tracking SS as they move from one base station to another and also protocols to enable an optimized handover which requires less time [2].

5) Strong Security:

WiMAX supports two encryption schemes which are Advanced Encryption Standard (AES) and Data encryption Standard (triple DES). New high performance coding schemes such as Low-Density Parity Check (LDPC) codes are also included **[2]**. These features enhance the security of the mobile WiMAX air interface.

6) Uses Orthogonal Frequency Division Multiple Access (OFDMA):

OFDMA is a multiuser version of OFDM scheme supported by mobile WiMAX. In this scheme, different users are allocated different subset of OFDM subcarriers. It offers frequency diversity by spreading the carriers all over the allocated spectrum, which significantly increases system capacity [2].

7) Scalability:

Mobile WiMAX provides scalability by means of the OFDMA scheme, in which Fast Fourier Transform (FFT) size can be scaled based on the available channel bandwidth. It can optionally support channel bandwidths ranging from 1.25 MHz to 20 MHz, which makes deployment relatively easy [2].

B. An Overview of TCP and TCP New Reno

The Transmission Control Protocol (TCP) is a connection-oriented, end-to-end reliable protocol. It is connection-oriented because it must set up a connection between two processes before one application process can begin to send data to another, which is also known as "three-way-handshake" as three types of preliminary segments are transferred in this phase to establish the parameters of the data transfer [3]. Process-to-Process Communication, Stream delivery services, full duplex communication and reliability are the services offered by TCP to process at application layer. It ensures reliability by sliding window, acknowledgement and retransmission schemes [3]. TCP is used by major Internet applications such as the World Wide Web, email, remote administration and file transfer.

An additional feature of TCP is its congestion control algorithms. These algorithms prevent a sender from overrunning the capacity of the network. TCP can adapt the sender's rate to network capacity and attempt to avoid potential congestion situations. Several congestion control enhancements have been added and suggested to TCP over the years. Modern implementations of TCP contain following four intertwined algorithms as basic Internet standards [3]-[5]:

- Slow Start
- Congestion Avoidance
- Fast Retransmit
- Fast Recovery

These algorithms introduce a window called Congestion Window (CWND). The sender can transmit the lower value of the congestion window or the advertised window sent by the receiver. **Slow Start** is designed to increase the congestion window after a connection is initialized and after a timeout. It is also known as the exponential growth phase. The algorithm begins in the exponential growth phase initially with a congestion window size (CWND) of 1, 2 or 10 segments and increases it by 1 Segment Size (SS) for each ACK received. If the receiver sends an ACK for every segment, this behavior effectively doubles the window size each round trip of the network. This happens until either an acknowledgment is not received for some segment, which is indicated by duplicate ACKs, or a predetermined threshold (SSThresh) value is reached. Once the CWND reaches the SSThresh, TCP goes into **Congestion Avoidance** mode where each ACK increases the CWND by SS*SS/CWND. This results in a linear increase of the CWND. To avoid waiting for a Time Out to occur, **Fast Retransmit** is employed. In this stage, after three sucessive duplicative ACKs, the sender assumes that the segment was lost, retransmits the segment and moves to **Fast Recovery** phase. In Fast Recovery, the sender decreases Congestion Window (CWND) twice of its original size, adds 3 (3 packets have left the network and buffered by the receiver) and continue to send new segments (if allowed by the CWND value) until receiving new different ACK, which should acknowledge receiving all segments sent till moving to Fast Recovery phase (assuming that no more segments were lost) [4].

The limitations of these basic algorithms are that if cwnd size is too small (smaller than 4 packets) then it's not possible to get 3 duplicate ACKs and run the First Retransmit algorithm. Besides, the algorithm cannot manage a loss of multiple packets from a single window of data, which may cause a time out. These algorithms also do not manage a loss of packets during the Fast Recovery stage.

To mitigate some of these limitations, TCP New Reno has made some modifications. It forces the sender to remember the number of the last segment that was sent before entering the Fast Retransmit phase. Then it can deal with a situation when a "new" ACK (which is not duplicate ACK) that does not cover the last remembered segment (called "partial ACK") is received. This partial

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ACK indicates that more segments were lost before entering the Fast Retransmit. In TCP New Reno implementation, after discovering such situation the sender will retransmit the new lost packet too and will stay at the Fast Recovery stage. The sender will finish the Fast Recovery stage when it will get ACK that covers last segment sent before the Fast Retransmit. These simple modifications let the TCP New Reno to overcome Multiple Packet loss from a single window [5].

C. An Overview of UDP

The User Datagram Protocol (UDP) is connectionless, unreliable transport layer protocol. It is formally defined in RFC 768 [3], [6]. UDP is connectionless since it lets the applications to send messages, referred to as datagrams, without prior communications to set up special transmission channels or data paths. UDP communication consequently does not incur connection establishment and teardown overheads and therefore the overhead is minimal. This connectionless behavior of UDP also makes it unreliable. UDP offers minimal transport layer functionalities non-guaranteed data delivery and gives applications a direct access to the network layer. It does not add anything to services of IP except to provide process-to-process communication instead of host-to-host communication [3]. It only performs multiplexing/de-multiplexing functions and some light error checking by means of checksum calculation. UDP does not provide any delivery report to the sender and neither has it provided any mechanism for duplicate protection or to detect and reorder out of order datagrams. Functionalities for Congestion Control are also absent in UDP. All these lacks of features of UDP are to provide high speed communication to the applications. UDP simply takes messages from an application process, attaches source and destination port for the multiplexing/de-multiplexing service, adds two other fields of calculated checksum and length information, and passes the resulting packet to the network layer [3]. The network layer encapsulates the UDP packet into IP datagram and then delivers the encapsulated packet at the receiver. When a UDP packet arrives at the receiving host, it is delivered to the receiving UDP agent identified by the destination port field in the packet header.

III. MAIN CONTRIBUTIONS OF THIS PAPER

Based on the simulation results, this paper presents an analysis of the behavior of the two Transport Layer Protocols, TCP and UDP, with the change of Transmission Ranges of Base Station over Mobile WiMAX network. This research work may guide the Mobile WiMAX service providers to figure out the appropriate Transmission Ranges and Transport Layer Protocol to select to provide their expected network performance.

IV. EFFECTS OF TRANSMISSION RANGE IN WIMAX NETWORKS

In Mobile WiMAX networks, the transmission range of the base station significantly influences the network performance, especially in areas where most of the users are highly mobile. In such areas, using the same transmission range as indicated by the WiMAX forum (1.4 km) may cause frequent handovers of mobile nodes between base stations which may result in a large number of packets losses. Obviously this refers to use higher transmission range of Base Stations in these particular areas. But it is not efficient in term of capacity. Besides, higher transmission range may also increase the traffic load due to having larger number of nodes within a cell. That means to select an appropriate transmission range for a particular scenario is a critical job. The designers must calculate an optimum range that will avoid unnecessary handovers and at the same time must provide the network with maximum capacity.

V. TECHNIQUES FOR ACHIEVING HIGHER TRANSMISSION RANGES IN MOBILE WIMAX NETWORKS AND THEIR LIMITATIONS

Achieving higher Transmission Ranges of Base Stations in mobile WiMAX networks is possible by employing a number of techniques including **high transmit power**, **subchannelization** and **adaptive modulation.** But there are some restrictions that put some limits on applying those techniques. This section briefly describes those techniques and their limitations are also illustrated.

A. High Transmission Power

It is obvious that Radio Frequency (RF) power translates directly into range, so higher transmission power equals longer range. To achieve the range indicated by WiMAX forum, WiMAX base stations transmit at power levels of approximately +43dBm (20W), while a WiMAX mobile station typically transmits at +23 dBm (200mW) [7]. To get higher transmission ranges, the transmission powers of both the base station and mobile nodes are needed to increase. But there are three important factors that limit the ability to transmit at higher power:

- Power Amplifier (PA) efficiency
- Available supply voltage
- Regulatory requirements

In PAs, efficiency is the measure of the RF power out versus the DC power in [7]. For example, if a PA has 10 percent efficiency, it would consume 3.55 W to transmit at +25.5dBm (355 mW). If the PA efficiency could be doubled to 20 percent, then the peak power consumption drops to 1.7W. Today's state-of-the-art WiMAX Power Amplifiers, like SiGe Semiconductor's SE7262, operate with about 20 percent efficiency [7]. The PA efficiency has a direct impact on battery life for mobile devices. To emit higher power, if PA efficiency is the same, a WiMAX mobile device will discharge battery more quickly, which distract the use of higher transmission power.

Mobile WiMAX devices are powered directly from the mobile station's battery, and battery supply voltages vary significantly during use. A fully charged battery can operate at about 4.8V. The supply voltage drops with consumption and the minimum practical supply voltage before the device shuts down is typically 2.7V. Manufacturers of Mobile Station want to ensure maximum

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use of the battery and therefore specify that the power amplifier must faithfully deliver fully rated power at 3.3V (and occasionally 3.0 V) [7]. Under these circumstances, delivering high power is much difficult due to the requirement of a low supply voltage which requires a high current and implies a very low output impedance. Matching this low impedance PA output to a 50 Ohm antenna is difficult to achieve. It becomes more difficult when higher output powers are required, because then the impedance becomes further low which makes it very difficult to achieve a good broadband match between the PA and the antenna.

In real-world implementations, PA non-linearity introduces out-of-band frequencies through Inter-Modulation distortion (IMD) which can interfere with users in adjacent channels. Regulatory bodies have imposed strict regulations on the amount of power that can be emitted out of band. For example, for mobile devices in the 2.5GHz band, the Federal Communications Commission (FCC) specifies that the emissions must be below -25dBm/MHz, measured 5.5MHz outside the device's assigned band [7]. When output power is increased, more and more rejection of out-of-band emissions is required, and the power amplifier must be made more and more linear, which will eventually drop PA efficiency [7].

Recognizing the Tradeoffs undoubtedly, higher transmit power is important for mobile WiMAX networks. As mentioned earlier, networks are currently being deployed specifying that the minimum transmit power is +23dBm [8].

B. Subchannelization and Adaptive Modulation

As like all others cellular wireless networks, there is a large difference (approximately 20 dB) between downlink power (from the BS to the MS) and uplink power (from the MS to the BS) in mobile WiMAX networks. This means that, it is relatively difficult for the BS to hear the mobile stations. One way to combat this mismatch is by using subchannelization technique, where only a subset of all of the available subchannels is used for any particular user [7]. In effect, each mobile concentrates its power over a smaller range of frequencies, and the net signal gain is $10*\log(N_{total}/N_{used})$, where N_{used} is the number of subcarriers assigned to the user, and N_{total} is the total number of subcarriers available.

Another technique to address the link imbalance is adaptive modulation [7]. In this case, the mobile transmits using a lower order modulation compared to the BS. For example, the mobile could transmit Quadrature Phase Shift Keying (QPSK) or 16QAM signals, while the BS transmits using 64QAM. Because the SNR required to receive QPSK or 16QAM is lower than 64QAM, using a lower order modulation allows the MS to communicate with the BS using less transmit power [7].

When subchannelization and adaptive modulation are combined, a network operator can effectively balance the uplink and downlink budgets, and the network will operate bi-directionally [7]. The downside is that when these techniques are used, the uplink throughput will be lower than the downlink throughput; subchannelization limits the number of subcarriers available for mobile transmission, and lower order modulation means that fewer bits are transmitted on each available subcarrier, eventually reduces the throughput.

VI. METHODOLOGY

A. Simulation Environment

All the result of this study is based on simulations using the network simulator (NS-2) from Lawrence Berkeley National Laboratory (LBNL) in Red hat 5.0 platform [9]. For the simulation of WiMAX network; a patch "WiMAX Module" from National Institute of Standards and Technology (NIST) is used, which implements the MAC layer (IEEE 802.16) and PHY (OFDMPHY) layer for creating WiMAX environment [10]. As QoS specification, only Best Effort (BE) service class is used. Best effort services are appropriate for applications such as web browsing and file transfers since these can tolerate intermittent interruptions and reduced throughput without serious consequence. To evaluate Simulation result we set the length of each simulation to 210 seconds. The traffic starts at 100 second to provide time for initial ranging and other synchronization and authentication. In the simulation area 10 mobile nodes can move randomly.

DSDV has been used as the routing protocol with an Interface Queue (IFQ) of 50 packets. The IFQ is a First in First out (FIFO) priority queue where routing packets gets higher priority than data packets. DSDV is chosen due to its better performance in mobile WiMAX environment [7], [11]. The propagation model that is used in this research paper is TwoRayGround propagation model. As the mobility model, Random Waypoint Mobility (RWM) model is used. Pause time is selected as zero, which means the WiMAX network considered in this simulation work is highly dynamic. All Mac and network layer operations of the wireless network interfaces are logged in trace files. Post simulation analyses are performed to each of the trace file by using Perl language.

To investigate the effect of the transmission range on the protocols performance in WiMAX network, a traffic scenario is kept fixed for the set of Transmission Ranges that have been considered and simulation is run for those ranges. This process is repeated for two other different scenarios. After each simulation the Packet Delivery Ratio (PDR), Throughput, Normalized Routing Overhead (NROH) and Average End to End (E2E) Delay values for both TCP and UDP are measured. Then the average values are calculated for generating the result graphs.

B. Simulation Parameters

The simulation parameters that are used in this simulation study for analysing the effects of Transport Layer protocols in mobile WiMAX network are provided in Table I.

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TABLE I
SIMULATION PARAMETERS

No. of Mobile Stations (MSs)	10
Minimum speed of MSs(m/s)	1
Maximum speed of MSs (m/s)	20
Base Station (BS) Height (m)	32
Mobile Station Height (m)	1.5
BS Transmission Power(dB)	43 (20 W)
BS Transmission Range (meters)	400-2000 (with an interval of 100)
Operating Bandwidth (GHz)	2.412
RXThreshold	1.225e-08, 7.837e-09, 5.442e-09, 3.999e-09, 3.061e-09, 2.419e-09, 1.959e-09, 1.619e-09, 1.361e-09, 1.159e-09, 9.996e-10, 8.708e-10, 7.653e-10, 6.780e-10, 6.047e-10, 5.427e-10, 5.898e-10 respectively.
Packet size (Byte)	1520
Traffic	TCP/FTP, UDP/CBR

VII. PERFORMANCE METRICS

To evaluate the performance of the Transport Layer Protocols, we use four different quantitative metrics to compare the performance. They are:

A. Packet Delivery Ratio (PDR)

PDR is the ratio of data packets delivered to the destination to those generated by the sources and is calculated as follows [12]:

$$\text{PDR}=\frac{Number\ \ of\ Packets\ \ Received}{Number\ \ of\ Packets\ \ Sent}\ \text{x}\ 100 \quad (1)$$

B. Throughput

Throughput is the number of data bytes received successfully and is calculated by [7], [12]:

$$\text{Throughput}=\frac{Number\ \ of\ bytes\ received\ \ x8}{Simulation\ \ \ time\ \ x\ 1000000}\ \text{Mbps} \quad (2)$$

C. Normalized Routing Overhead (NROH)

Normalized Routing Overhead is the number of routing packets transmitted per data packet towards destination and calculated as follows [12]:

$$\text{NROH} = \frac{Number\ \ of\ Routing\ \ Packets\ \ Received}{Number\ \ of\ Packets\ \ Received} \quad (3)$$

D. Average End to End (E2E) Delay

Average End-to-End delay is the average time of the data packet to be successfully transmitted across a network from source to destination. It includes all possible delays such as buffering during the route discovery latency, queuing at the interface queue, retransmission delay at the MAC (Medium Access Control), the propagation and the transfer time, processing time at Transport Layer [12]. The average e2e delay is computed by,

$$\text{D} = \frac{\sum_{i=1}^{n} Ri-Si}{n}\ \text{m sec} \quad (4)$$

Where n is the number of data packets successfully transmitted over the network, ' i ' is the unique packet identifier, Ri is the time at which a packet with unique identifier ' i ' is received and Si is the time at which a packet with unique identifier ' i ' is sent.

VIII. RESULT ANALYSIS AND DISCUSSION

The graphs given below (Fig. 5.1-5.4) show the effect of transmission range on the Performance metrics (PDR, Throughput, NROH and Average E2E Delay) for both Transport Layer protocols.

A. Results Analysis- Transmission Range vs. Packet Delivery Ratio (PDR)

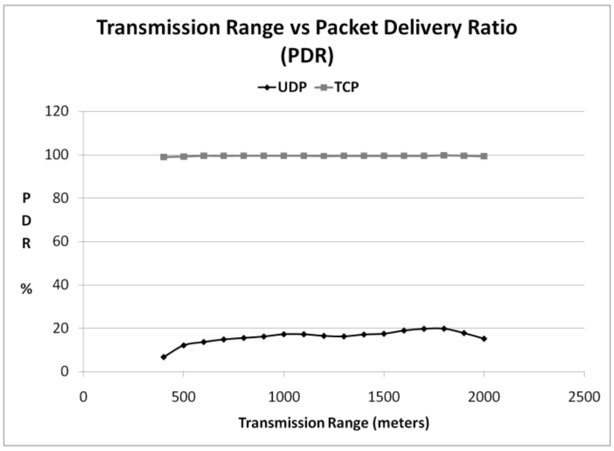

Fig. 1. Packet Delivery Ratio (PDR) as function of the Transmission Range

Fig. 1 shows the PDR values for both TCP and UDP, where TCP maintains almost constantly ratios that are very close to 100%. TCP PDRs are not affected significantly by the transmission rang variations. But UDP performs poorly in terms of PDR when compared with TCP and its curve has some fluctuations as the figure illustrates.

The reliability provided by TCP acknowledgements, retransmissions and Congestion Control Algorithms ensures the proper reception of segments by desired receivers, which results in the excellent and stable PDR curve for TCP. The poor values of UDP PDR curve is the reflection of its unreliable nature. Due to absence of acknowledgements and retransmission facilities, the connection less UDP could not provide any guarantee that the packets are received by their proper destinations and even provide any information about dropped packets. These lacks of functionalities causes highly congested receiver queues and eventually packet losses and poor PDR values in case of UDP. The instability of UDP PDR curve illustrates the effects of mobile nodes that are on the cell edges or near the edges. The packets sent by these nodes posses low powers at the receiver end and some of these packets are dropped for being under the receiving power threshold. The time required processing these low

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power packets that might be dropped kept the receiver busy and some other packets get stuck on that particular receiver's queue. As in UDP, the number of sent packets is much greater than TCP due to no congestion control function; UDP is more sensitive to this event. This makes the UDP network unstable.

B. Results Analysis- Transmission Range vs. Throughput

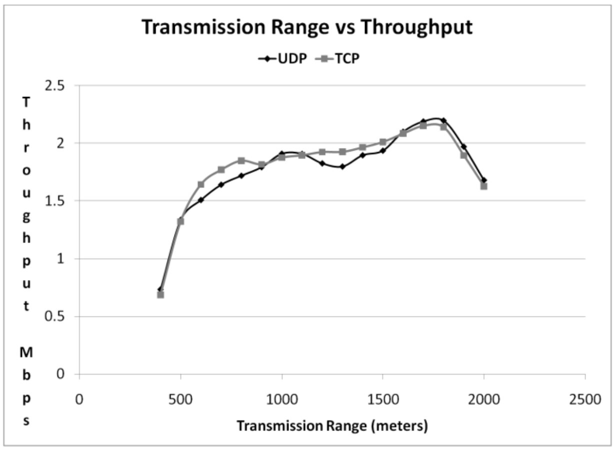

Fig. 2. Throughput as function of the Transmission Range

The throughputs for both the protocols that were found during simulation while transmission range was gradually increased with an interval of 100m are presented in Fig. 2. According to the graph, TCP gains better throughputs than UDP at lower Transmission ranges. At higher ranges, specifically from 1600 meters range, UDP achieves marginally better throughputs. Both the curve goes upward with increased transmission ranges till 1800 meters range. This upward portion of the curves is relatively smoother for TCP compared to UDP. UDP throughputs curve shows lack of stability due to the unpredictable nature of wireless network scenarios.

In a scenario of a particular range, most of the nodes may stay near the Base Station while some are outside the cell, in that scenario for the next range; a larger portion of nodes may be found near the cell edges. Since TCP is less sensitive to the effects of edge nodes, its curve shows better stability.

The downward region of the curves shows the effects of increased traffic load as at higher ranges, maximum nodes are well within the cell range. TCP throughput drops because of frequent congestion avoidance phase for aiding congestion problem, while UDP suffers as a result of saturated receiver queues.

C. Result Analysis- Transmission Range vs. Normalized Routing Overhead (NROH)

Fig. 3 illustrates the Normalized Routing Overhead (NROH) curves for uniformly increasing transmission ranges. Both overheads increase with the range increment. TCP overheads are higher than UDP from 1400 meters range, while both the overheads are similar for lower ranges. Both the curves have instable regions, for TCP, it is from 1500 to 1800 meters, 1400 to 1800 meters for UDP. From 1900 meters range, the curves drastically move upward.

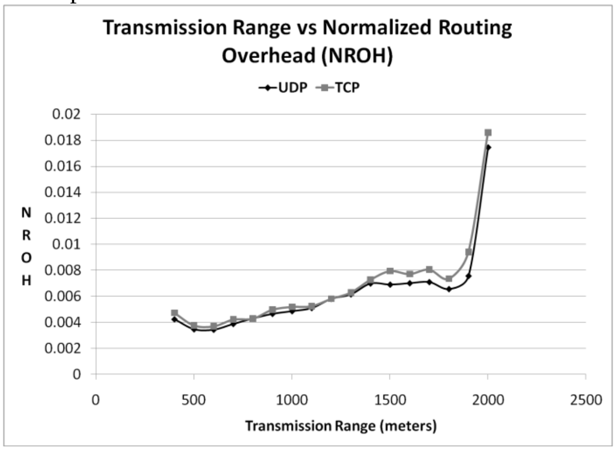

Fig. 3. Normalized Routing Overhead (NROH) as function of the Transmission Range

Number of routing packets transferred depends on the underlying routing protocol. In this simulation study, DSDV is used, which makes both time driven and event driven updates. As TCP transfers not only data packets but also acknowledgements, event driven update is more frequent in TCP network. At lower ranges, this does not exceed UDP overhead since number of sent data packets in UDP network is relatively very higher. But at higher ranges, increased number of nodes makes TCP overheads to climb over UDP overhead values. As the graph shows, both overhead values increases with transmission range since increased number of nodes can be accommodated in a larger cell. The regions of uncertainty are illustrating the influence of the number of edge or near the edge nodes on both protocols. The length of these regions also indicates that UDP is more influenced by these nodes.

D. Results Analysis- Transmission Range vs. Average End to End (E2E) Delay

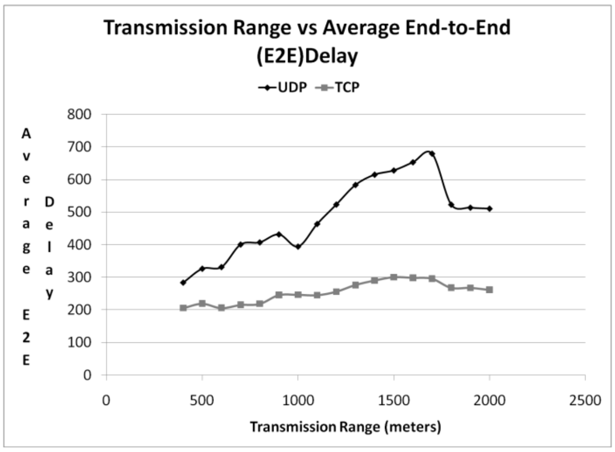

Fig. 4. Average End to End (E2E) Delay as function of the Transmission Range

As the graph in Fig. 4 shows, both TCP and UDP average E2E delay values increase when transmission range is increased up to a certain range. After this range, which is 1700 meters in specific, delay values suddenly start to fall.

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In TCP, retransmission that are caused by edge nodes do not resulted in congestion avoidance stage since in these cases more than three successive duplicate acknowledgements is a rare incident. But retransmissions due to increased traffic load lead the TCP network to congestion avoidance phase. Both of these two events reduce the throughputs. The first event reduces throughputs in the form of retransmitting the same packets several times. But since this does not make any change to the cwnd window, average E2E delay of data packets is not affected by this event. But the second event forces the congestion window cwnd to deflate which reduces the load on the receiver queue and eventually reduces the E2E delay also. In the TCP throughput curve of Fig. 2, 1700 meters and 1800 meters TRs have similar throughputs, but there is a significant reduction in E2E Delay value. This is due to these two events that have been discussed. At 1700 meters, some nodes are near the edges, at 1800 meters, they are well in most of the time, at 1900 meters and 2000 meters, more time to spend within the cell range. As a result, the delay curve starts to decay from 1700 meters range instead of 1800 meters. As opposite of TCP, UDP has greater influence of edge nodes, which causes the Delays to be reduced drastically at 1800 meters range. UDP network suffers greater delays due to the saturated receiver queues and the range of variations of the UDP curve, which is much higher than TCP, illustrates the unstable nature of UDP network.

IX. LIMITATIONS

The NIST WiMAX module that is used in this research for simulation of WiMAX network supports only subchannelization. It does not support adaptive modulation. To achieve an average result, 16QAM-1/2 is used.

X. CONCLUSION

According to the simulation outcomes upon which this paper is based on, TCP clearly performs better in terms of PDR and Delay values. Although UDP is designed as connectionless and it has lack of reliability to provide high data speed, throughput values are found almost similar for both UDP and TCP. The reliability feature of TCP also makes it stable in mobile WiMAX environment. Based on the simulation results it can be stated that TCP should be used for applications where any kinds of reliability is required, even in case of real time, high speed data transmission, TCP may be preferred when network load is less. The results also illustrate that, performance upgrades explicitly while increasing the Transmission Range of the BS for both the protocols up to a certain range. After this range performance degrades. Therefore, to get the optimum performance in a highly dynamic WiMAX network, transmission range should be chosen carefully to obtain the best suited range for the network.

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