
Performance Analysis of Transport Protocols for Multimedia Traffic Over Mobile Wi-Max Network Under Nakagami Fading

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Abstract

Due to increased demand in multimedia streaming over Internet, research focus is directed to derive new protocols with added features. These protocols can be best adjusted to multimedia traffic over the Internet. The operations of multimedia networks require fast and high processing communication systems, because sometimes prompt delivery of information is critical. The behavior of different transport protocol can affect the quality of service. The advancement in hardware technologies used at the physical layer and TCP/IP suite has made a great progress, but other layers could not progress with the same pace. This paper argues the challenges posed by multimedia traffic on other layers and their effect on the transport layer services. Numerous papers have illustrated transport protocols for multimedia networks in some preferred and particular scenarios. However, no one have discussed the effect of channel fading on these protocols. Six existing protocols and their performances under Nakagami- m channel are the theme of this paper with respect to suitability for multimedia applications, and the performances of these protocols have been evaluated. This paper concludes that extra effort are essential on the transport protocol for achieving better performance of multimedia networks which will fulfill the various necessities of evolving multimedia applications, specifically under fading channels.

Keywords

Transport protocol • Nakagami fading • Performance comparison • Wi-max IEEE 802-16e • MPEG-4 video

14.1 Introduction

With the advent of new technologies like smart phones, tablets, and personal digital assistance etc., and advancement in the Internet technologies have demanded a large number of multimedia applications. These applications have

different needs such as in time delivery, bandwidth utilization, and reliability, for fulfilling quality of service requirements. However, some applications require higher delays and uneven bandwidth instead of high bandwidth and high speed networks, i.e. 3G mobile networks or Wi-Fi.

A significant aspect that has greater influence on the quality of service in multimedia applications is the selection of appropriate transport protocol. The conventional applications used for transferring a file, online transaction, or sending an email make use of relatively slow but reliable transport protocol, e.g. Transmission Control Protocol (TCP) [1]. Whereas for multimedia and game applications a fast and partially reliable transport protocol may be a good choice. However, the diverse nature of multimedia traffic

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and communication networks poses different challenges to transport protocol. For example, User Datagram Protocol (UDP) [2] and TCP protocol cannot perform well in high performance wireless mobile networks due to the complex service model compared to wired networks. These complex models are necessary for running multimedia applications on new generation application layers which affects the conventional transport layer. In this context UDP and TCP do not perform upto the mark, therefore, a Real-time Transport Protocol (RTP) [3] is a good choice to transmit multimedia traffic on Internet Protocol.

Normally UDP is used for transportation of multimedia applications due to the delay intolerant nature of these applications, where the main hindrance is need of a mechanism for congestion control. To overcome these issues protocols like Multi-path Transport Protocol (MPTCP) [4], Stream Control Transmission Control Protocol (SCTP) [5], and Datagram Congestion Control Protocol (DCCP) [6] are better options to avail for multimedia traffic transportation. Deployment and maintenance of various wireless access technologies used for transportation of multimedia traffic, are easy compare to wired access technologies. These technologies have diverse factors affecting quality of service, e.g. features of Mobile Wi-Max such as high data rates, scalability, security, and mobility. Wi-Max, UMTS, 4G networks are getting popular and have maximized requirements for multimedia applications. UDP is fast, unreliable, and have no congestion control mechanism which puts it into no quality of service category but still a better choice for video streaming. SCTP have multi-homing, multi-streaming features, better bandwidth utilization with ordered and reliable data transportation. DCCP have mechanism of congestion control with un-reliable data transportation.

The de facto standard for Mobile Wi-Max networks is IEEE802.16e [7], which can operate in mobile and static broadband networks. The IEEE 802.16e have data rate up to 63 Mbps, with improved quality of service. It supports end-to-end Internet Protocol based quality of service in different flows. Another feature of this standard is scalability and flexibility based on the offered services as well as network deployment.

The rest of the paper in accordance to the following pattern. In Sect. 14.2, related work is presented followed by transport protocols in Sect. 14.3. Section 14.4 discusses the Nakagami model for channel fading. Section 14.5 describes mobile Wi-Max technology and MPEG-4 Video traffic, and experimental work and evaluations are given in Sect. 14.6. The conclusion of the paper and future research directions and gaps are elaborated in Sect. 14.7.

14.2 Related Work

In wireless multimedia networks transportation is a hard area due to dynamic nature of multimedia applications and routing protocols. Selection of appropriate routing protocol for

Wi-Max technology is considered as a smart task. A good survey on transport protocols in wireless multimedia networks can be found [8], and real-time transport protocols for wireless networks [9]. The performance of routing protocols used in mobile Wi-Max is studied [10], quality of service mechanism and routing techniques in a mesh-network of mobile Wi-Max is described [11], whereas [12] analyses statistical performance of Wi-Max and Wi-Fi heterogeneous mesh network.

The performance of Wi-Max technologies has been investigated in [13]. A comparison of transport protocol for voice over IP in Wi-Max network is drawn in [14]. The affects of jitter, throughput, delay, and packet loss on video streaming in Wi-Max and ADSL networks is evaluated [15]. Furthermore, packet loss ratio and throughput is evaluated with TCP and its variants using modulation schemes, i.e. BPSK, QPSK, and 16-QAM and 64-QAM. Time is very critical factor in transmission of multimedia applications, even more important than reliability. The transmission of MPEG-4 video is analyzed by using UDP, TCP, and DCCP with her variants in [16], with performance parameter similar to [15]. Similarly, [17] have evaluated the same parameters and traffic as in [16] with SCTP protocol over 802.11 wireless networks. Multimedia traffic parameters like throughput and link-capacity using UDP and TCP is analyzed in [18], then comparison among Wi-Max and High-speed Down-link Packet Access is made. The effect of Nakagami fading channel on transport protocol has been studied in [19–23].

In multi-hop wireless networks throughput is measured based on various window sizes in SCTP protocol [24]. Similarly, [25] have analyzed video streaming in CDMA200 wireless networks using SCTP protocol. A comparative study of various transport protocols for video streaming in 802.11 networks is shown in [26] and in fixed Wi-Max in [14]. In short, different performances of transport protocols using different applications can be found in literature. However, the theme of this paper is to use mobile Wi-Max as the wireless access technology with different transport protocols is a good contribution to this field.

14.3 Transport Protocols

In this section different transport protocol are studied briefly. Table 14.1 show names, features, and services offer by these protocols. Following are the details of these protocols;

14.3.1 Multi-path Transport Protocol (MPTCP)

MPTCP is the extended version of TCP. It mainly focuses on efficient utilization of network resources via multi-homing abilities of end users and provides flexibility network connectivity. The efficient utilization is maximized by

Table 14.1 Features and services offered by DCCP, SCTP, RTP, MPTCP, UDP, and TCP

Features and services	DCCP	SCTP	RTP	MPTCP	UDP	TCP
Connection Oriented	✓	✓	✓	✓	×	✓
Unordered Delivery	✓	✓	✓	×	✓	✓
Reliability	×	✓	×	✓	×	✓
Congestion Control	✓	✓	✓	✓	×	✓
Flow Control	✓	✓	✓	✓	×	✓
Multistreaming	×	✓	×	✓	×	✓
Multihoming	×	✓	×	✓	×	×

concurrent usage of resources on different network interfaces whereas network connectivity is achieved through multi-paths. MPTCP provides two transport functions; network and application oriented. MPTCP is has better congestion mechanism which is fair with TCP flows and it moves traffic from congested links.

14.3.2 Datagram Congestion Control Protocol (DCCP)

DCCP is unreliable connection oriented protocol used by multimedia applications. It combines unreliable and quick services of UDP with controlling congestion by establish a bidirectional uni-cast connection. DCCP have some versions like Quick-Start for DCCP or faster restart for TFRC. DCCP is a good choice for transferring data in bulks with controlled trade-off between timeliness and reliability.

14.3.3 Stream Control Transmission Protocol (SCTP)

SCTP is reliable session oriented multi-homing. It has the ability to recover lost, corrupt, unordered, and duplicate packets by using re-transmission with selective acknowledgment scheme. It has inherited congestion and flow control mechanism from TCP along with mechanisms of slow start, fast recovery, and congestion avoidance. It is secure against flooding and masquerading attacks.

14.3.4 Real-Time Transport Protocol (RTP)

RTP is designed for real-time applications with end-to-end transportation functionalities. It carries data for uni-cast and multi-cast communication, however, this protocol by itself does guarantee quality-of-service (QoS). RTP is a connectionless protocol but is uses the services of RTP Control Protocol (RTCP) for achieving QoS.

14.3.5 User Datagram Protocol (UDP)

UDP is unreliable connectionless, message-oriented core member of the TCP/IP suite with no error, flow, and congestion control mechanism used in uni-cast/multi-cast networks.

14.3.6 Transmission Control Protocol (TCP)

TCP is the core member of TCP/IP suite, a reliable and connection oriented most widely used transport protocol. It has mechanism for flow control, error control, congestion control, error reporting, and error correction. TCP has many versions with better congestion control mechanism such as FAST TCP, TCP Vegas, and TCP Reno. TCP has extensions for mobile wireless networks, satellite networks and some advancement in original TCP mechanisms.

14.4 Nakagami Fading Channel

In this section we derived the outage probability of Nakagami- m fading channel model. In this model we assume that γ_0 is the required signal-to-noise threshold for decoding the received packet at a node j . The arising of outage event is subject to the condition, where $\gamma(i) < \gamma_0$ within a coherent time of node j . In our model, when node i sends a packet to node j , the outage probability is represented by p_{out_i} as shown in Eq. 14.2.

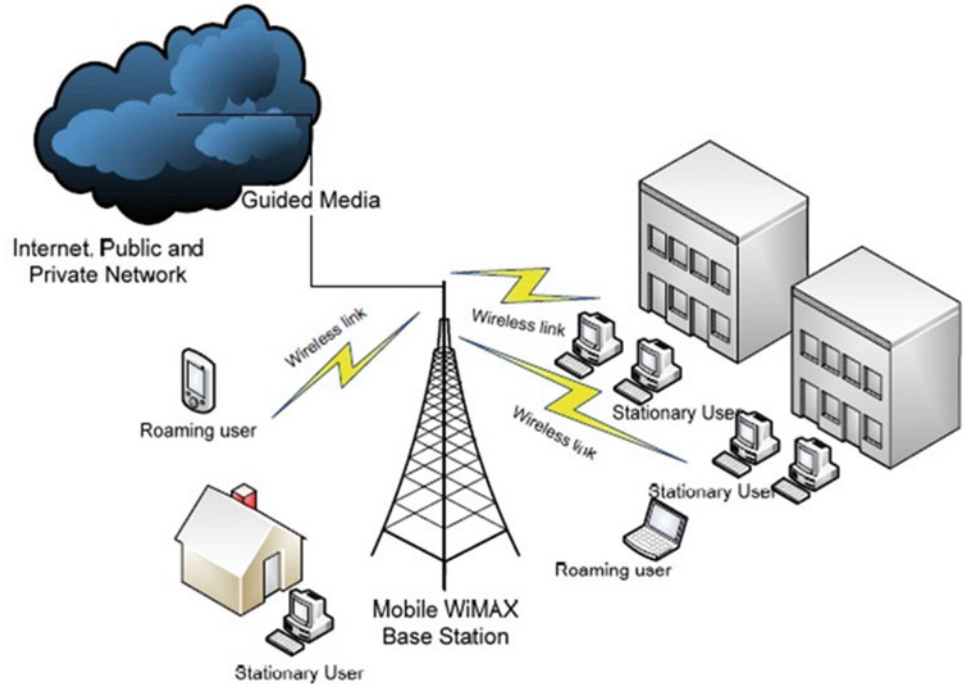
The Probability Density Function (PDF) of signal-to-noise for Nakagami- m fading channel is given in Eq. 14.1 below,

$$f_\gamma = \frac{1}{\Gamma(m)} \left(\frac{m}{\bar{\gamma}(i)} \right)^m \gamma^{m-1} \exp\left(-\frac{m\gamma}{\bar{\gamma}(i)}\right). \quad (14.1)$$

where m is the shape parameter of the distribution and $\Gamma(\cdot)$ is the gamma function.

$$\begin{aligned}
p_{out_i} &= P_r(\gamma < \gamma_0) \\
&= \int_0^{\gamma_0} \frac{1}{\Gamma(m)} \left(\frac{m}{\bar{\gamma}(i)} \right)^m \gamma^{m-1} \exp\left(-\frac{m\gamma}{\bar{\gamma}(i)}\right) d\gamma \\
p_{out_i} &= 1 - \exp\left(-\frac{m\gamma_0}{\bar{\gamma}(i)}\right) \times \sum_{s=0}^{m-1} \frac{\left(\frac{m\gamma_0}{\bar{\gamma}(i)}\right)^s}{s!} \\
p_{out_i} &= 1 - \exp\left(-\frac{m\gamma_0 N_0 (4\pi)^2 d_i^l}{G_T G_R \lambda^2 P_t}\right) \\
&\quad \times \left(\sum_{s=0}^{m-1} \frac{\left(-\frac{m\gamma_0 N_0 (4\pi)^2 d_i^l}{G_T G_R \lambda^2 P_t}\right)^s}{s!} \right). \quad (14.2)
\end{aligned}$$

Fig. 14.1 Mobile Wi-Max system



Since packet is divided into M blocks hence, packet error probability will be shown in Eq. 14.3,

$$p_{PHY_i} = 1 - (1 - p_{out_i})^M. \quad (14.3)$$

$$p_{PHY_i} = 1 - \exp\left(-\frac{m\gamma_0 N_0 (4\pi)^2 d_i^l}{G_T G_R \lambda^2 P_t}\right) \times \left(\sum_{s=0}^{m-1} \frac{\left(\frac{m\gamma_0 N_0 (4\pi)^2 d_i^l}{G_T G_R \lambda^2 P_t}\right)^s}{s!}\right)^M. \quad (14.4)$$

14.5 Mobile Wi-Max Technology and MPEG-4 Video Traffic

Worldwide Interoperability for Microwave Access (Wi-Max) is wireless metropolitan area network technology. The de facto standard for Wi-Max broadband technology is the IEEE 802.16e (Wi-Max). It offers mobile and fixed broadband networks. It is standardized version of Ethernet intends to provide compatibility and interoperability in broadband wireless access technologies. Theoretically it can transfer data with a speed of 64 Mbps down-link and 28 Mbps up-link on single channel. For mobile networks its bandwidth capacity is 15 Mbps over a range of 3 Km. Wi-Max can operate in different frequency spectrum such as 2.3 GHz, 2.5 GHz, 3.5 GHz, and 5.8 GHz. A mobile

Wi-Max system is depicted in Fig. 14.1. The characteristics that make mobile Wi-Max better than Wi-Fi and other wireless technologies are; high bandwidth, quality of service, scalability, freedom from wires, and security. In the context of mobile Wi-Max mobile stations are connected to a static base station via air interface. Beside connectivity services the base station provides quality of service policies, traffic and radio resource management to mobile stations.

Moving Picture Expert Group version 4 (MPEG-4) is a compression standard for multimedia compression and has reduce bit rate i.e. 4.8 kbps to 64 kbps, with good quality, is a good choice for video streaming over network. It consists of Intra-coded frame, Predicted frame, and Bidirectional frame, where every frame has its own characteristics. We have used MPEG-4 traffic at the application layer and have taken from [27]. The basic object in MPEG-4 is video object plane, a grouping of I frames, P frames, and B frames [26]. MPEG-4 has higher resolution ranging from 176×144 to 1920×1080 [28]. MPEG-4 is video must be compressed before transferring it over a network [29].

14.6 Simulations

This section discusses only one simulation scenario, performance matrices, and results. We have used network simulator version 2 (NS-2) for simulation with assumption of error free communication and perfect radio conditions. This performance of MCTCP, DCCP, TCP, SCTP, UDP, and RTP have been assessed for video steaming over Wi-Max

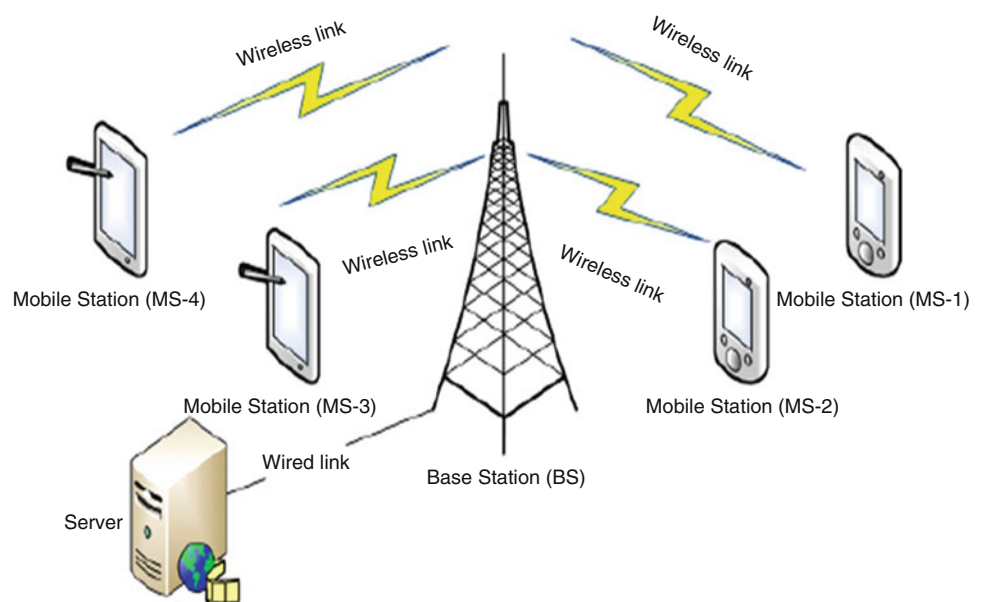
network shown in Fig. 14.1, and performed extensive experiments on it and the mean results are shown in the paper. Table 14.2 shows list of parameters used in the simulation. The performance matrices used are throughput, packet loss, delay, and jitter. A scenario shown in Fig. 14.2, with multiple mobile stations is simulated for assessing performance of given protocols. The performance matrices are calculated as follow;

- Throughput: It is the number of bits successfully transferred in a particular period of time, i.e., total amount of data received/ total time.
- Packet Loss: It is the number of packets sent but not received correctly, and is calculated as; total number of packet transmitted – total number of packet received.

Table 14.2 The simulation conditions

Parameters	Values
Modulation Schemes	64 QAM
Antenna type	Omni direction
Radio Propagation	Two-ray ground
Queue type	Drop-tail
MAC protocol	IEEE802.11
Routing protocol	AODV
Offered Load	1, 2, 1 3,,12 Mbps
Maximum Queue length	100[packet]
Distance between stations	200[m]
Packet size	1024[Byte]
Number of Mobile Stations	8
Speed of Mobile Stations	3Km/h and 5Km/h
Number of Servers	1
Simulation time	50[s]

Fig. 14.2 Multiple mobile stations downloading video



- Delay: It is the difference in time for a packet being transmitted and received, i.e., Packet A's receiving time – Packet A's sending time.
- Jitter: It is the jerk time between two consecutive delays, i.e., Delay(A) – Delay (B).

14.6.1 Multiple Mobile Stations Scenario

The simulation results are based on multiple mobile stations that are connected to the server through base station as depicted in Fig. 14.1. Same simulation parameters are used with different speed of the mobile stations i.e. 3 Km/h and 5 Km/h. The scenario is depicted in Fig. 14.2.

14.6.1.1 Mobile Nodes at a Speed of 3 Km/h

In this scenario video traffic is multi-cast towards the mobile stations and the transport protocol performance with growing number of mobile stations is analyzed. The mobile nodes are moving at a speed of 3 Km/h. The throughput of the network is depicted in Figs. 14.3 and 14.7, where the x-axis shows the number of mobile stations and y-axis represents the throughput in Mbps. It is evident from Figs. 14.3 and 14.7 that the throughput is continuously increasing when the number of mobile nodes is increasing. MPTCP gives us better results in term of throughput because it makes use of multiple paths to avoid congestion and load on a particular node in the network.

Similarly, in Figs. 14.4 and 14.8 the number of packets lost is shown on x-axis with mobile stations on y-axis. The better result in terms of packet lost is given by MPTCP, TCP and DCCP. For MPTCP less packet drops has a reason

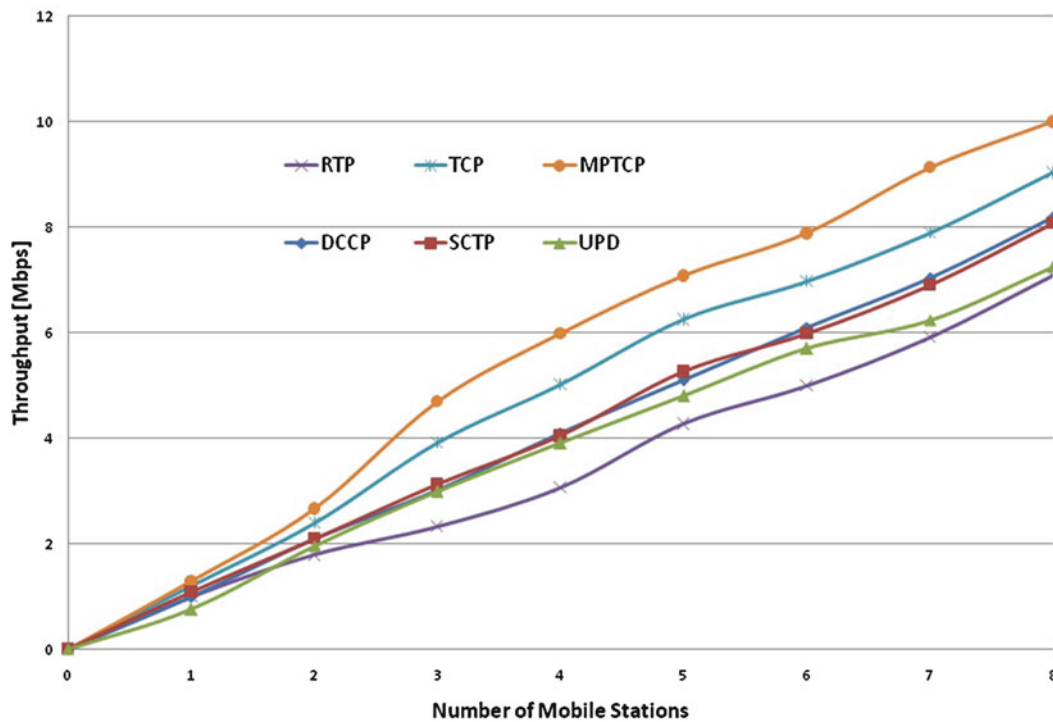


Fig. 14.3 Throughput at a speed of 3 Km/h

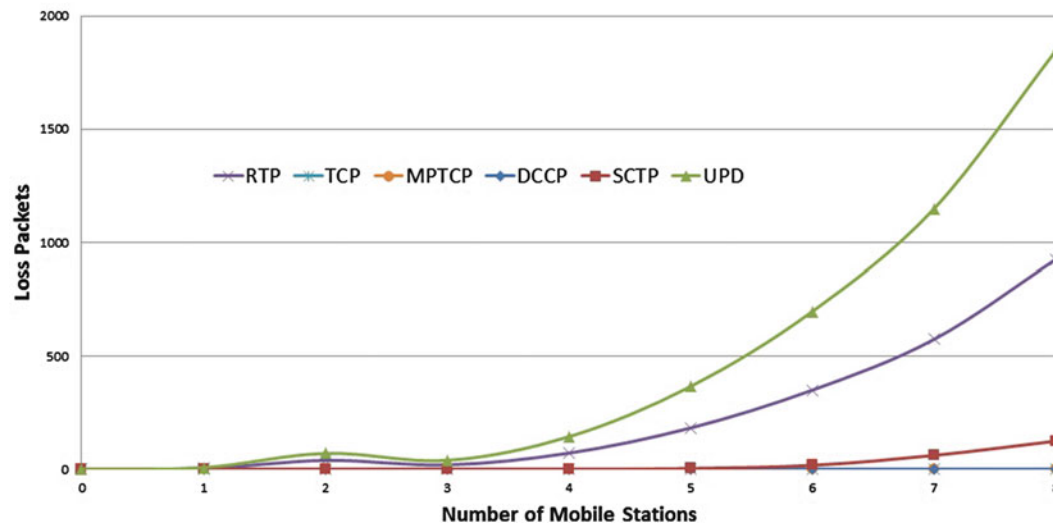


Fig. 14.4 Packet lost at a speed of 3 Km/h

discussed above, whereas TCP and DCCP make connections and have congestion control mechanism with and without reliability and that's why drop less packets compare to other protocols. On the other hand UDP have no congestion con-

control mechanism and due to that it drops highest number of packets. In Figs. 14.5 and 14.9, delay of the network is drawn and MPTCP outperforms all other protocols. In Figs. 14.6 and 14.10, Jitter of the networks is shown where

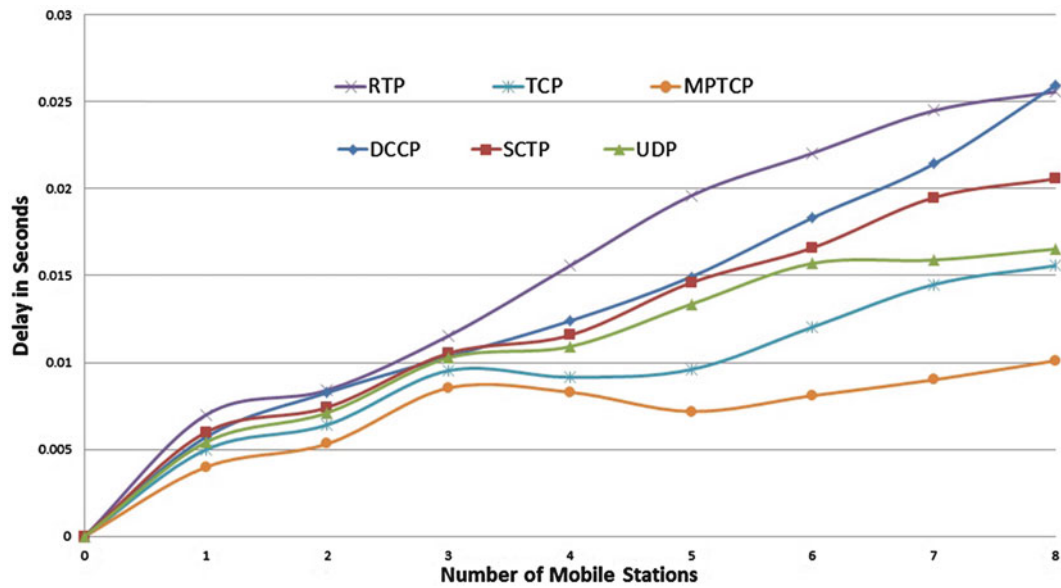


Fig. 14.5 Delay at a speed of 3 Km/h

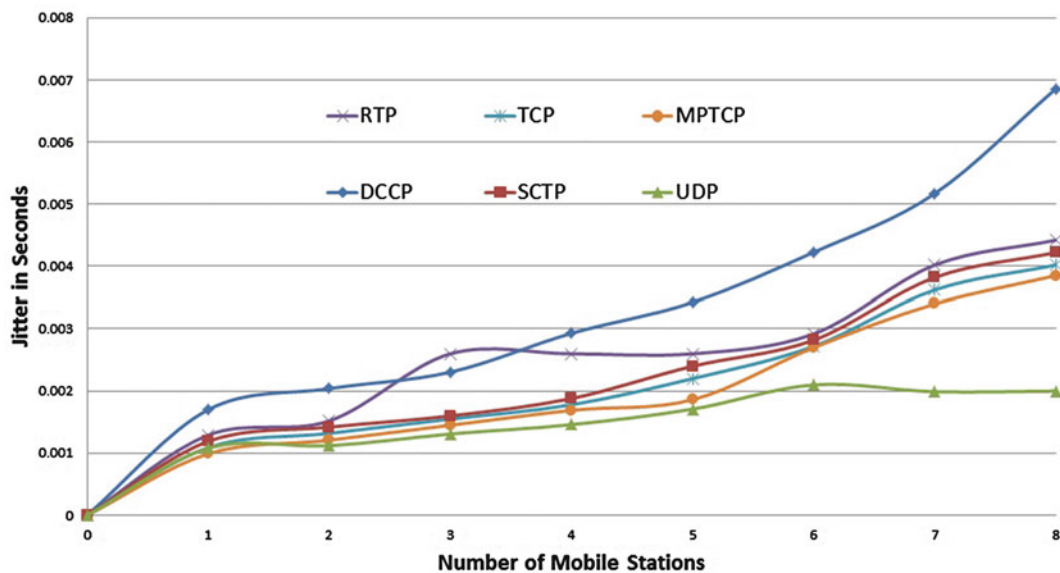


Fig. 14.6 Jitter at a speed of 3 Km/h

UDP gives better results compare to other protocols because the packets arrival rate is constant and continuous.

14.6.1.2 Mobile Nodes at a Speed of 3Km/h

In this scenario video traffic is multi-cast towards the mobile stations and the transport protocol performance with growing number of mobile stations is analyzed. The mobile stations are moving at a speed of five meters per second (Figs. 14.7, 14.8, 14.9, and 14.10).

14.7 Conclusion

In this paper analysis of different transport protocols has been performed by using video traffic under Nakagami channel over mobile Wi-Max network. The results show that MPTCP outperforms all other scheme in terms of throughput, packet loss, jitter, and delay. Similar DCCP and SCTP works better that TCP and UDP in packet loss and throughput but

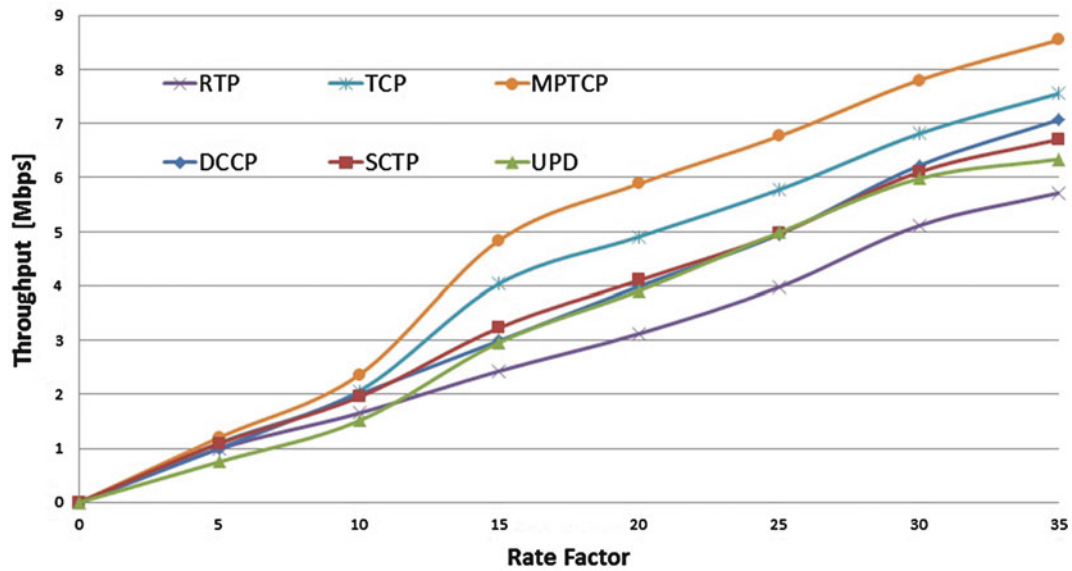


Fig. 14.7 Throughput at a speed of 5 Km/h

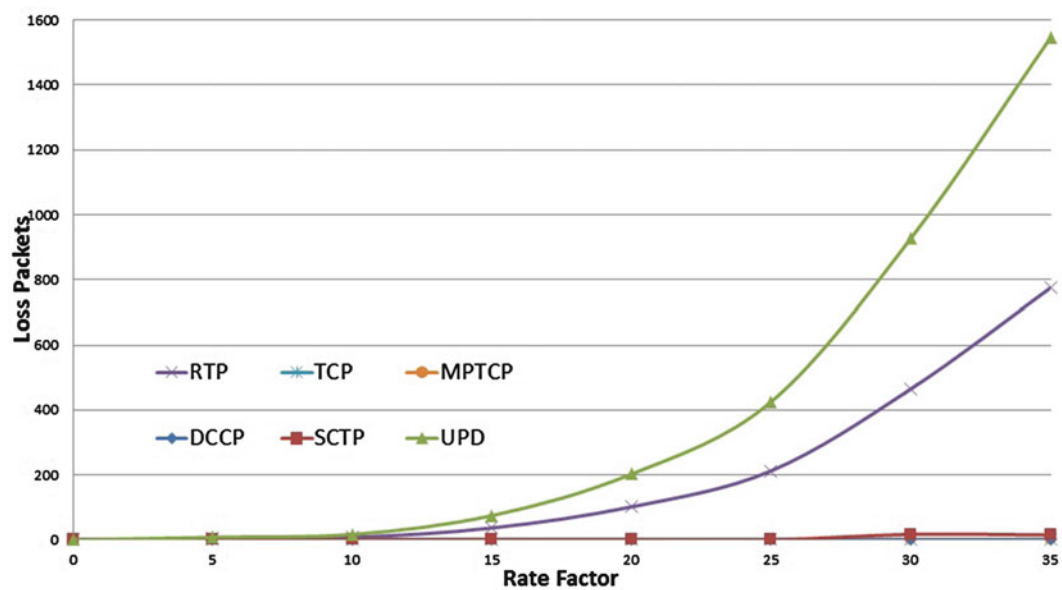


Fig. 14.8 Packet Lost at a speed of 5 Km/h

DCCP and SCTP have more jitter and delay due to which they are not suitable for video traffic. RTP and UDP have higher packet loss which badly affects MPEG transmission. In near future we will extend this research work by considering multiple cells and will analyze transport protocols for transferring voice over Internet protocol and MPEG traffic on Wi-Max and other wireless technologies under different

fading channels, such as Suzuki, Weibull, and Rician. Moreover, the protocols will be analyzed on various complex scenarios and a hybrid approach will be proposed. In short we conclude that after analyzing these six protocols MPTCP provides better QoS for transmission of video traffic over IEEE 802.16e under Nakagami channel, due to the congestion control mechanism it provides.

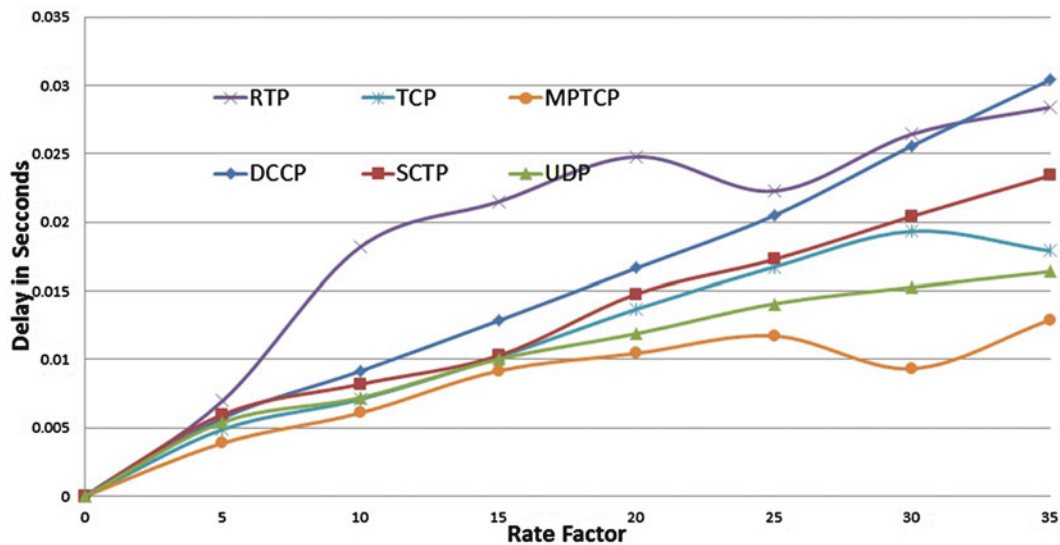


Fig. 14.9 Delay at a speed of 5 Km/h

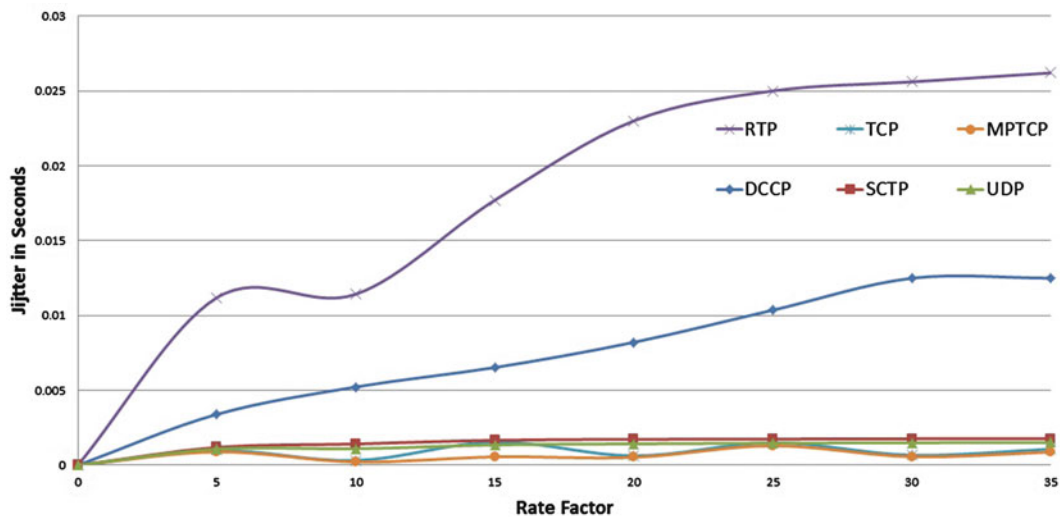


Fig. 14.10 Jitter at a speed of 5 Km/h

References

- Postel, J. (1981). Transmission control protocol.
- Postel, J. (1980). User datagram protocol (No. RFC 768).
- Schulzrinne, H., Casner, S., Frederick, R., & Jacobson, V. (2003). RTP: A transport protocol for real-time applications (No. RFC 3550).
- Raiciu, C., Handley, M., & Wischik, D. (2011). Coupled congestion control for multi-path transport protocols (No. RFC 6356).
- Stewart, R. (2007). Stream control transmission protocol (No. RFC 4960).
- Floyd, S., Handley, M., & Kohler, E. (2006). Datagram congestion control protocol (DCCP).
- IEEE 802.16e/D5-2004, Part 16: Air interface for fixed and mobile broadband wireless access systems – amendment for physical and medium access control layers for combined fixed and mobile operation in licensed bands, Nov. 2004.
- Loo, J., Mauri, J. L., & Ortiz, J. H. (Eds.). (2016). Mobile ad hoc networks: current status and future trends. CRC Press.
- Khan, F., ur Rehman, A., Arif, M., Aftab, M., & Jadoon, B. K. (2016, April). A survey of communication technologies for smart grid connectivity. In *2016 International Conference on Computing, Electronic and Electrical Engineering (ICE Cube)* (pp. 256–261). IEEE.
- Kaur, G., & Kaur, N. (2016). Dissect the performance of network layer protocols for video streaming over Worldwide Interoperability of Microwaves Access WiMAX). *International Journal of Applied Engineering Research*, 11(10), 7196–7199.
- Apostolaras, A., Iosifidis, G., Chounos, K., Korakis, T., & Tassioulas, L. (2016). A mechanism for mobile data offloading to wireless mesh networks. *IEEE Transactions on Wireless Communications*, 15(9), 5984–5997.
- Amin, R., & Martin, J. (2016). Assessing performance gains through global resource control of heterogeneous wireless

- networks. *IEEE Transactions on Mobile Computing*, 15(2), 292–305.
13. Abdullah, S. A., & Abduljabbar, R. B. (2014). Streaming video content over NGA (next generation access) network technology. *International Journal of Science, Engineering and Computer Technology*, 4(3), 56.
 14. Sarkar, S., Misra, S., Bandyopadhyay, B., Chakraborty, C., & Obaidat, M. S. (2015). Performance analysis of IEEE 802.15.6 MAC protocol under non-ideal channel conditions and saturated traffic regime. *IEEE Transactions on Computers*, 64(10), 2912–2925.
 15. Abdullah, N., & Singh, H. (2014). Performance based comparison of different routing protocol under different modulation over WiMAX.
 16. Kamil, W. A. (2015). Performance evaluation of TCP, UDP and DCCP for video traffics over 4G network (Doctoral dissertation, Universiti Utara Malaysia).
 17. Yunus, F., Ariffin, S. H., Syed-Yusof, S. K., Ismail, N. S. N., & Fisal, N. (2014). Transport protocol performance for multi-hop transmission in wireless sensor network (WSN). In *Handbook of research on progressive trends in wireless communications and networking* (pp. 389–409). Hershey: IGI Global.
 18. Othman, H. R., Ali, D. M., Yusof, N. A. M., Noh, K. S. S. K. M., & Idris, A. (2014, August). Performance analysis of VoIP over mobile WiMAX (IEEE 802.16 e) best-effort class. In *2014 I.E. 5th Control and System Graduate Research Colloquium (ICSGRC)* (pp. 130–135). Piscataway: IEEE.
 19. C. Mukasa., Aalo, V. A., & Efthymoglou, G. (2016, October). On the performance of a dual-hop network with a mobile relay in a Nakagami fading environment. In *2016 I.E. 21st International Workshop on Computer Aided Modelling and Design of Communication Links and Networks (CAMAD)*, Toronto, ON (pp. 43–47). IEEE.
 20. Khan, F., ur Rahman, I., Khan, M., Iqbal, N., & Mujahid, A. (2016, September). CoAP-based request-response interaction model for the Internet of things. In *International Conference on Future Intelligent Vehicular Technologies* (pp. 146–156). Cham: Springer.
 21. Fida, N., Khan, F., Jan, M. A., & Khan, Z. (2016, September). Performance Analysis of Vehicular Adhoc Network Using Different Highway Traffic Scenarios in Cloud Computing. In *International Conference on Future Intelligent Vehicular Technologies* (pp. 157–166). Springer, Cham.
 22. Younas, N., Asghar, Z., Qayyum, M., & Khan, F. (2016, September). Education and socio economic factors impact on earning for Pakistan-A bigdata analysis. In *International Conference on Future Intelligent Vehicular Technologies* (pp. 215–223). Cham: Springer.
 23. Khan, F., Khan, M., Iqbal, Z., ur Rahman, I., & Mujahid, A. (2016, September). Secure and safe surveillance system using sensors networks-internet of things. In *International Conference on Future Intelligent Vehicular Technologies* (pp. 167–174). Cham: Springer.
 24. Yunus, F., Ariffin, S. H., Syed-Yusof, S. K., Ismail, N. S. N., & Fisal, N. (2014). Transport protocol performance for multi-hop transmission in wireless sensor network (WSN). In: M. A. Matin (Ed.), *Handbook of research on progressive trends in wireless communications and networking* (Vol. 389). Hershey, PA: IGI Global.
 25. Bhebe, L. (2016). Service Continuity in 3GPP Mobile Networks.
 26. Ahmad, S., & Khan, S. A. (2016). Performance evaluation of TCP and DCCP protocols in mobile Ad-Hoc networks (MANETS). *VFAST Transactions on Software Engineering*, 10(1)
 27. Tanwir, S., & Perros, H. (2013). A survey of VBR video traffic models. *IEEE Communications Surveys & Tutorials*, 15(4), 1778–1802.
 28. <http://www.apple.com/quicktime/technologies/h264/>
 29. <http://www.pcguides.com/ref/video/modesColor-c.html>