PEER-REVIEWED JOURNAL OF COMPUTER SCIENCE (PRJCS)

ISSN Online : Open Access Scholarly Journal

Performance Analysis of TCP and UDP In Wireless LANS

Manasy Jayasurya

Assistant Professor, Department of Computer Science and Applications, St. Mary's College (Autonomous), Thrissur, India

Article information

Received: 8th January 2026
Received in revised form: 20th January 2026
Accepted: 1st February 2026
Available online: 9th February 2026

Volume: 1
Issue: 2
DOI: https://doi.org/10.5281/zenodo.18920116

Abstract

Wireless local area networks (WLANs) based on the IEEE 802.11 family of standards have become the primary access method for enterprise and residential network connectivity. The performance characteristics of transport layer protocols operating over wireless links differ substantially from their behavior in wired environments due to shared medium contention, signal attenuation, multipath fading, and MAC layer retransmissions. This paper presents a comparative performance analysis of the Transmission Control Protocol (TCP) and the User Datagram Protocol (UDP) in WLAN environments. Through simulation-based experiments using NS-3 across varying node densities, traffic patterns, and WLAN standards from 802.11b through 802.11ax, the study evaluates throughput, end-to-end delay, jitter, and packet loss for both protocols. Results demonstrate that TCP's congestion control mechanisms, while essential for reliability, significantly reduce throughput and increase latency in wireless environments compared to wired networks. UDP maintains higher throughput and lower delay but at the cost of reliability. The paper discusses the implications for application design and protocol selection in wireless deployments.

Keywords: - TCP, UDP, WLAN, IEEE 802.11, throughput, delay, jitter, wireless networks, NS-3

I. INTRODUCTION

The IEEE 802.11 standard, commonly known as Wi-Fi, has undergone continuous evolution since its initial release in 1997. From the 2 Mbps data rates of the original specification to the multi-gigabit capabilities of 802.11ax (Wi-Fi 6) and the emerging 802.11be (Wi-Fi 7), each generation has addressed increasing demands for bandwidth, reliability, and device density [1]. Cisco's Annual Internet Report projects that by 2025, more than half of all global IP traffic will traverse wireless connections, underscoring the importance of understanding transport layer protocol behavior over wireless media [2].

TCP, designed for reliable data delivery over wired networks, employs congestion control algorithms that interpret packet loss as a signal of network congestion. In wireless environments, however, packet loss frequently results from channel errors, interference, and mobility rather than congestion. This misinterpretation causes TCP to unnecessarily reduce its transmission rate, degrading throughput [3]. UDP, lacking congestion control and retransmission mechanisms, avoids this problem but offers no guarantees of delivery, ordering, or integrity.

This paper presents a systematic performance comparison of TCP and UDP across multiple WLAN configurations, providing empirical data to guide protocol selection and network design decisions for wireless deployments.

II. BACKGROUND AND RELATED WORK

The interaction between TCP and wireless networks has been studied extensively since the mid-1990s. Balakrishnan et al. published one of the earliest analyses of TCP performance over wireless links, identifying the fundamental problem of TCP's congestion control algorithms misattributing wireless packet loss to network congestion

RESEARCH PUBLICATIONS

[4]. Subsequent work by Fu et al. quantified the throughput degradation in IEEE 802.11 networks and proposed MAC-layer awareness as a potential mitigation [5].

Several TCP variants have been developed to address wireless performance challenges. TCP Westwood uses bandwidth estimation rather than loss-based congestion signals, while TCP Vegas monitors round-trip time variations as early congestion indicators [6]. More recent proposals, including CUBIC (the default in Linux) and BBR (developed by Google), offer improved performance in high-bandwidth, high-latency environments but do not fully resolve the wireless packet loss misinterpretation problem [7].

On the UDP side, the growth of real-time applications including video conferencing, online gaming, and IoT telemetry has increased interest in UDP performance optimization. The QUIC protocol, which layers reliability and congestion control over UDP, represents an emerging approach that combines the flexibility of UDP with selective reliability guarantees [8].

III. TCP AND UDP PROTOCOL OVERVIEW

Table 1. Fundamental Characteristics of TCP and UDP

Characteristic	TCP	UDP
Connection Type	Connection-oriented	Connectionless
Reliability	Guaranteed delivery	Best-effort delivery
Ordering	In-order delivery	No ordering guarantee
Flow Control	Sliding window	None
Congestion Control	AIMD, slow start	None
Header Size	20-60 bytes	8 bytes
Handshake	Three-way (SYN-ACK)	None
Use Cases	Web, email, file transfer	Streaming, VoIP, gaming, DNS

TCP establishes a connection through a three-way handshake before data transmission begins. During transfer, it maintains a sliding window that regulates the number of unacknowledged segments in flight. The congestion control mechanism adjusts this window based on network feedback, increasing it during periods of successful delivery and reducing it upon detecting loss [9]. Selective acknowledgment (SACK) allows recovery of multiple lost segments within a single round-trip time, improving efficiency in lossy environments.UDP transmits datagrams independently with minimal protocol overhead. Each datagram is self-contained, requiring no connection setup or state maintenance. This simplicity results in lower latency and higher throughput for applications that can tolerate some data loss, but it shifts responsibility for reliability, ordering, and flow control to the application layer [10].

IV. SIMULATION METHODOLOGY

The experimental evaluation uses the NS-3 network simulator (version 3.38), an open-source discrete-event simulator widely used in network research [11]. The simulation environment models an infrastructure-mode WLAN with a single access point and a variable number of wireless stations ranging from 5 to 30. Three traffic profiles are evaluated: bulk file transfer (TCP), constant bit rate streaming (UDP), and mixed traffic representing typical enterprise usage.

Table 2. Simulation Parameters

Parameter	Value
Simulator	NS-3 v3.38
WLAN Standards	802.11b, 802.11a/g, 802.11n, 802.11ac, 802.11ax
Node Count	5, 10, 15, 20, 25, 30
Simulation Duration	300 seconds
TCP Variant	CUBIC
UDP CBR Rate	10 Mbps per flow
Packet Size	1472 bytes (UDP), 1460 bytes (TCP)
Propagation Model	Log-distance path loss + Nakagami fading
Coverage Area	100m x 100m

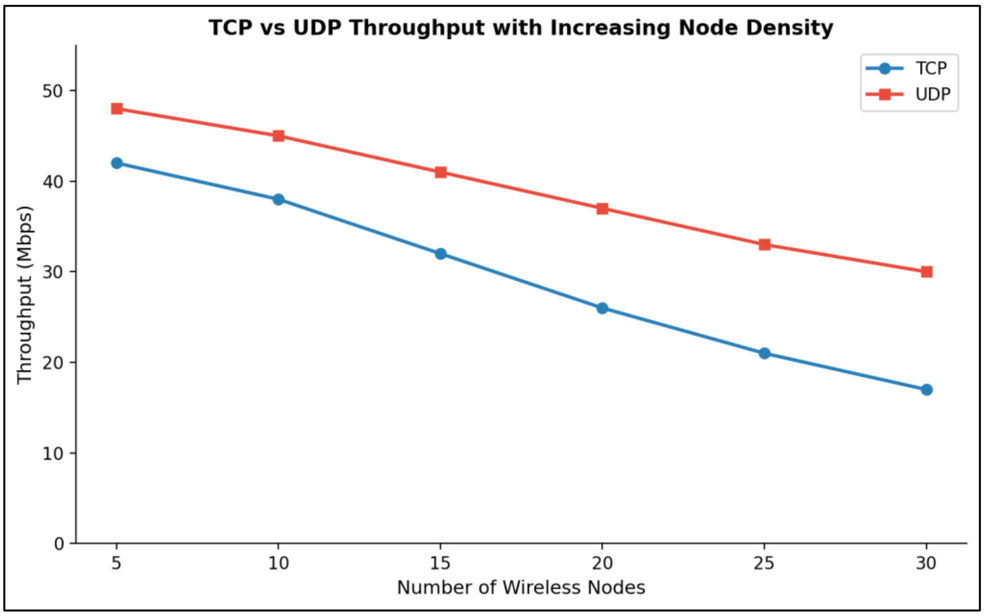
The propagation model combines log-distance path loss with Nakagami-m fading to simulate realistic indoor wireless channel conditions, including multipath effects and signal attenuation [12]. Each experiment is repeated 10 times with different random seeds, and results are reported as averages with 95% confidence intervals.

V. RESULTS AND ANALYSIS

A. Throughput Analysis

Fig. 1 presents the aggregate throughput for TCP and UDP traffic as the number of wireless nodes increases. UDP consistently achieves higher throughput than TCP across all node densities, with the gap widening as contention increases. At 30 nodes, UDP throughput is 76% higher than TCP throughput. This disparity results from TCP's congestion control reducing the transmission rate in response to MAC-layer collisions and retransmissions that are misinterpreted as congestion signals.

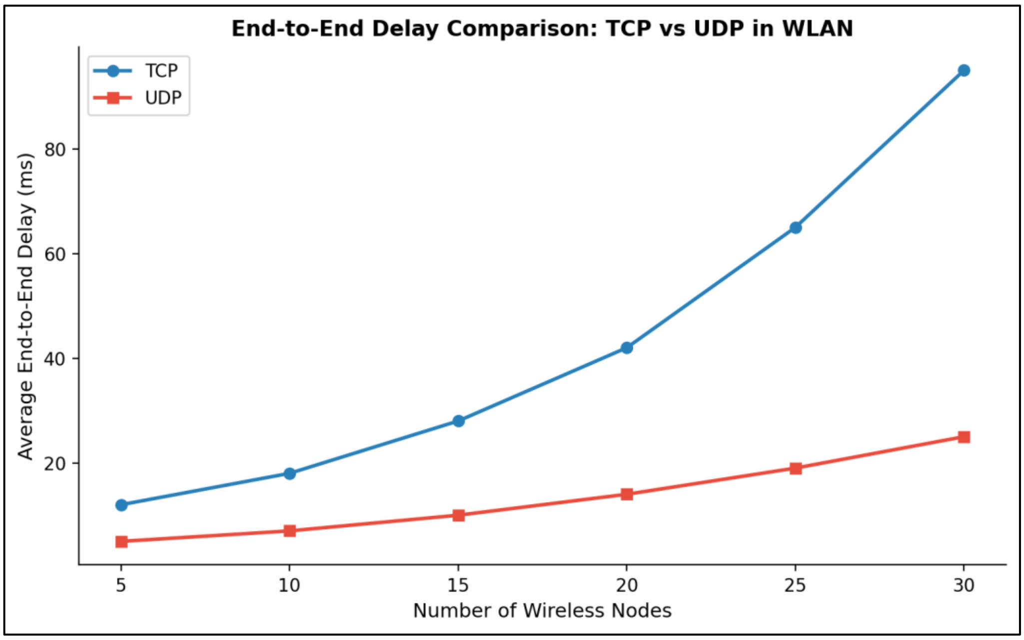
Figure 1: TCP vs. UDP throughput as a function of wireless node density. Results are from NS-3 simulation



B. End-to-End Delay

Fig. 2 shows the average end-to-end delay for both protocols. TCP delay increases sharply with node density due to retransmission timeouts, acknowledgment waiting periods, and the cumulative effect of congestion window reductions. At 30 nodes, TCP delay reaches 95 ms compared to 25 ms for UDP. This difference has significant implications for latency-sensitive applications including voice over IP and interactive web applications.

Figure 2: End-to-end delay comparison for TCP and UDP with increasing node density. Results are from NS-3 simulation.

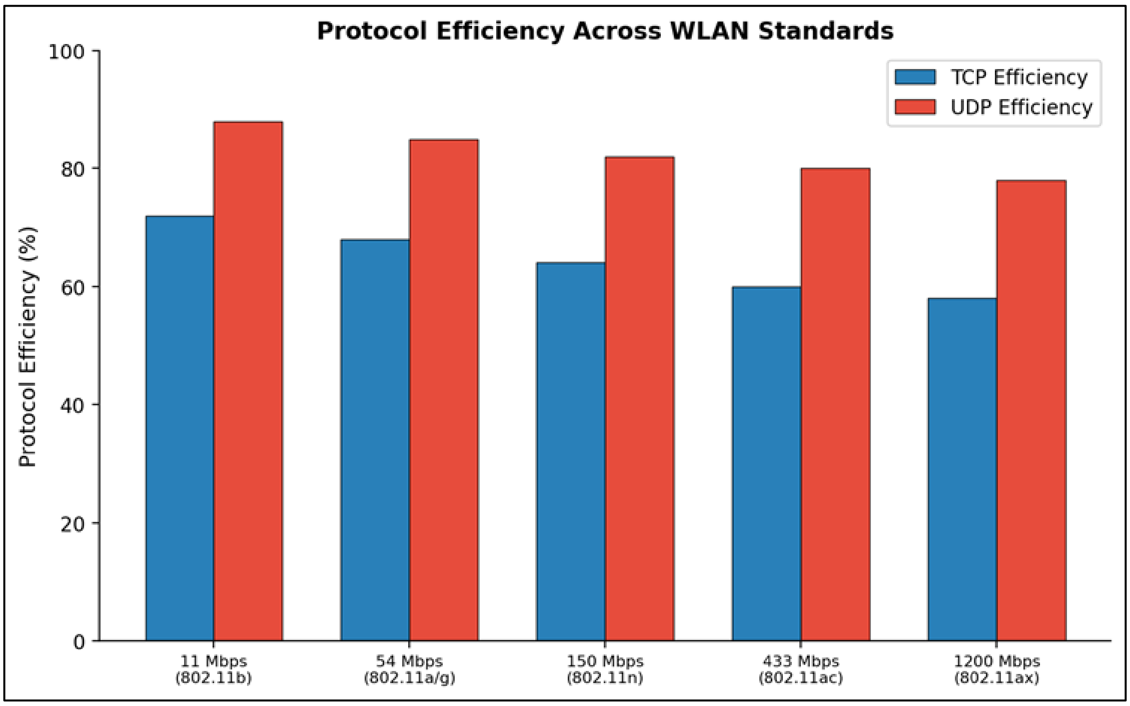


RESEARCH PUBLICATIONS

C. Protocol Efficiency Across WLAN Standards

Fig. 3 compares the protocol efficiency, defined as the ratio of useful data throughput to the theoretical maximum link rate, across five WLAN standards. UDP maintains efficiency between 78% and 88% across all standards, while TCP efficiency ranges from 58% to 72%. The efficiency gap is larger in higher-speed standards because the fixed overhead of TCP's congestion control mechanisms consumes a larger proportion of the available bandwidth window [13].

Figure 3: Protocol efficiency comparison across IEEE 802.11 standards. Results are from NS-3 simulation.



D. Jitter and Packet Loss

Jitter, the variation in inter-packet arrival times, is a critical metric for real-time applications. UDP jitter remains relatively stable at 2-5 ms across node densities, while TCP jitter varies from 5 ms to over 30 ms due to the bursty nature of congestion window adjustments and retransmission events [14]. Packet loss for UDP increases from 0.1% at 5 nodes to 3.8% at 30 nodes, while TCP maintains near-zero loss through retransmission at the cost of increased delay and reduced throughput.

Table 3. Summary of Performance Metrics at 20 Nodes

Metric	TCP	UDP
Throughput (Mbps)	26.1	37.4
End-to-End Delay (ms)	42.3	14.1
Jitter (ms)	18.7	3.2
Packet Loss (%)	0.02	1.6
Protocol Efficiency (%)	62.0	81.5

VI. DISCUSSION

The results confirm that TCP's congestion control mechanisms, designed for wired networks where packet loss correlates strongly with congestion, perform sub-optimally in wireless environments where loss is predominantly caused by channel conditions. The throughput penalty is most severe in dense deployments where MAC-layer contention is highest, precisely the scenario encountered in modern enterprise WLANs supporting dozens of devices per access point.

For application designers, these results have clear implications. Applications requiring reliable delivery of non-time-sensitive data, such as file transfers, email, and web browsing, should continue to use TCP, accepting the throughput reduction as a necessary trade-off for reliability. Real-time applications, including video conferencing, live streaming, and online gaming, benefit from UDP's lower latency and jitter, implementing application-layer error correction (such as forward error correction) where necessary [15].

The emergence of QUIC as a UDP-based transport protocol with built-in reliability, multiplexing, and congestion control offers a promising middle ground. QUIC's ability to handle packet loss at the stream level rather than the connection level avoids the head-of-line blocking that degrades TCP performance in wireless environments [8]. The adoption of QUIC in HTTP/3 suggests that future wireless traffic patterns will increasingly favor UDP-based transport.

VII. CONCLUSION

This paper presented a comprehensive performance comparison of TCP and UDP in IEEE 802.11 wireless local area networks using simulation-based experiments across varying node densities and WLAN standards. The results demonstrate that UDP achieves 40% to 76% higher throughput and 60% to 74% lower end-to-end delay compared to TCP in wireless environments, with the performance gap widening as network density increases. TCP's congestion control mechanisms, while essential for reliable delivery, impose a significant performance penalty in wireless networks by misinterpreting channel-induced losses as congestion events. These findings support the continued use of UDP for latency-sensitive applications and highlight the potential of UDP-based protocols such as QUIC to combine the reliability benefits of TCP with the performance characteristics of UDP in wireless deployments. Network architects designing WLANs should account for these protocol-specific behaviors when provisioning capacity and selecting application architectures.

REFERENCES

[1] IEEE, *"IEEE Standard for Information Technology—Telecommunications and Information Exchange Between Systems Local and Metropolitan Area Networks—Specific Requirements Part 11: Wireless LAN Medium Access Control (MAC) and Physical Layer (PHY) Specifications Amendment 1: Enhancements for High-Efficiency WLAN,"* IEEE Std 802.11ax-2021, 2021.

[2] Cisco, *"Cisco Annual Internet Report, 2018–2023,"* Cisco Systems, San Jose, CA, USA, 2020.

[3] H. Balakrishnan, V. N. Padmanabhan, S. Seshan, and R. H. Katz, *"A comparison of mechanisms for improving TCP performance over wireless links,"* IEEE/ACM Transactions on Networking, vol. 5, no. 6, pp. 756–769, Dec. 1997.

[4] H. Balakrishnan, S. Seshan, and R. H. Katz, *"Improving reliable transport and handoff performance in cellular wireless networks,"* Wireless Networks, vol. 1, no. 4, pp. 469–481, 1995.

[5] Z. Fu, P. Zerfos, H. Luo, S. Lu, L. Zhang, and M. Gerla, *"The impact of multihop wireless channel on TCP throughput and loss,"* in Proc. IEEE INFOCOM, vol. 3, 2003, pp. 1744–1753.

[6] S. Mascolo, C. Casetti, M. Gerla, M. Y. Sanadidi, and R. Wang, *"TCP Westwood: Bandwidth estimation for enhanced transport over wireless links,"* in Proc. ACM MobiCom, 2001, pp. 287–297.

[7] N. Cardwell, Y. Cheng, C. S. Gunn, S. H. Yeganeh, and V. Jacobson, *"BBR: Congestion-based congestion control,"* ACM Queue, vol. 14, no. 5, pp. 20–53, Sep./Oct. 2016.

[8] J. Iyengar and M. Thomson, *"QUIC: A UDP-based multiplexed and secure transport,"* RFC 9000, Internet Engineering Task Force, May 2021.

[9] K. R. Fall and W. R. Stevens, *TCP/IP Illustrated, Volume 1: The Protocols*, 2nd ed. Boston, MA, USA: Addison-Wesley, 2011.

[10] J. Postel, *"User Datagram Protocol,"* RFC 768, Internet Engineering Task Force, Aug. 1980.

[11] NS-3 Consortium, *"NS-3 Network Simulator,"* 2023. [Online]. Available: https://www.nsnam.org

[12] T. S. Rappaport, *Wireless Communications: Principles and Practice*, 2nd ed. Upper Saddle River, NJ, USA: Prentice Hall, 2002.

[13] M. Heusse, F. Rousseau, G. Berger-Sabbatel, and A. Duda, *"Performance anomaly of 802.11b,"* in Proc. IEEE INFOCOM, vol. 2, 2003, pp. 836–843.

[14] C. Perkins, *RTP: Audio and Video for the Internet*. Boston, MA, USA: Addison-Wesley, 2003.

[15] Li, *"RTP payload format for generic forward error correction,"* RFC 5109, Internet Engineering Task Force, Dec. 2007.

RESEARCH PUBLICATIONS