Buffering Techniques to Enhance the Multimedia Streaming Quality over Ad Hoc Network

Swetha[1], Shwetha D[2], Devaraju J T[3]

[1] Department of Electronics, Karnataka State Akkamahadevi Women University, Vijayapura, Karnataka, India

[2] Department of Electronics, University College of Science, Tumkur University, Tumakuru, Karnataka, India

[3] Department of Electronic Science, Bangalore University, Bangalore, Karnataka, India

DOI: https://dx.doi.org/10.51584/IJRIAS.2026.11050044

Received: 01 May 2025; Accepted: 06 May 2026; Published: 27 May 2026

ABSTRACT

Multimedia streaming over Ad Hoc networks has recently gained significant attention due to the highly scalable, self-starting, self-managing, self-healing and infrastructure-free nature of these networks. In Ad Hoc networks, data packets are transmitted from the source to the destination through multiple intermediate nodes using distributed multi-hop communication. At each intermediate node, packets experience varying buffering delays before being forwarded to the next hop. Furthermore, intermediate nodes may require a random number of retransmission attempts before successfully forwarding packets. Consequently, some packets may arrive at the destination earlier than preceding packets, some may be dropped, and others may arrive later than subsequent packets, leading to packet reordering at the destination. This issue becomes more critical in multimedia streaming applications based on the User Datagram Protocol (UDP), where stale packets are discarded to maintain real-time delivery. As a result, the effective throughput between the transport layer and the application layer decreases, thereby degrading the quality of multimedia streaming. To address these challenges, this paper proposes two novel buffering algorithms: the Static Reordering Buffer (SRB) and the Improved Reordering Buffer (IRB) algorithms. The performance of the proposed algorithms is evaluated using the QualNet 6.1 network simulator with the objective of improving multimedia streaming quality in Ad Hoc networks.

Keywords: Ad hoc networks, Buffering algorithms, Multimedia streaming, Quality of Service, Simulation

INTRODUCTION

The emergence and widespread use of mobile devices with advanced wireless technologies such as Bluetooth and Wi-Fi are increasingly being used to access the ubiquitous internet and multimedia services. Ad Hoc network technology has become the de facto standard for connecting billions of wireless devices over infrastructure-free network architecture. Over the years, Ad Hoc networks have gained acceptance in various fields such as environmental monitoring, file sharing, wireless printing, healthcare, transportation and military operations because they are low-cost, easy to deploy, scalable, self-starting and self-healing. The proliferation of multimedia streaming over Ad Hoc network is growing exponentially and has initiated novel applications such as wireless surveillance, voice calls, video conferencing, on-demand video streaming, interactive gaming, disaster monitoring and public safety services [1][2][3]. These applications demand high network bandwidth, reliable data transmission, lower end-to-end delay and jitter performances.

AdHoc networks consist of multiple nodes that cooperate with each other to route data to the destination using distributed multi-hop technology. There exist multiple paths for forwarding data from the source to the destination node. Each path can differ in the number of intermediate nodes involved in receiving, buffering, and forwarding packets. Intermediate nodes temporarily store data packets under conditions such as route unavailability, limited network resources, transmission retries, or link failures, until the packets can be successfully forwarded to the next node [4][5]. Therefore, the data packets experience a random end-to-end delay before reaching the destination node [6]. This variance in end-to-end delay and jitter causes some data packets to reach the destination node earlier than the preceding packets, some data packets to reach the destination node later than the succeeding packets, and some of the data packets to be discarded by the network before they reach the destination node, causing the packets to arrive at the destination node out of sequence [7]. As a result, the throughput performance and Quality of Service (QoS) for multimedia applications decrease, which is undesirable for multimedia streaming [8].

To address these issues, this paper proposes two new reordering buffers for the transport layer. In the Static Reordering Buffer (SRB) scheme, received data packets are buffered at the transport layer for a fixed duration before being reordered and forwarded to the application layer. Although packet reordering helps reduce packet loss, the buffering process introduces significant end-to-end delay and jitter for all packets. To overcome this limitation, the Improved Reordering Buffer (IRB) scheme has been proposed. In IRB, only the packets that arrive out of order are temporarily stored in the buffer until proper reordering is achieved, after which they are delivered to the application layer. This approach significantly reduces end-to-end delay and jitter while maintaining effective packet reordering performance. The proposed algorithms are implemented in the QualNet 6.1 network simulator and their performance has been studied and the results are presented in this paper.

Related Work

Multimedia communication over Ad Hoc networks is an increasing trend due to ease of deployment, low cost compared to infrastructure networks. However, Ad Hoc arrangement of nodes require establishment of route dynamically with multiple nodes and out-of-sequence arrival of packets is an increasingly common phenomenon on the Ad Hoc networks. Further out-of-sequence delivery problem may be mitigated using end node solutions and using network-centric solutions which have received increased attention of many researchers and network vendors. Authors in [9] investigate the problem of packet reordering over internet and define a metric called Reorder Buffer-occupancy Density (RBD) for capturing reordering in a stream and to measure resources required to recover from such reordering. In [10], a theoretical foundation for analysis of the impact on packet reordering on sensor network performance is established considering the analytical relationships for variation of Reorder Density (RD) and Reorder Buffer-occupancy Density (RBD) as packet streams pass through re-sequencing buffers. Further the model presented in [10] provides an overall perspective of the impact of re-sequencing buffers and allows for buffer allocation to meet bounds related to packet reordering. In [11], authors use simulation studies to confirm the existence of the stochastic compensation effect to reduce packet reordering with randomized packet scheduling at source into multiple paths of random transmission delays and its considerable influence on the application performance using simulation studies. Authors in [12] demonstrate the catastrophic effect of packet reordering on the performance of the TCP flavours using simulation studies. Authors of paper [13] conclude that parallel processing in the network processor and the allocation scheme for the transmit buffer also adversely impacts packet ordering. Also [14] explores different transmit buffer allocation schemes namely, contiguous,local and global which reduces the packet retransmission to 24%. In [15], authors propose a receiver-initiated duty cycling MAC protocol called reordering-passive MAC (RP-MAC) for wireless sensor networks to achieve high energy efficiency and network throughput with lower end-to-end delay for heavy traffic or low duty cycle. Paper [16] proposes a cross layer approach between network layer and transport layer to identify the dropped and reordered packets for TCP in MANETs. The proposed work in [16] identifies the dropped or reordered packet and the source retransmits only the dropped but not reordered packets and the unnecessary retransmissions are reduced.

Overview of Multimedia Streaming Over Ad Hoc Networks

Multimedia streaming applications demands higher bandwidth and lower packet loss for uninterrupted services. To minimize bandwidth consumption and latency during streaming, raw multimedia data is compressed with codecs before transmission. In particular, Constant Bit Rate (CBR) codecs are used for multimedia compression in Ad Hoc networks because they can generate a suitable target bit rate to maximize the use of limited bandwidth [17]. In addition, most multimedia applications prefer transmission control protocols based on User Datagram Protocol (UDP) because these protocols require only a small channel bandwidth for data transmission compared to Transmission Control Protocols-Internet Protocol (TCP-IP).

Conventional services based on the User Datagram Protocol (UDP) transport mechanism use packet sequence numbers to identify packet freshness, detect duplication, and maintain packet ordering at the destination [18]. At the start of a transmission session, the source assigns a sequence number of zero to the first data packet, and the sequence number is incremented by one for each subsequent packet. Since packets may traverse different paths to reach the destination, variations in end-to-end delay can occur, leading to packet reordering, packet drops, and sequence discrepancies at the receiving end. To handle these discrepancies, the destination node maintains an Expected Packet Sequence (EPS) number, which represents the sequence number of the next expected packet or the sequence number of the last successfully received packet. Upon receiving a data packet, the transport layer compares the EPS value with the sequence number of the received packet. If the EPS value is less than or equal to the received packet's sequence number, the packet is considered valid and delivered to the application layer; otherwise, it is discarded as a duplicate or stale packet [18]. After each successful packet reception, the EPS value is updated to the received packet's sequence number plus one, indicating the next expected packet from the source. This process continues throughout the transmission session. Consequently, even when packets successfully reach the correct destination, they may still be discarded due to sequence mismatches, thereby reducing packet delivery ratio and application-layer throughput in Ad Hoc networks. To overcome these limitations, this paper proposes two buffering algorithms, namely the Static Reordering Buffer (SRB) and Improved Reordering Buffer (IRB) algorithms, with the objective of improving throughput performance and enhancing the Quality of Service (QoS) for multimedia applications in Ad Hoc networks.

Proposed Static Reordering Buffer (SRB) Algorithm

The proposed Static Reordering Buffer (SRB) algorithm uses a priority queue-based buffer at the transport layer to reduce the probability of packet loss due to packet discrepancies by using the following control parameters at the destination node:

- Expected Packet Sequence (EPS): The EPS value is a sequence number of the packet to be received.
- Buffer Retention Timer (BRT): The BRT expires periodically according to a static interval. Shortly after its expiration, the data packets are sent from the transport layer buffer to the application layer.
- Priority Queue: The received data packets are inserted temporarily in the priority queue according to the sequence number of the packet before being sent to the application layer.

In the proposed Static Reordering Buffer (SRB) mechanism, when the first data packet is received at the destination node, the transport layer resets the Expected Packet Sequence (EPS) value to zero, initializes the Buffer Release Timer (BRT) with a fixed time interval, and inserts the received packet into a priority queue based on its sequence number. Subsequently, all received packets having sequence numbers greater than or equal to the EPS value are inserted into the priority queue in the correct order according to their sequence numbers, whereas packets with sequence numbers less than the EPS value are discarded as stale or duplicate packets. This procedure continues until the BRT expires. Upon expiration of the BRT, the reordered packets stored in the priority queue are sequentially forwarded to the application layer. Thereafter, the BRT is reinitialized with the same static time interval, and the EPS value is updated to the sequence number of the last delivered packet plus one, indicating the next expected packet. This process continues until the termination of the transmission session, as illustrated in Figure 1. The proposed SRB algorithm reduces the probability of packet loss by reordering out-of-sequence packets before delivery, thereby improving throughput performance and enhancing the Quality of Service (QoS) for multimedia applications. However, since all received packets remain buffered in the priority queue until the BRT expires, the algorithm may introduce increased queuing delay and latency. To overcome this limitation, an enhanced mechanism called the Improved Reordering Buffer (IRB) algorithm has been proposed.

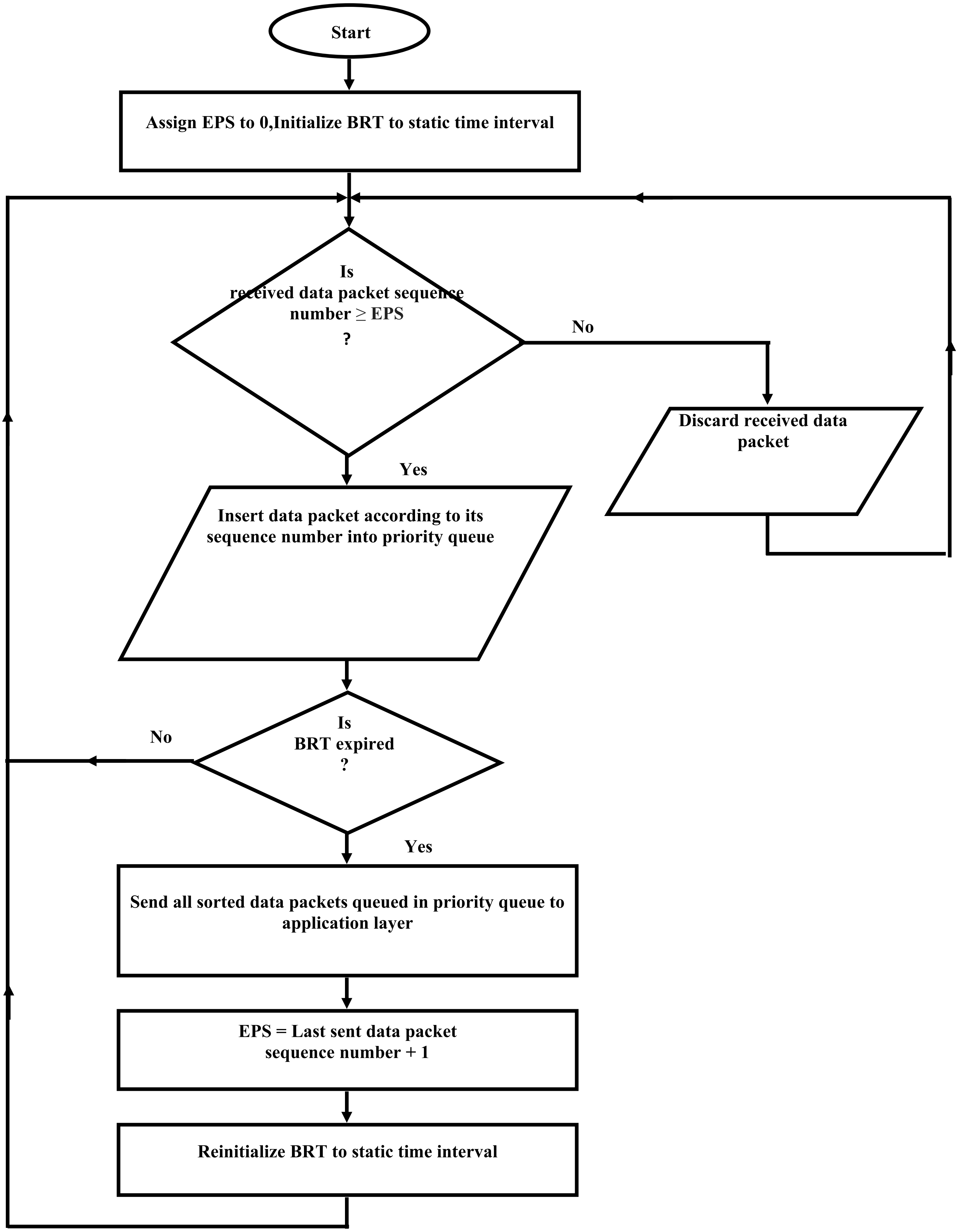


Figure 1: Flowchart of proposed Static Reordering Buffer (SRB) algorithm

Improved Reordering Buffer (IRB) Algorithm

In the proposed Improved Reordering Buffer (IRB) algorithm, the transport layer initializes the control parameters such as EPS and BRT similar to the SRB algorithm. In the proposed Improved Reordering Buffer (IRB) algorithm, received data packets whose sequence numbers match the current Expected Packet Sequence (EPS) value are directly forwarded to the application layer without being stored in the priority queue. In contrast, packets with sequence numbers greater than the EPS value are inserted into the priority queue at the appropriate position based on their sequence numbers. The packets stored in the priority queue are immediately delivered to the application layer whenever the sequence number of a buffered packet becomes equal to the EPS value, or when the Buffer Release Timer (BRT) expires. After each successful delivery of a packet to the application layer, the EPS value is incremented by one to indicate the next expected packet sequence number. Once the BRT expires, any remaining packets in the priority queue are forwarded to the application layer by the transport layer. Subsequently, the BRT is reinitialized with a fixed time interval, and the EPS value is updated to the sequence number of the last delivered packet plus one. Compared with the SRB mechanism, the proposed IRB algorithm significantly reduces queuing delay because only out-of-sequence packets are buffered. As a result, the IRB mechanism improves end-to-end delay performance and enhances the Quality of Service (QoS) for multimedia applications.

SIMULATION AND RESULTS

In this scenario, performance of the proposed SRB and IRB algorithms are evaluated and compared with the AODV routing protocol using QualNet 6.1 network simulator. The scenario designed for the simulation studies includes nodes arranged in a grid with the inter node distance of 49, 81, 121, 169 and 225 nodes. Two ray path loss model with constant shadowing of mean 4dB has been selected to carry out the simulation studies. The remaining simulation parameters considered for simulation studies are listed in Table 1.

Table 1. Simulation Parameters

Parameter	Value
Area	2500m x 2500m
Simulation Time	30 seconds
Number of nodes	49, 81, 121, 169 and 225
Node Placement	Grid
Traffic type	CBR
Buffering Techniques	SRB, IRB and without buffer
Radio type	IEEE 802.11g
Channel frequency	2.412 GHz

Initially, for the 49-node scenario (Figure 2), a 1 Mbps Constant Bit Rate (CBR) link is considered to evaluate the performance of the proposed Static Reordering Buffer (SRB) algorithm by setting the BRT time interval to 125ms. To introduce the out-of-sequence arrival of data packets at the destination node, a random propagation delay with a maximum variance of 30ms is introduced at the air interface of the source node while data packets are transmitted to destination during the simulation process. The performance metrics such as throughput, aggregate messages received, average delay and jitter are recorded. Similar simulation studies are performed by increasing the buffer retention period to 250ms and 500ms. Also, the simulation studies are repeated by considering the scenarios with node density of 81, 121, 169 and 225 nodes. Similar simulation studies are repeated by changing the CBR data rate to 2Mbps and 3Mbps. A similar simulation procedure is also performed for the proposed IRB algorithm. To evaluate the performance of conventional transport layer mechanism, a similar procedure is repeated without varying the BRT time interval.

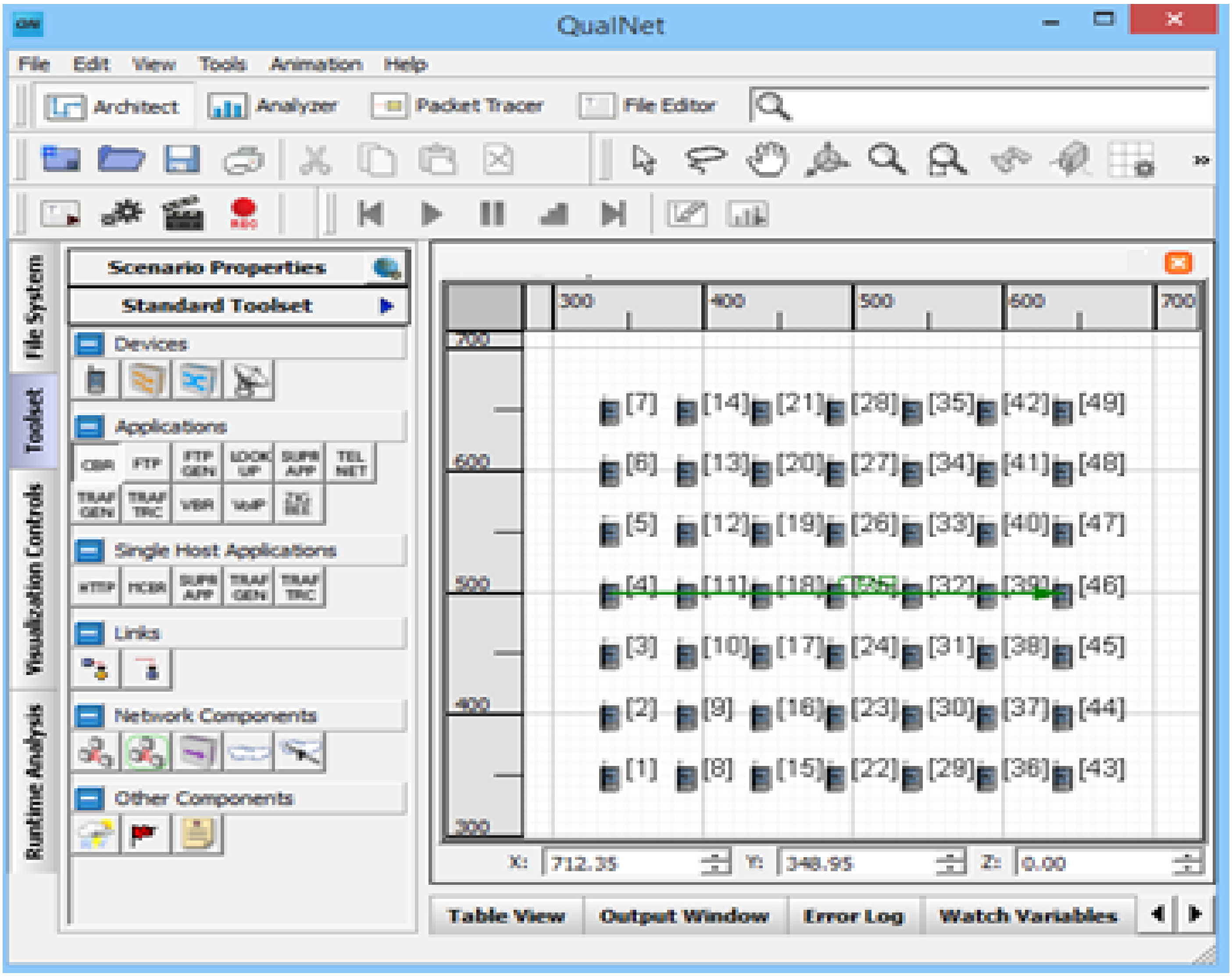


Figure 2: Snapshot of simulation scenario with 49 nodes

Figures 3, 4, 5 and 6 shows the throughput, aggregate messages received, average delay and jitter performance of the proposed SRB, IRB and conventional transport layer mechanisms with increasing node density and data rate for different BRT expiration intervals.

In this simulation study, a random propagation delay is introduced at the air interface of the source node to deliver out-of-sequence data packets to the destination node. In conventional transport layer service, the destination node discards all data packets with a sequence number lower than EPS, resulting in a decrease in the throughput and aggregate message received performance as depicted in Figures 3 and 4. On the other hand, in the proposed SRB algorithm at the transport layer, the received data packets that are greater than or equal to EPS are stored and sorted for the duration of BRT using a priority queue before being sent to the application layer. With this proposed reordering process, the loss of data packets at destination node due to out-of-sequence delivery is reduced, improving the throughput and aggregate message received performance of SRB algorithm.In contrast, if the sequence number of the received data packet matches the EPS value, the IRB sends data packets directly to the application layer without storing them in the priority queue. Only the data packets with a sequence number greater than the EPS value are retained in priority queue until a suitable data packet with a sequence number matching the EPS value arrives or for the duration of the BRT. Therefore, the

ISSNNo.2454-6194 | DOI:10.51584/IJRIAS | Volume XI Issue V May 2026

number of data packets rendered to application layer is slightly higher for IRB compared to the SRB mechanism, resulting in improved throughput and aggregate message received performance of IRB algorithm. From Figures 3 and 4, it is also evident that as the BRT expiration interval increases, the throughput and aggregate message received performance of proposed SRB and IRB algorithms increase because longer BRT intervals reduce out-of-sequence packet drop [19].

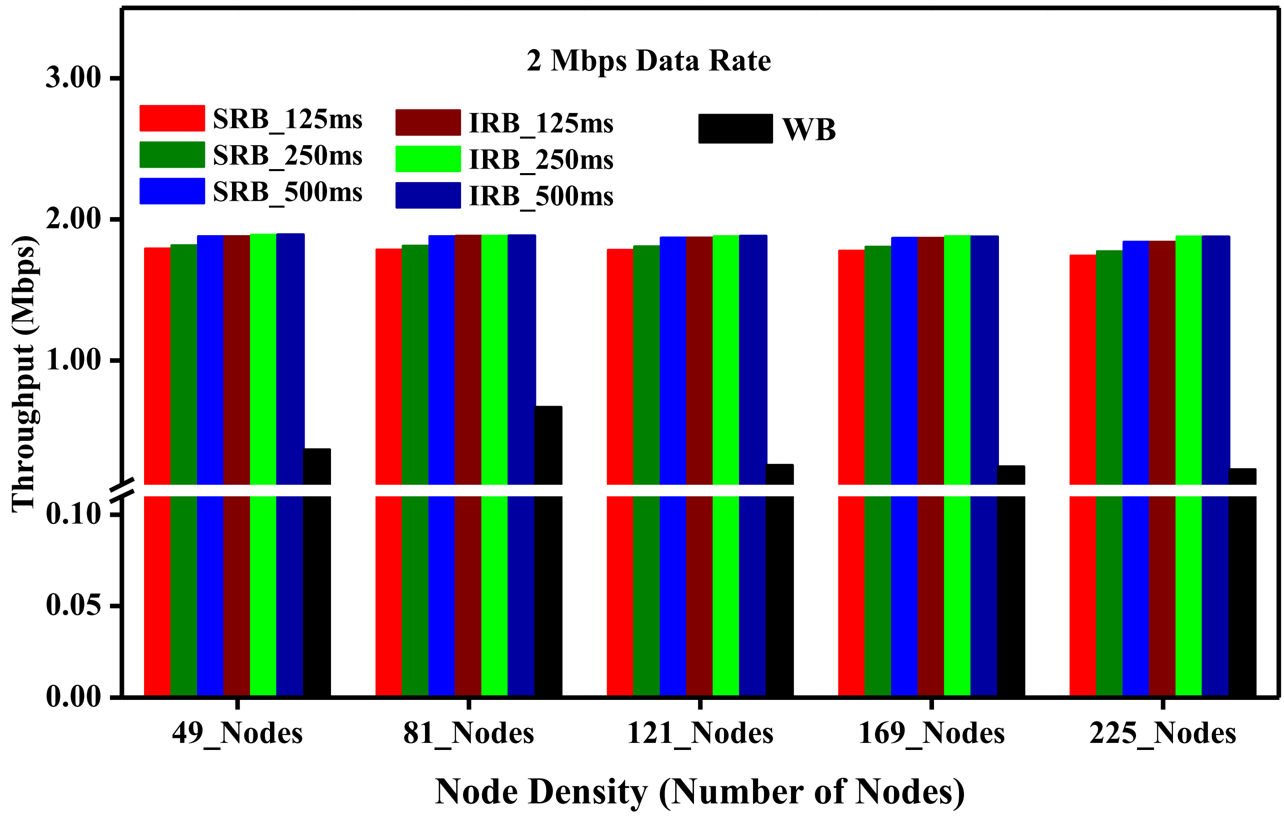


Figure 3: Throughput Performance of SRB algorithm, IRB algorithm and conventional mechanism without bufferfor increasing node density and BRT interval for 2Mbps data rate

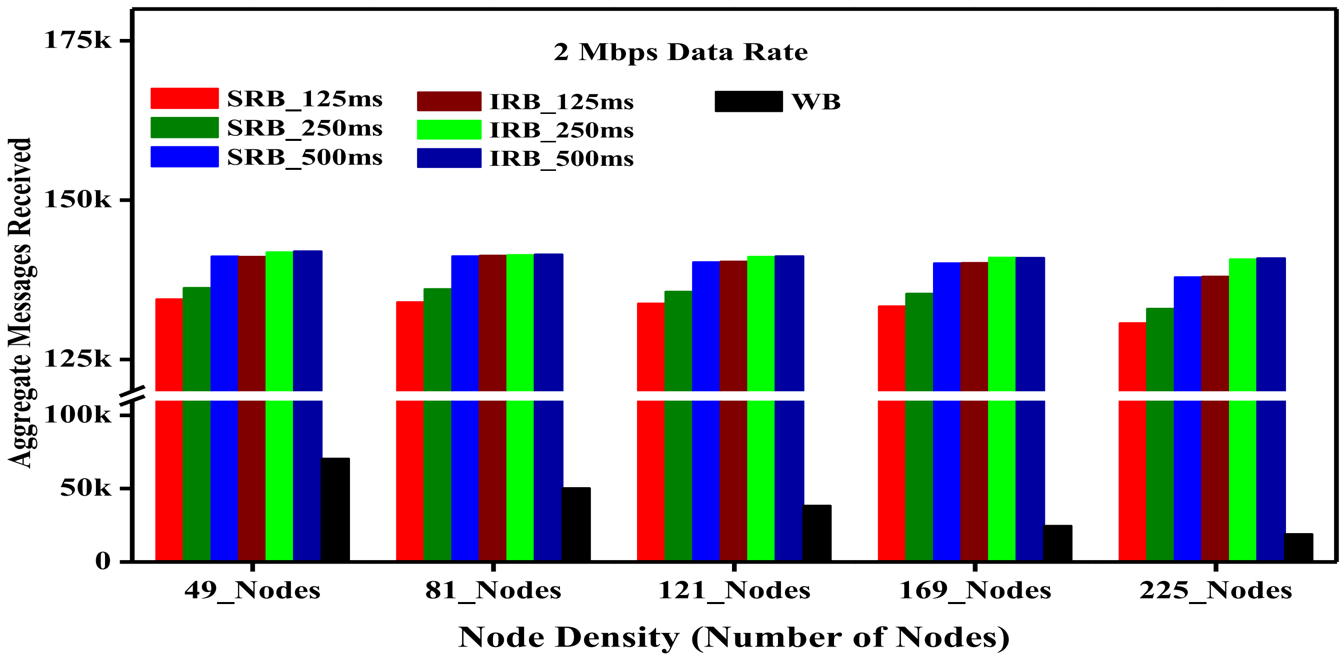


Figure4: Aggregate Messages Received Performance of SRB algorithm, IRB algorithm and conventional mechanism without buffer for increasing node density and BRT interval for 2Mbps data rate

From Figure 5, it is evident that delay performance in the conventional transport layer is better than that of the proposed SRB and IRB mechanisms. This is because in conventional transport layer, the received data packets that match the EPS value are directly sent to the application layer without queuing and processing overhead. Whereas, in proposed SRB algorithm, all sorted data packets are retained in priority queue until the BRT expires, which increases the number of packets occupied in the priority queue. Therefore, there is a large queuing delay and processing overhead for each data packet, resulting in an increase in average delay performance compared to the IRB algorithm [19][20]. In contrast, the IRB algorithm holds back a smaller number of data packets whose sequence number is greater than EPS value and sends them to application layer immediately after the arrival of proper data packet with expected sequence number in accordance with EPS value or after the BRT expires. Therefore, the number of packets queued in the priority queue decreases, which in turn decreases queuing delay and processing overheads leading to improved average delay performance of IRB. Figure 5 also shows that the average delay performance of SRB and IRB algorithms increases as the BRT expiration interval increases, because increasing the BRT interval increases the packet retention time in a priority queue, resulting in a higher average packet delay.

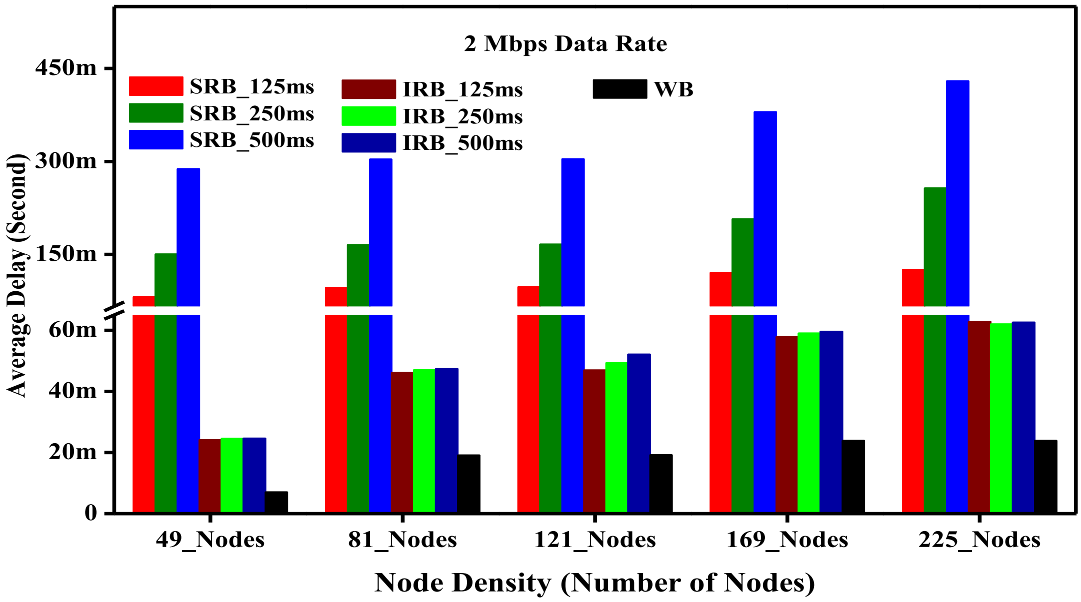


Figure 5: Average delay performance of SRB algorithm, IRB algorithm and conventional mechanism without buffer for increasing node density and BRT interval for 2Mbps data rate

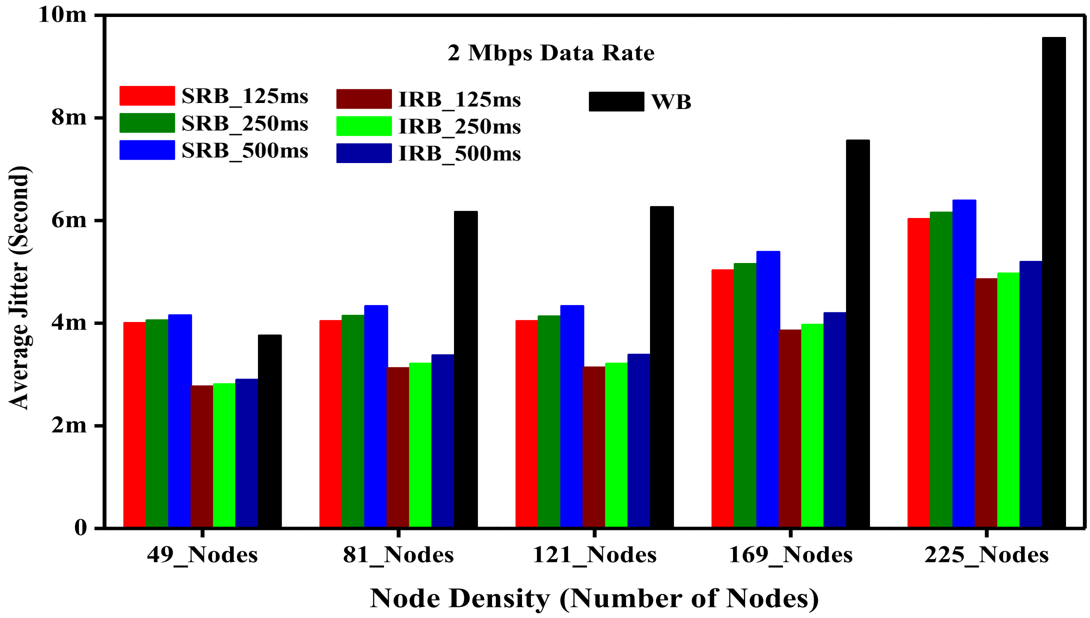


Figure 6: Average jitter performance of SRB algorithm, IRB algorithm and conventional mechanism without buffer for increasing node density and BRT interval for 2Mbps data rate

From Figure 6, it is evident that the jitter performance of the conventional transport service is higher than that of the proposed SRB and IRB algorithms. This is because the received data packets that are not in the correct order are discarded at a random time, which leads to an increase in the variance of the inter-packet interval or jitter performance. It can also be seen from Figure 6 that the jitter performance of SRB algorithm is higher than that of the IRB algorithm. This is because data packets retained in the priority queue experience a high variance in queuing delay until the BRT expires, which increases the jitter performance compared to the IRB algorithm. In the IRB algorithm, on the other hand, data packets with sequence numbers that match EPS are sent directly to application layer and only the data packets that do not match the sequence are retained in priority queue, reducing the variance in queuing delay and processing overhead, resulting in a reduction in jitter performance. Figure 6 also shows that the jitter performance of SRB and IRB algorithms increases as the BRT expiration interval increases, since the data packets are routed through a larger number of intermediate nodes.

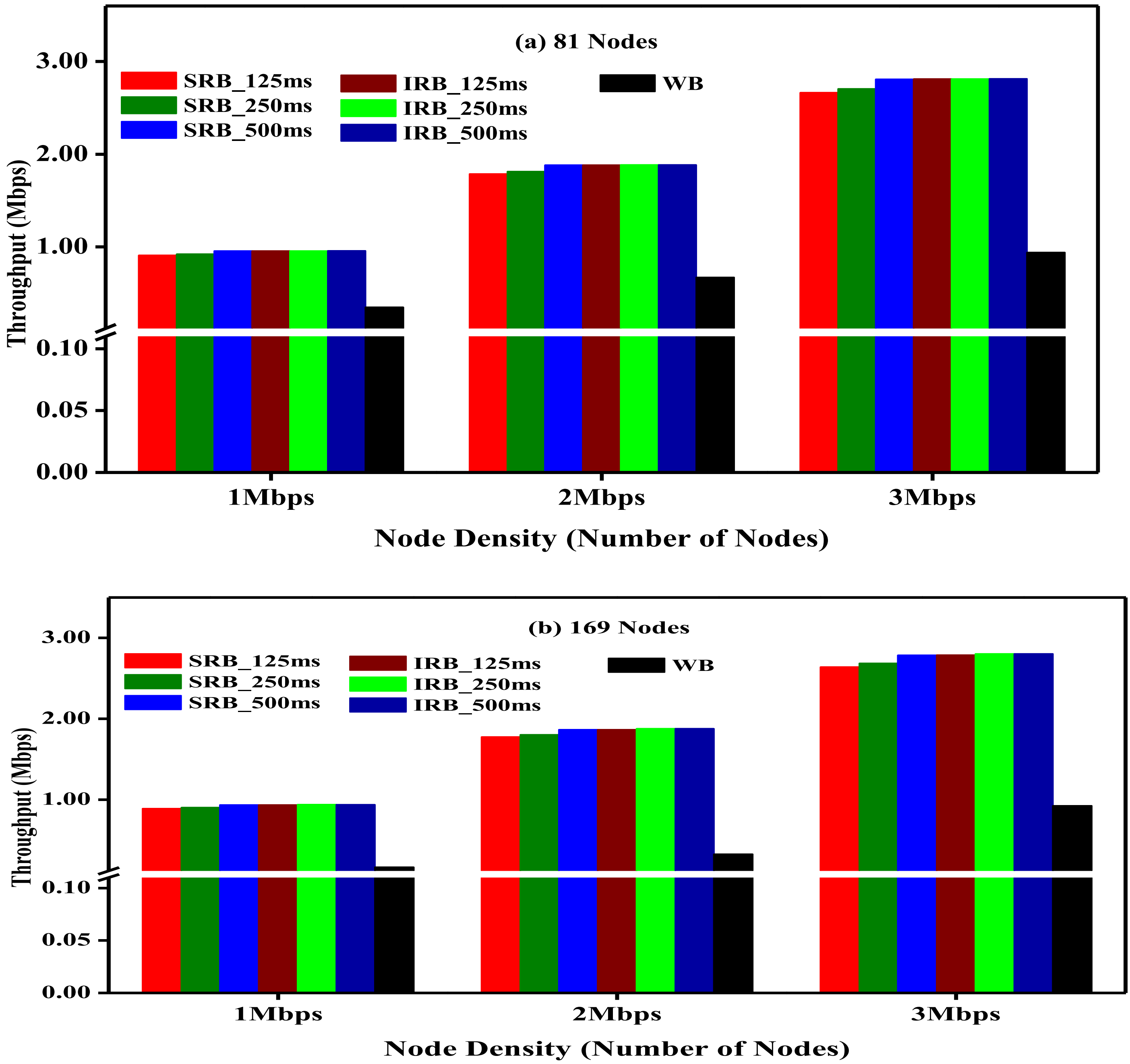
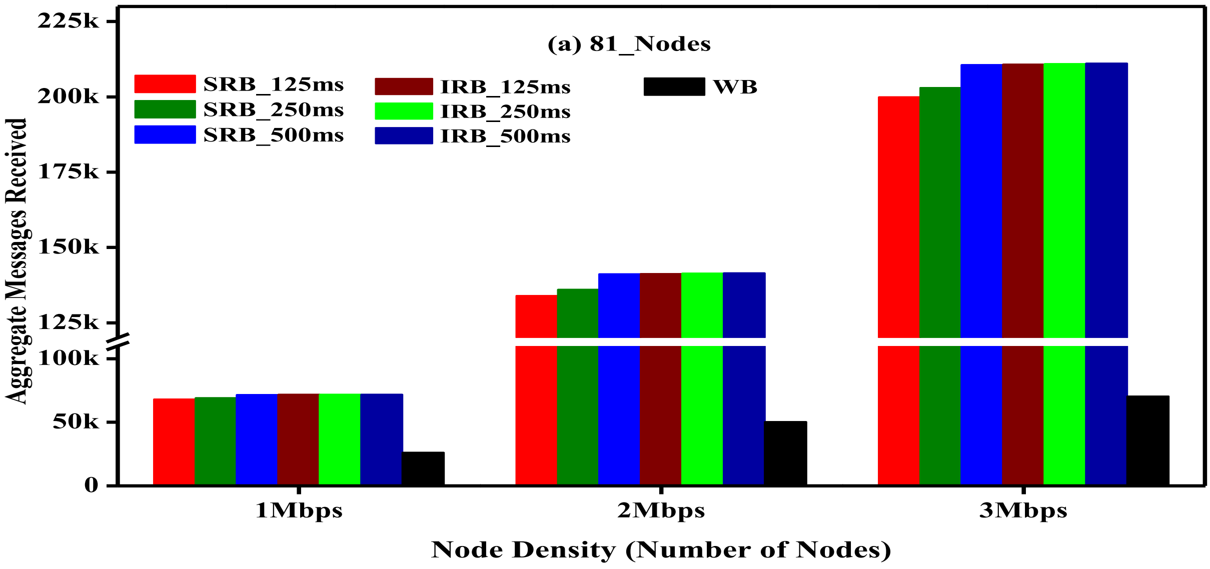


Figure 7(a, b): Throughput performance of SRB algorithm, IRB algorithm and conventional mechanism without buffer for increasing data rates and Buffer Retention period for (a) 81 and (b) 169 node density



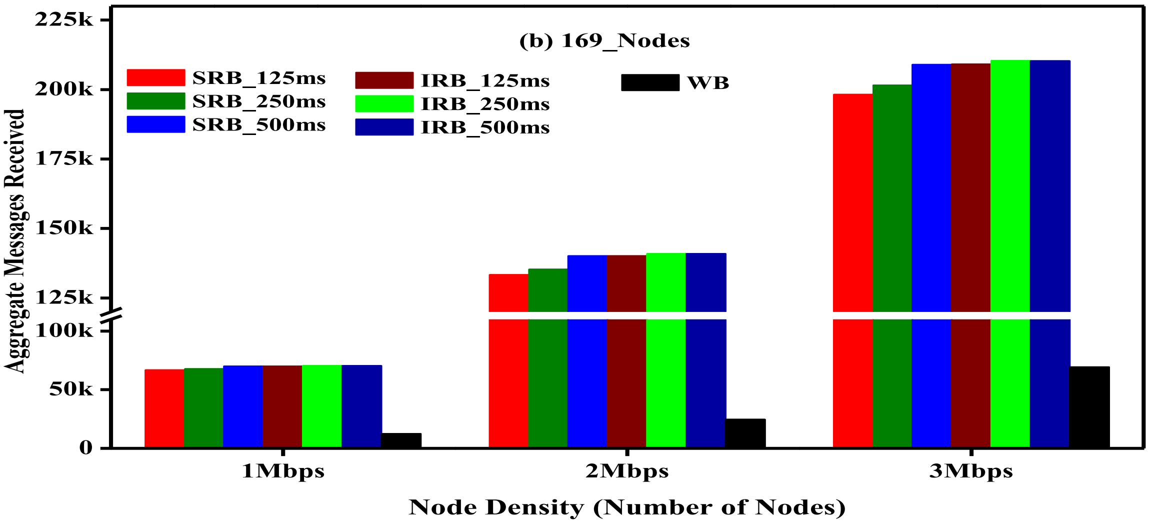
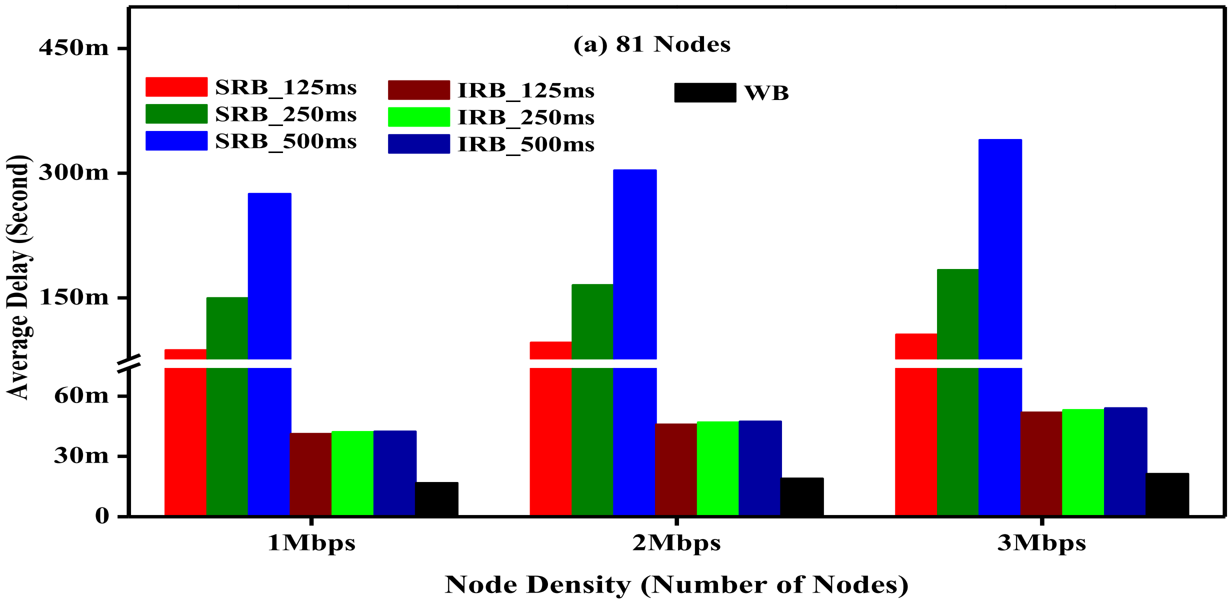


Figure 8(a,b):Aggregate Messages Received Performance of SRB algorithm, IRB algorithm and conventional mechanism without buffer with increasing data rates and Buffer Retention period for (a) 81 and (b) 169 node density



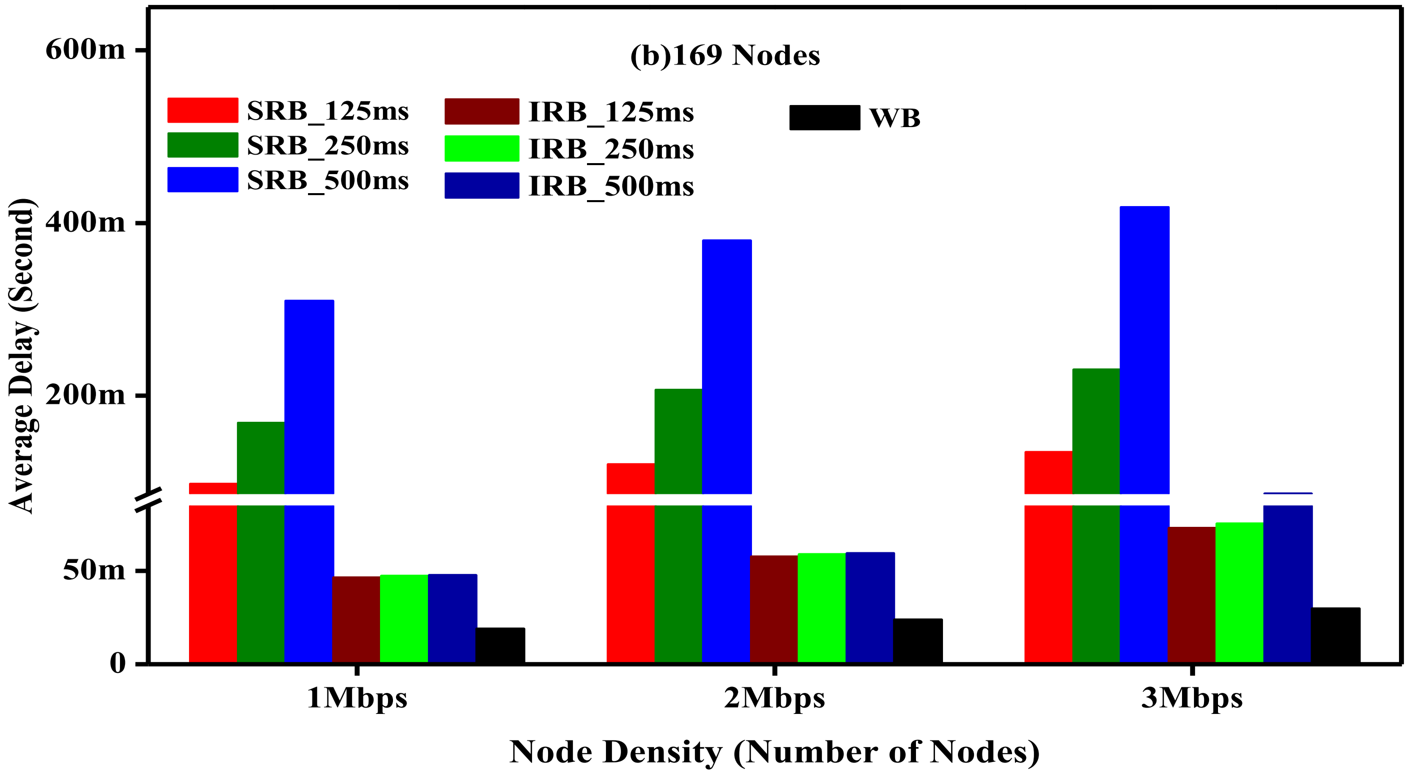
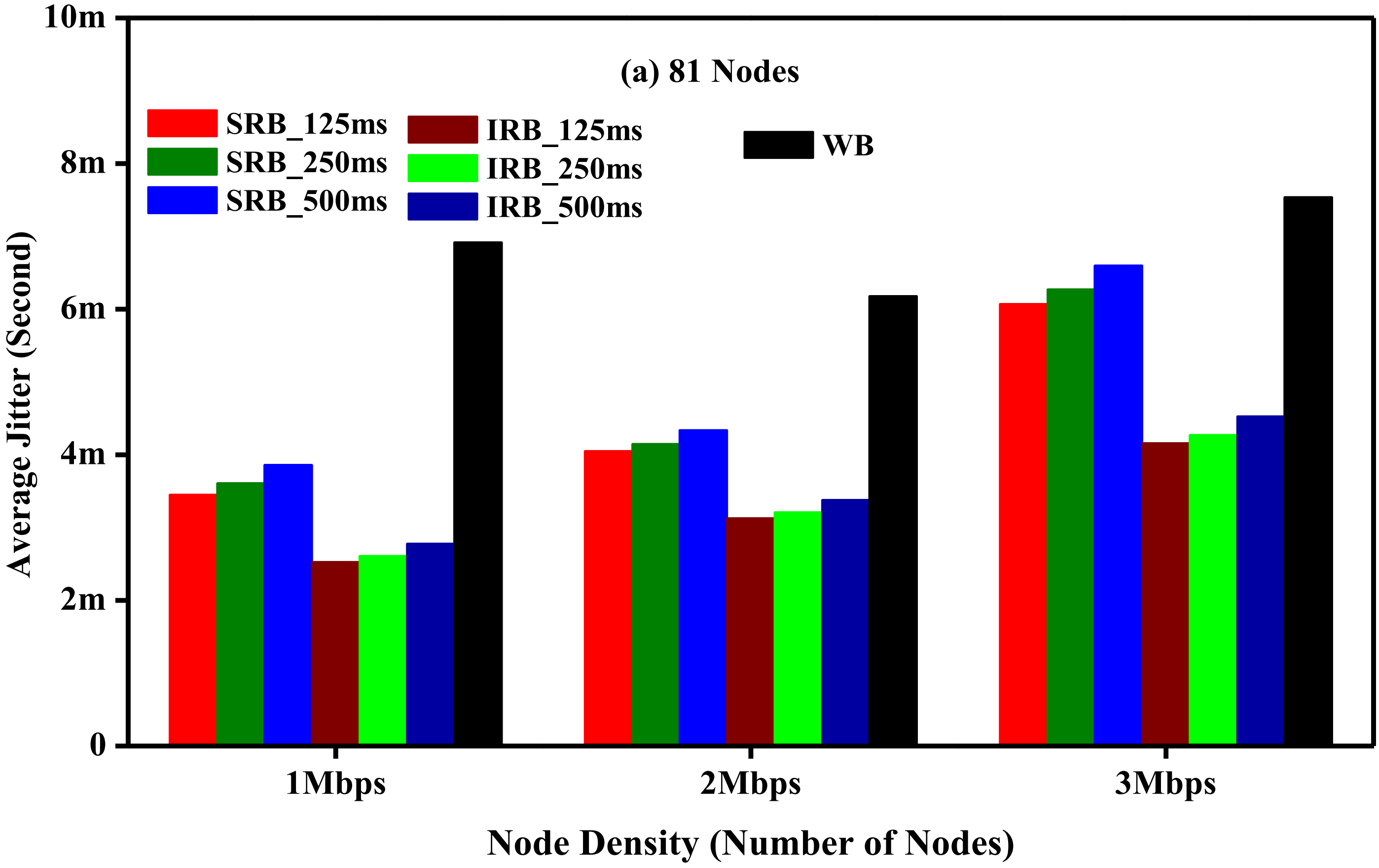


Figure 9(a, b): Average delay performance of SRB algorithm, IRB algorithm and conventional mechanism without buffer with increasing data rates and Buffer Retention period for (a) 81 and (b) 169 node density



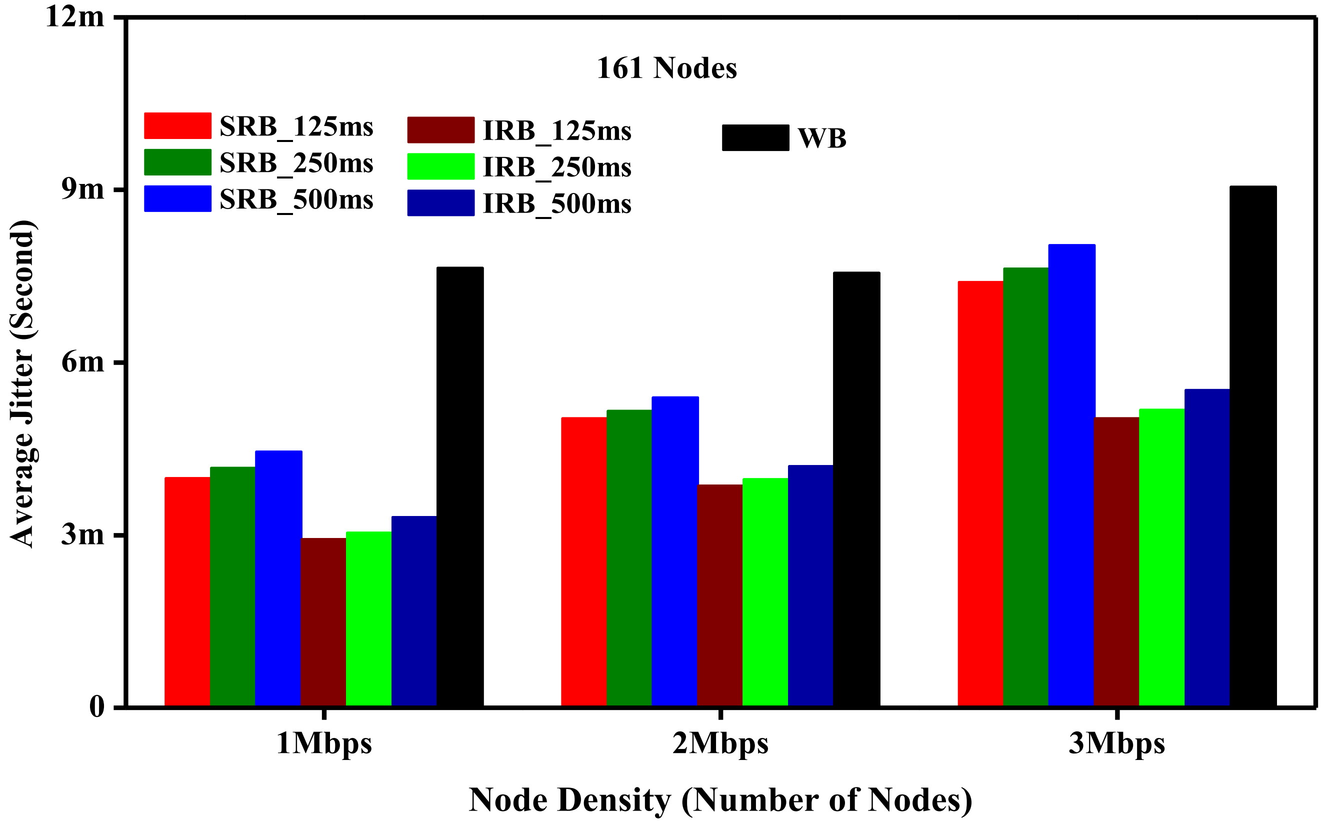


Figure 10(a, b): Average jitter Performance of SRB algorithm, IRB algorithm and conventional mechanism without buffer with increasing data rates and Buffer Retention period for (a) 81 and (b) 169 node density

Figures 7(a, b), 8 (a, b), 9 (a, b), and 10 (a, b) show the throughput, aggregate messages, average delay and jitter performance of the proposed SRB algorithm, IRB algorithm and conventional transport services with increasing data rate and BRT expiration interval for node densities of 81 and 169 nodes. Figures 7(a, b) and 8 (a, b) depicts that the throughput and aggregate messages received performances of both the proposed SRB algorithm, IRB algorithm and the conventional transport layer services increase with increasing data rate as a larger number of data packets are routed to the destination. Figures 9 (a, b) and 10 (a, b) show that the average delay and jitter performance of the proposed SRB algorithm, IRB algorithm and the conventional transport layer services increases with increasing data rate. Since, network congestion and co-channel interference increase the number of retransmissions, thereby increasing the transmission delay [21]. In addition, the number of data packets that do not arrive at the destination in the correct order increases queuing overheads.

CONCLUSION

In Ad Hoc networks, the performance of each communication link continuously varies over time due to factors such as changes in hop count, buffer occupancy, traffic density, and route discovery processes. These variations often lead to packet mismatch, as data packets may not arrive at the destination node in the correct sequence. In conventional transport layer services, such packet discrepancies result in packet discarding, which reduces the achievable throughput of the Ad Hoc network and consequently degrades the Quality of Service (QoS) of multimedia applications. To address this issue, this paper proposes two transport-layer buffering algorithms, namely the Static Reordering Buffer (SRB) and the Improved Reordering Buffer (IRB) algorithms. The proposed mechanisms reorder out-of-sequence packets by arranging them in a priority queue before delivering them to the application layer, thereby improving throughput performance and enhancing QoS for multimedia applications. Simulation results demonstrate that the proposed IRB algorithm achieves better delay and jitter performance than the SRB algorithm due to its reduced processing and queuing delay.

REFERENCES

1. Patrick Stuedi and Gustavo Alonso, "Wireless Ad Hoc VoIP", MNCNA '06 ,2007
2. J. Karlsson; H. Li; J. Eriksson, "Real-time video over wireless ad-hoc networks", 14th International Conference on Computer Communications and Networks, 2005.
3. Sadaf Momeni, Bernd E Wolfinger, "Availability evaluations for IPTV in VANETs with different types of access networks ", EURASIP Journal on Wireless Communications and Networking, 2014
4. Kamal Mistry, Sanjay Srivastava, R. B. Lenin," Buffer Aware Routing in Interplanetary Ad Hoc Network ", COMSNETS'09 Proceedings of the First international conference on COMmunication Systems and NETworks, 2009
5. Thomas D. Lagkas, Georgios I. Tsiropoulos,"Bandwidth Allocation in Cooperative Wireless Networks: Buffer Load Analysis and Fairness Evaluation", Elsevier Physical Communication, 2011.
6. Baban A. Mahmood, D. Manivannan, "GRB: Greedy Routing Protocol with Backtracking for Mobile Ad Hoc Networks", arXiv:1804.01133 [cs.NI], April 2009
7. Xiaoming Zhou and Piet Van Mieghem, "Reordering of IP Packets in Internet"
8. White Paper, "Media Quality and Network Connectivity Performance in Skype for Business Online", Microsoft
9. Nischal M. Piratla, Anura P. Jayasumana, Abhijit A. Bare, Tarun Banka, “Reorder buffer-occupancy density and its application for measurement and evaluation of packet reordering”, 10.1016/j.comcom.2007.03.001, Computer Communications, Elsevier,30 (2007) 1980–1993
10. Raghunandan M. Narasiodeyar, Anura P. Jayasumana, “Improvement in packet-reordering with limited re-sequencing buffers: An analysis”, 38th Annual IEEE Conference on Local Computer Networks, 21-24 Oct. 2013, ISSN(p): 0742-1303, DOI: 10.1109/LCN.2013.6761274
11. Korzun, D.G., Kuptsov, D., Gurtov, A, “A comparative simulation study of deterministic and stochastic strategies for reduction of packet reordering in multipath data streaming”, International Journal of Simulation: Systems, Science and Technology, 2016.
12. Yasin W., Ibrahim H., Abdul Hamid N.A.W., Udzir N.I, “Performance Analysis of Transport Control Protocol Flavours in the Existence of Packet Reordering Phenomena”, Digital Enterprise and Information Systems. DEIS 2011. Communications in Computer and Information Science, vol 194. Springer, Berlin, Heidelberg, DOI-https://doi.org/10.1007/978-3-642-22603-8_50, ISBN(p):978-3-642-22602-1, ISBN(O):978-3-642-22603-8
13. S. Govind, R. Govindarajan and J. Kuri, "Packet Reordering in Network Processor", 2007 IEEE International Parallel and Distributed Processing Symposium, Rome, 2007, pp. 1-10. doi: 10.1109/IPDPS.2007.370287
14. Ka-Cheong Leung, Chengdi Lai, Victor O. K. Li, Daiqin Yang, “A packet-reordering solution to wireless losses in transmission control protocol”, Wireless Networks, October 2013, Volume 19, Issue 7, pp 1577–1593
15. Q. Hu, Q. Tian & Z Tang, “RP-MAC: A Passive MAC Protocol with Frame Reordering for Wireless Sensor Networks”, International Journal of Wireless Information Networks, March 2013, Volume 20, Issue 1, pp 74–80 https://doi.org/10.1007/s10776-012-0184-2
16. Dodda Sunitha, Aitha Nagaraju, Gugulothu Narsimha, “Cross-layer based routing protocol and solution to packet reordering for TCP in MANET”, 2017, DOI 10.1007/s10586-017-1179-7
17. Pooja J Kotian, P Vaishnavi, Shifana Begum, “Review on Data Traffic in Real Time for MANETS”, International Research Journal of Engineering and Technology (IRJET), Vol.6(11), Nov 2017
18. White paper, H. Schulzrinne, GMD Fokus, S. Casner," RTP: A Transport Protocol for Real-Time Applications Audio-Video Transport Working Group”, IETF, Jan. 1996
19. Syed Humair Ali, Syeda Areeba Nasir, Sameer Qazi, "Impact of Router Buffer Size on TCP/UDP Performance", Proceedings of 2013 3rd IEEE International Conference on Computer, Control & Communication.
20. Sarfraz Ahmed Soomro, M. Mujtaba Shaikh, Abdul Fattah Chandio, Nasreen Nizamani, Ehsan Ali Buriro," Impact of Buffer Size on UDP Performance for Real-Time Video Streaming Application ", IJCSNS International Journal of Computer Science and Network Security, VOL.18 No.6, June 2018, p. 157-160

21. Pan Li, Yuguang Fang, Jie Li, "Throughput, Delay, and Mobility in Wireless Ad Hoc Networks", publication in the IEEE INFOCOM 2010 proceedings This paper was presented as part of the main Technical Program at IEEE INFOCOM 2010, IEEE